A digital method to determine the transmissibility of a MEMS device

by

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A thesis submitted to the Graduate Faculty of
Auburn University
in partial fulfillment of the
requirements for the Degree of
Master of Science
Auburn, Alabama
May 5, 2014

Key words: transmissibility, microcontroller, Modified FFT

Abstract

A microcontroller based technique for measuring the transmissibility of a MEMS device is very useful and meaningful. The resulting data can provide feedback to improve the design, modeling and fabrication process. A new method for measuring the transmissibility has been investigated using a microcontroller where the results are displayed on a personal computer real time. The microcontroller used in the research is the Stm32f407. The purpose of the application is proposed and the practicability is investigated as well. A cantilevered structure and three materials are chosen to verify the technique. The cost for the implementation is estimated and found to be low. The resulting technique can be used in research to process the transmissibility data real time and analyze the data for future applications.

Acknowledgments

I would like to thank my advisor, Dr. Robert Dean, without whom none of this work could have been done. His wise instruction and patient support provided me with constant help to deal with every academic problem. I would also like to thank to Dr. Thomas Baginski and Dr. Thaddeus Roppel for their guidance of my graduate studies. I would also like to thank Chong Li, Haoyue Yang and Luke L. Jenkins for their generous help throughout this project. Without their help, the design would have been difficult to finish. I am also appreciative of my parents for their consistent support.

Table of Contents

List of Figures

List of Abbreviations

APB	Advanced Peripheral Bus
ADC	Analog-to-digital Converter
DMA	Direct Memory Access
DAC	Digital-to-Analog Converter
DFT	Discrete Fourier Transform
GUI	Graphical User Interface
FFT	FAST Fourier Transform
FSM	Finite state Machine
IDFT	Inverse Discrete Fourier Transform
MCU	Microcontroller Unit
PIC	Peripheral Interface Controller
SDOF	Single Degree of Freedom
USB	Universal Serial Bus
VCP	Virtual COM port

List of tables

Chapter 1 Introduction

The relation between force on physical systems and the motion caused by the force is a subject that has been studied by humans since ancient times. The analysis and prediction for the dynamic character of physical systems is important for modern engineering. One omnipresent kind of dynamic behavior is vibration, where the system oscillates about a certain equilibrium position. The research for the thesis is to develop a microcontroller based system to measure the transmissibility of a vibrating MEMS structure.

The analysis of transmissibility is an effective method to analyze the MEMS device's characteristics and vibration features. The transmissibility can display two important characteristics for MEMS device, one is resonant frequency and the other is mechanical quality factor. A dynamic signal analyzer is often used to analyze the transmissibility of the MEMS device. It is relatively a high cost and complex piece of laboratory test equipment. Furthermore, the transmissibility is difficult to extract from the analyzer in real time. Finding a convenient and simple way to measure the transmissibility of a MEMS device in real time is considered in this research. The Stm32f407 microcontroller is based on the high-performance ARM Cortex M4 32-bit RISC core, which has a sufficiently strong calculation ability. By using a Stm32f407, the output and input data of the MEMS device under test can be extracted and processed. For the MEMS device, the ratio of the magnitudes of the output to the input of the vibration is the transmissibility of the system. A modified FFT algorithm is used to transform the system from time domain to frequency domain. Visual studio is used to display the results, with a refresh period of 4 seconds. By averaging the data, the noise in the data can be decreased and the figure will be updated every 20s. A function generator is used to preliminary verify the results, and then a vibration system and a laser system are used to test the accuracy of the results for the device under test. From the results, the resonant frequency of a MEMS device can be determined from the transmissibility plot for the MEMS device and this information can be observed in real time successfully. This results if this research can reduce the cost of the equipment and can meet the requirement to display the data in real time.

Furthermore, the data to display the transmissibility can be written out to a personal computer real time, and it is easy to take the data and analyze it, thus it will be helpful to the MEMS researcher.

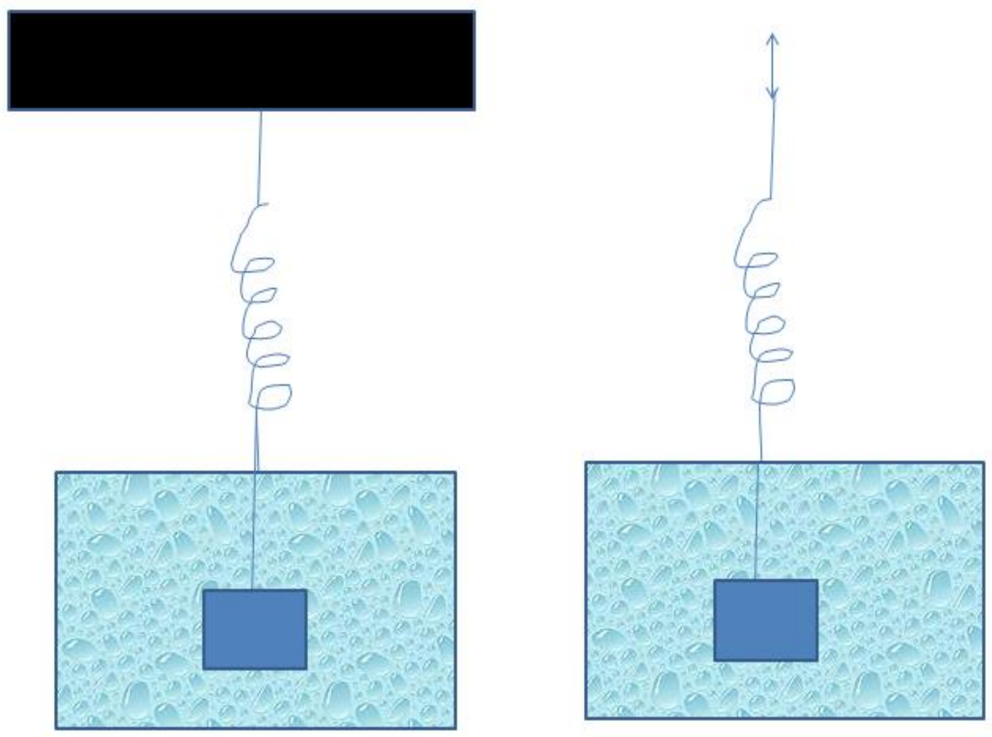
Chapter 2 Literature review

Section 1 MEMS

Micro-electro-mechanical Systems (MEMS) can be defined as a technology of miniaturized mechanical and electro-mechanical elements which are made by the process of microfabrication. MEMS technology evolved from the integrated circuit industry. Microfabrication is the engine for MEMS. From the 1990s, the development of MEMS came to a worldwide rapid and dynamic growth period. Two applications are notable; one is the integrated inertial sensor that is made by Analog Devices for the deployment of automotive air-bags, and the other is the Digital Light Processing chip that is made by Texas Instruments for projection displays. MEMS have the advantages of high sensitivity and low noise, and it also reduces the costs of each sensor by removing manual assembly steps and by using batch fabrication to manufacture them. Nowadays, one can find that different micromachined acceleration sensors are the basis of sensing principles and fabrication technologies. Furthermore, they have better features and better reliability. In all, MEMS has three generic and distinct merits, microelectronics integration, miniaturization, and mass fabrication with precision.

Section 2 Transmissibility

Figure 1 shows two types of mechanical oscillator; they can be seen as simple mechannical systems.



(a) (b)

Figure 1 Mechanical System

Mechanical resonance

Mechanical resonance is a kind of response of the mechanical system where it can obtain the highest amplitude of motion at that frequency when subjected to a force with components at that frequency. The mechanical system can sometimes experience violent swaying motions at the resonant frequency, and furthermore, this has the possibility to cause catastrophic failure for structures such as bridges, buildings and airplanes if they are designed improperly. Commonly, a dynamic signal analyzer is used to test the mechanical resonance.

To find the mechanical resonance is to find the resonance of a mechanical oscillator. Any mechanical body, if it is free to vibrate, has its natural periods of oscillation. For an example, a thin steel rod, when it is struck, will oscillate if it is held at one end. The oscillation of the rod depends on its length, mass, and other factors. If the wire or string is plucked it will vibrate back and forth if the string is supported at the ends.

The natural frequencies of the wire depend on the diameter, density, tension and length of the wire. For both cases they vibrate for a time depending on their own natural, peculiar periods. The structure will lose the energy little by little, and at last stop vibrating. If the rod is designed to be connected to an oscillating source of energy that has the natural frequency of it, the oscillation of the rod will be very large, and it will start to vibrate in sympathy with that source. One type of vibrating structure that loses energy fairly quickly is the damped oscillator.

The quality or Q factor for a mechanical system is a parameter that tells how under-damped an oscillator is. The Q factor can also be used to describe the system’s bandwidth relative to the center frequency. High Q means lower energy loss relative to the energy stored in the oscillator and it will damped more slowly.

As shown in Figure 1 (a)， the mass is at rest at first and the weight is balanced by the spring force. When the mass is moved from the original position and released, it starts to oscillate. The resistive force acts on the mass because it is immersed in a fluid, usually air, and it has high Q factor. The force opposes the motion of the mass and it is proportional to the object's velocity. Then, by this force, the oscillations are damped.

The other type is an external oscillator as in Figure 1 (b). If the spring's upper part is not fixed, but the spring moves up and down by way of simple harmonic motion. Therefore, the spring has an applied force that changes sinusoidally. So a damped oscillator is driven by a kind of external oscillator. The frequency of the external oscillator sometimes is not the same as the frequency of the damped oscillator, but when the damped oscillator can be driven by the force and the frequency of this force is close to its natural oscillation frequency, the mass will vibrate by the "resonance".

The resonant behavior can be performed by two different ways. One is amplitude and the change of the amplitude of the resonance is more dramatic. The other type is phase shift, it is sometimes clearer and it is the shift of angular position between the oscillating object and the external driver. The amplitude is at the point of maximum f =w/2π, if the spring constant is k, the mass is m, the result should be smaller off of the point of the resonance, both for $f>f_0$ and for $f<f_0$. Actually, because of the damping, the frequency response is down to $f_0 = \frac{1}{2\pi}\sqrt{k/m - R^2/2m^2}$.

Transmissibility

Transmissibility is the magnitude response of the system when the input is a time varying displacement to the frame and the output is the displacement of the proof mass. Transmissibility is widely used in the domain of vibration isolation and shock where one can minimize or control the transmission of an undesired motion and force from an external source to a mechanical system or device, such as a machine tool, shop floor or a delicate package. Transmissibility can be defined as a ratio of output motion to input motion for a linear system.

If |T(jω)|>1, it means amplification of the system response and the maximum amplification will occur when the forcing frequency is the same as the resonant frequency. Transmissibility has two major kinds, one is force transmissibility and the other is motion transmissibility. The condition is similar to the frequency response function that can be used to define transmissibility functions: At steady state, a linear system is excited at the frequency of interest by a harmonic input. This example can explain how the frequency response function and H(jw) can dictate the amplitude and the phase for a linear system

$$f(t) = F_0 \cos(\omega_0 t) \tag{2.1}$$

$$x(t) = F_0 \| H(j\omega_0) \| \cos(\omega_0 + \angle H(j\omega_0)) \tag{2.2}$$

$$x(t) = -\omega_0 F_0 \| H(j\omega_0) \| \sin(\omega_0 + \angle H(j\omega_0)) \tag{2.3}$$

Although transmissibility functions are almost the same as frequency response functions, because transmissibilities are ratios, they are inherently normalized unlike the frequency response functions.

Force Transmissibility

The first concept is force transmissibility, a forced response of Single Degree of Freedom linear system. Force transmissibility is an important concept in transmissibility. The Figure 2 is a SDOF system. In the figure, m, k and c are the mass, linear stiffness parameter and linear viscous damping. The input is f and the responses are x and y.

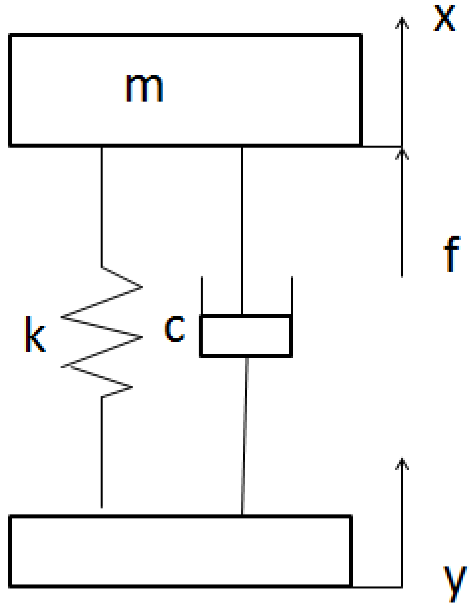


Figure 2 SDOF System

The equation of motion when the base is fixed is

$$m\ddot{x}(t)+c\dot{x}(t)+kx(t)=f(t) \tag{2.4}$$

At the frequency ω_0, the force function with the amplitude F_0 is:

$$f(t)=F_0\cos(\omega_0 t) \tag{2.5}$$

Under the giving force, X_p is the amplitude of the response and the phase lag for the response is ϕ_{XF}. And the function is:

$$x_p(t) = X_p \cos(\omega_0 t - \phi_{XF}) \tag{2.6}$$

Then, putting (2.5) and (2.6) into the(2.4) , the new equation can be obtained:

$$X_p[(k - m\omega_0^2)\cos(\omega_0 t - \phi_{XF}) - c\omega \sin(\omega_0 t - \phi_{XF})] = F_0 \cos(\omega_0 t) \tag{2.7}$$

Since normalization is gotten from the vibration concept, the undamped natural frequency is

$$\omega_n = \sqrt{k/m} \tag{2.8}$$

Where ω_n is the normalized frequency, ω_0 is the forcing frequency, and r is the ratio of these two values.

$$r = \omega_0 / \omega_n \tag{2.9}$$

From the (2.7) the amplitude X_p and the phase angle ϕ_{XF} can be found:

$$X_p = \frac{F_0}{\sqrt{(k - m\omega_0^2)^2 + (c\omega_0)^2}} \tag{2.10}$$

$$\phi_{XF} = \tan^{-1}\left(\frac{c\omega_0}{k - m\omega_0^2}\right) = \tan^{-1}\left(\frac{2\zeta r}{1 - r^2}\right) \tag{2.11}$$

where ζ is defined as:

$$\zeta = \frac{c}{c_c} = \frac{c}{2\sqrt{km}} \tag{2.12}$$

and c_c is the critical damping, it represents the transition between non-oscillatory and oscillatory motion for the homogenous solution. Then force is transmitted through the damper and spring, and the transmitted force model can be found between the base and the mass.

$$f_t = kx(t) + cx(t) = kX_p \cos(\omega_0 t - \phi_{XF}) - c\omega_0 X_p \sin(\omega_0 t - \phi_{XF}) \tag{2.13}$$

The transmitted force can be reduced to a harmonic function including the amplitude F_T and its phase angle ϕ_{FF} .

$$f_T(t) = F_T \cos(\omega_0 t - \phi_{XF} - \tan^{-1}(c\omega_0 / k)) = F_t \cos(\omega_0 t - \phi_{FF}) \tag{2.14}$$

So F_T and ϕ_{FF} can be written as:

$$F_T = X_p \sqrt{k^2 + (c\omega_0)^2} = \frac{F_0 \sqrt{k^2 + (c\omega_0)^2}}{\sqrt{(k - m\omega_0^2)^2 + (c\omega_0)^2}} \tag{2.15}$$

$$\phi_{FF} = \tan^{-1}[\frac{mc\omega_0^3}{k(k - m\omega_0^2) + (c\omega_0)^2}] = \tan^{-1}[\frac{2\zeta r^3}{(1 - r^2) + (2\zeta r)^2}] \tag{2.16}$$

Then, the transmissibility ratio, TR, is defined as

$$TR = \frac{F_T}{F_0} = \sqrt{\frac{k^2 + (c\omega_0)^2}{(k - m\omega_0^2)^2 + (c\omega_0)^2}} = \sqrt{\frac{1 + (2\zeta r^2)}{(1 - r^2) + (2\zeta r)^2}} \tag{2.17}$$

Motion Transmissibility

The second concept is motion transmissibility. The base is moved and testing how much the motion is transmitted to the mass, and the external force is set to zero at this concept. The amplitude and the phase lag should be considered for this condition. The equation of motion for motion transmissibility is:

$$m\ddot{x}(t) + c\dot{x}(t) + kx(t) = c\dot{y}(t) + ky(t) \tag{2.18}$$

For y(t), it can be defined as:

$$y(t) = Y_0 \cos(\omega_0 t) \tag{2.19}$$

where the right side of Eq.(2.18) can be converted to :

$$c\dot{y}(t)+ky(t)=-c\omega_0 Y_0\cos(\omega_0 t)+kY_0\cos(\omega_0 t)=Y_0'\cos(\omega_0 t-\phi_{YY}') \qquad (2.20)$$

Through the property of the orthogonality for sine and cosine, the amplitude and the phase lag is:

$$Y_0' = Y_0\sqrt{k^2+(c\omega_0)^2} \qquad (2.21)$$

$$\phi_{YY}' = \tan^{-1}(\frac{-c\omega_0}{k}) \qquad (2.22)$$

Eq. (2.6) is the form for the motion in steady state.

Then put (2.6) and (2.20) into the (2.18), the new equation becomes:

$$\{Xp[(k-m\omega_0^2)\cos(\phi_{XX})+c\omega_0\sin(\phi_{XX})]=Y_0'\cos(\phi_{YY}')\}\cos(\omega_0 t) \qquad (2.23)$$

$$\{Xp[(k-m\omega_0^2)\sin(\phi_{XX})-c\omega_0\cos(\phi_{XX})]=Y_0'\sin(\phi_{YY}')\}\sin(\omega_0 t) \qquad (2.24)$$

Xp and ϕ_{XX} can be gotten:

$$Xp=\frac{Y_0'}{\sqrt{(k-m\omega_0^2)^2+(c\omega_0)^2}}=\frac{Y_0\sqrt{k^2+(c\omega_0)^2}}{\sqrt{(k-m\omega)_0^2+(c\omega_0)^2}} \qquad (2.25)$$

$$\phi_{XX}=\tan^{-1}[\frac{mc\omega_0^3}{\sqrt{(k-m\omega_0^2)^2+(c\omega_0)^2}}]=\tan^{-1}\left[\frac{2\zeta r^3}{(1-r^2)+(2\zeta r)^2}\right] \qquad (2.26)$$

So the output displacement over the input displacement transmissibility is:

$$TR=\frac{X_P}{Y_0}=\sqrt{\frac{k^2+(c\omega_0)^2}{\sqrt{(k-m\omega_0^2)^2+(c\omega_0)^2}}}=\sqrt{\frac{1+(2\zeta r)^2}{(1-r^2)^2+(2\zeta r)^2}} \qquad (2.27)$$

It is equivalent to measure the transmissibility for any two displacement or acceleration response variables. And the equivalence for the transmissibilities of various response variables is very important since acceleration is easier to measure than displacement or velocity due to the inertial frame that can be used to measure acceleration. Acceleration is a good motion measurement since the Fourier transform only can be used in the environment at steady state. Furthermore, regardless of the excitation level, the magnitude and the phase lag of the transmissibilities will keep the same in the forced linear system.

Section 3 A technique Way to analyze and display the mechanical resonance

Analyzer

Figure3 is a type of analyzer to test the transmissibility of the MEMS device.

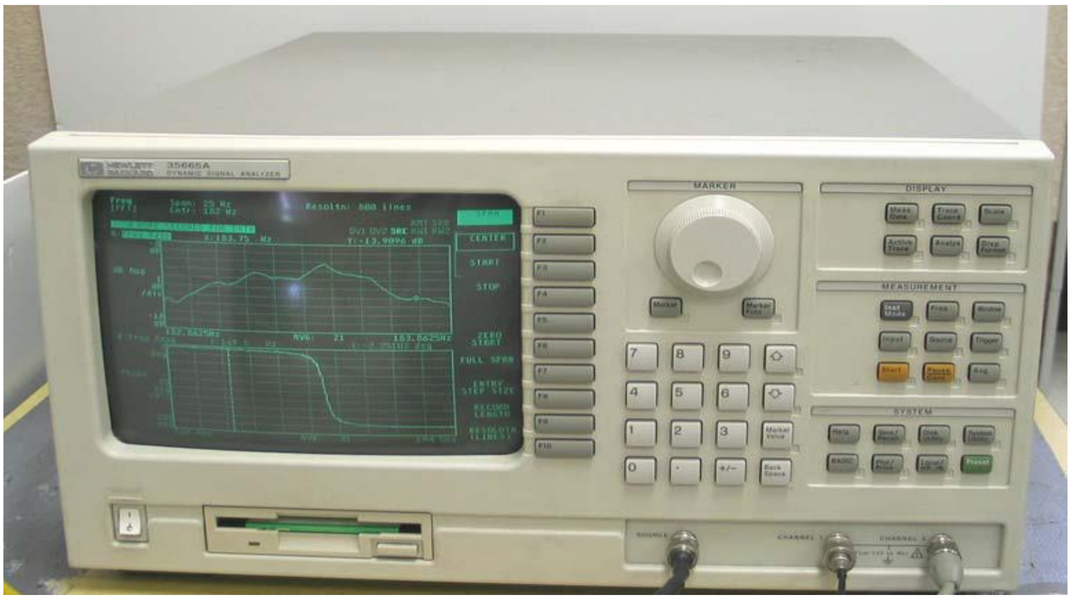


Figure 3 Hp 35665 Dynamic Signal Analyzer

There are some tools to test the transmissibility of a mechanical system. For an example, the Hp 35665 Dynamic Signal Analyzer, as shown in Figure 3, is produced by Agilent Technologies and is used to analyze the results of the mechanical system. This analyzer has two input signals and the maximum resolution of the signal is 800 lines, the other options for the lines of resolution are 100, 200 and 400 lines. The analyzer has one channel at the frequency of 102.4 KHz which has spans from 102.4 KHz to 0.019531Hz, two channels at 51.2 KHz which have spans from 51.2 KHz to 0.097656 Hz. The average real time bandwidth is above 12.8 KHz. The graphs can be observed both in the time and frequency domains. It has a source output port, and this port can provide different kinds of signals, such as random, burst random, pink noise, sine, burst chirp, swept-sine, and arbitrary signals. Furthermore, the small shaker can use the dynamic analyzer to control the power amplifier. It can be used for different types of measurement, such as linear spectrum, power spectral density, time waveform, PDF, CDF, frequency response, and cross-spectrum. For the measurement rate, it has a 401 point FFT to display, and the fast average for one channel mode above 33 averages/ second (typical) and for one channel it is above 15 averages/ second (typical). Its input noise level is 140 dbVrms/(sq.R)Hz when the frequency is above 1280Hz and -130 dbVrms/(sq.R)Hz when the frequency is 160Hz to 1.28 kHz. The full span FFT noise floor for the analyzer is less than -76db, and it is -85db in the typical condition. The gain accuracy of the FFT Cross-Channel is ± 0.04 db (0.46%) and the phase accuracy of the FFT Cross-Channel is ± 0.5 degree. The minimum frequency resolution for the analyzer is 122μHz. This machine has very good performance, it has low noise inputs and high speed computation ability, but it is a very complex design and requires a high cost to purchase it.

Bode figure

Figure 4 is an example of Bode figure. A Bode figure is a graph to display the property of the system frequency response, using a log-frequency axis, as a transfer function. The Bode magnitude figure (in decibels) is the frequency response gain and the phase (in degrees) response is the frequency response phase shift. A Decibel is a not a number, it is a ratio relationship of two number using a logarithmic scale.

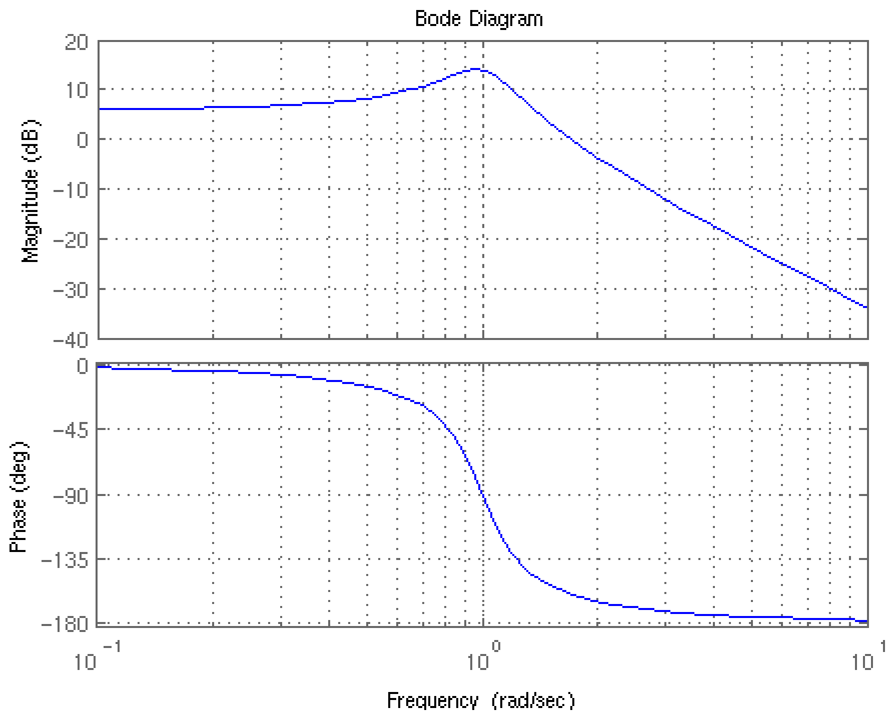


Figure 4 Bode Figure

The decibel is used to express the magnitude axis by the 20 log rule. 20 log rule means the value is 20 times the value of the log of the amplitude. Every tickmark of the axis represents a 10 times power. Because each unit is a power of 10, the decade is referred in the figure.

For the $\angle T(j\omega) = \tan^{-1}(\frac{X}{R})$ Bode Gain Calculations

The magnitude for the transfer function T can be defined as:

$$|T(j\omega)| = \sqrt{R^2 + X^2} \tag{2.28}$$

Where R is a real part, X is an imaginary part.

Because it is sometimes difficult to transfer a function from the "numerator/denominator" to "real+ imaginary", so it uses a fraction with denominator and numerator to defined it:

$$T(j\omega) = \frac{\prod_n | j\omega + z_n |}{\prod_n | j\omega + P_m |} \tag{2.29}$$

Where Zn and Pm are the breakpoints of the figure, at these points, in direction, they get the largest change.

Gain can be calculated by converting both sides to decibels, and using logarithms:

$$Gain = \sum_n 20\log(j\omega + z_n) - \sum_m 20\log(j\omega + p_m)\ . \tag{2.30}$$

Bode phase shift

Bode phase figures reflect the properties of the phase shift at each input frequency and the Fourier Transform can account for its characteristics of the system. In the "real + imaginary" form, the function is given as:

$$\angle T(j\omega) = \tan^{-1}(\frac{X}{R}) \tag{2.31}$$

Section4 Sampling and FFT

The Fourier transform is one of the most important methods to transform data functions and sequences, to transform the time domain of the data to the frequency domain. The Fourier transform is often used in the domain of linear system analysis, optics, antenna studies, random process modeling, quantum physics, probability theory and boundary-value problems, furthermore, and it can be used in the restoration of astronomical data successfully.

Any periodic function and frequency can be written as a linear function made by sine and cosine functions. This feature is called the Fourier series expansion of the function. The FFT is a way to get such amplitude and frequency information for some functions and sequences, which are not periodic.

Discrete Fourier Transform (DFT)

The DFT is to transfer the discrete periodic sequence f[k] in the time domain to another discrete sequence F[j] in the frequency domain. K is an integer and N is the period.

The DFT can be defined as:

$$F[j]=\sum_{k=0}^{N-1} f[k]e^{-2\pi iki/N}, \qquad 0\le j\le N-1; \qquad (2.32)$$

$$f[k]=\frac{1}{N}\sum_{k=0}^{N-1} F[j]e^{2\pi iki/N}, \qquad 0\le k\le N-1; \qquad (2.33)$$

The sequence f[k] is a linear sequence of the N sinusoids, and it is the sum of e^{0} … $e^{(2\pi/N)k(N-1)}$, $e^{(2\pi/N)k(N-1)}$ and F[0], F[1]…F[N-1] are the coefficients of the sinusoids, meantime the frequencies are j/N pre sample.

Sampling

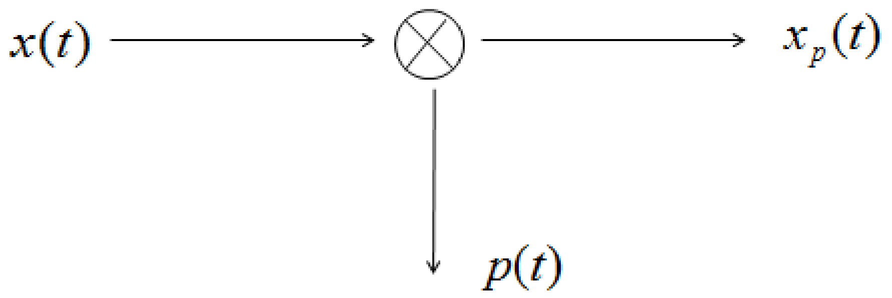


Figure 5 Sampling Process

x(t) is a continuous signal that needs to be sampled and p(t) is an impulse train.

$$p(t)=\sum_{k=-\infty}^{+\infty}\delta[t-kT], \tag{2.34}$$

As shown in the Figure 5, $x_p(t)$ is the new sequence that the data of this sequence equals to the data of x(t) at the times of the sampling period, and other points are zero.

$$x_p(t)=x(t)p(t)=\sum_{n=-\infty}^{+\infty}x[nT]\delta[t-nT], \tag{2.35}$$

In certain conditions, a continuous signal can be represented by discrete sequences and this method is sampling the data points from the periodic signal.

$p(j\omega)$ is the result of the Fourier transform for an impulse, it can be expressed as

$$p(j\omega)=\frac{2\pi}{T}\sum_{k=-\infty}^{+\infty}\delta(\omega-k\omega_s), \tag{2.36}$$

where

$$\omega_s=2\pi/N \tag{2.37}$$

Combining the (2.35) and (2.36), we have

$$X_p(j\omega) = \frac{1}{T}\sum_{k=-\infty}^{+\infty} X(j(\omega - k\omega_s)) \quad (2.38)$$

Figure 6, 7, 8 shows the process of the sampling, they are X(t), P(t), $x_p(t)$ respectively.

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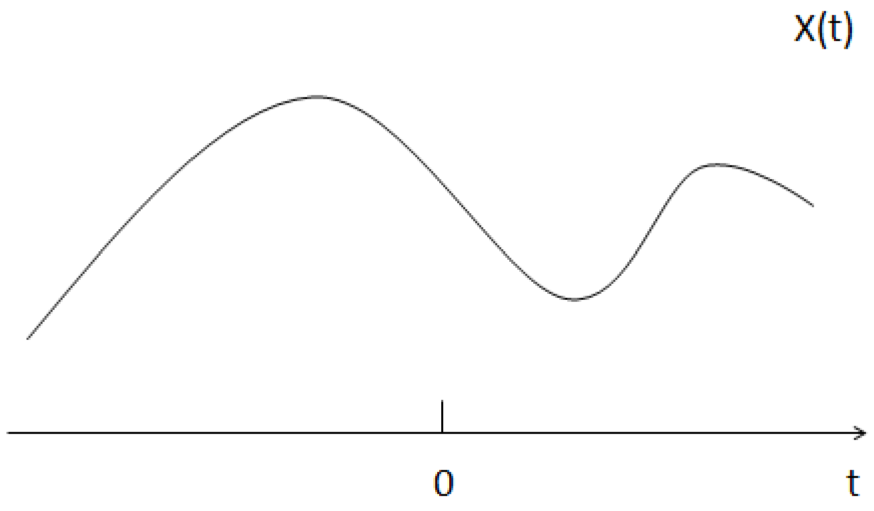

Figure6

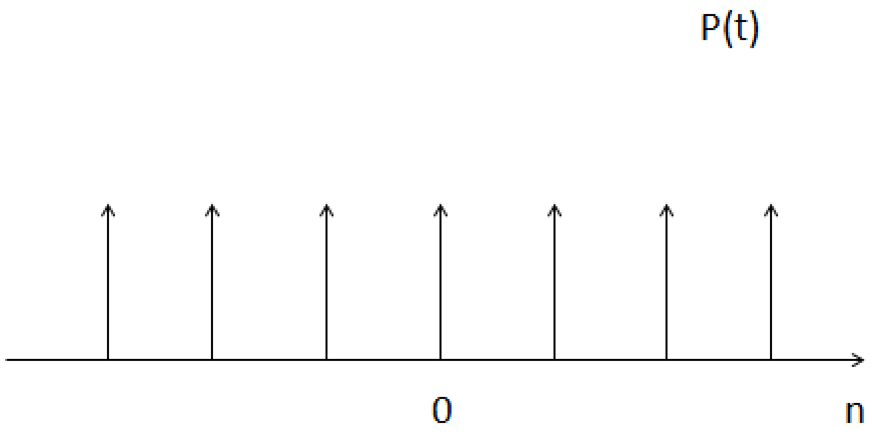

Figure7

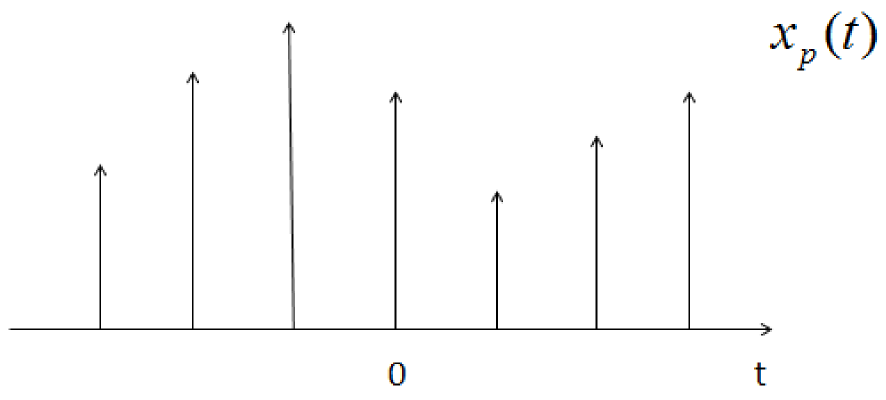

Figure 8

Figure 9 is an original signal. Figure 10 and Figure11 are two possible results when it is reconstructed.

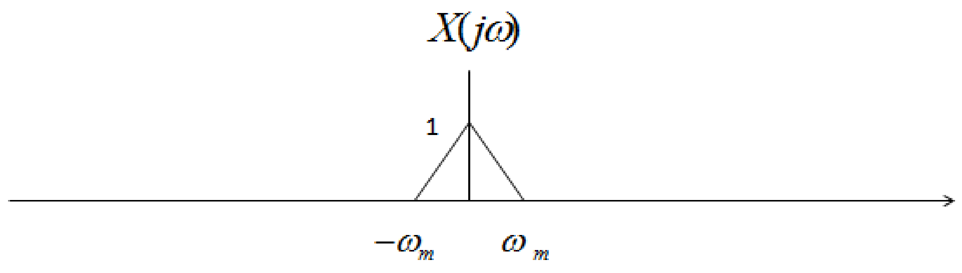

Figure 9

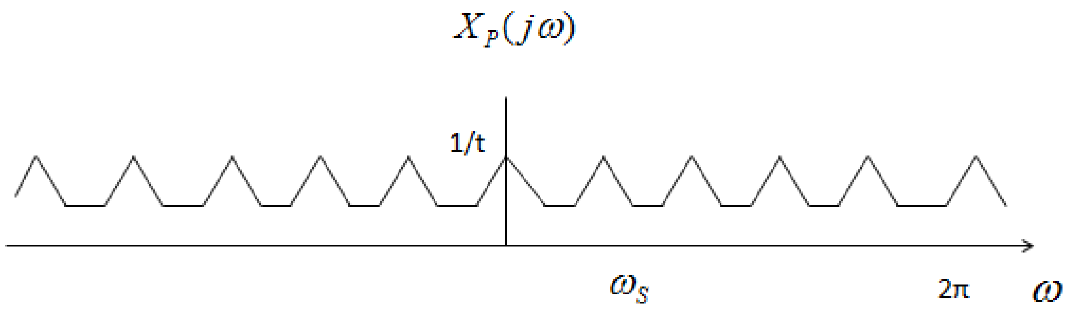

Figure 10

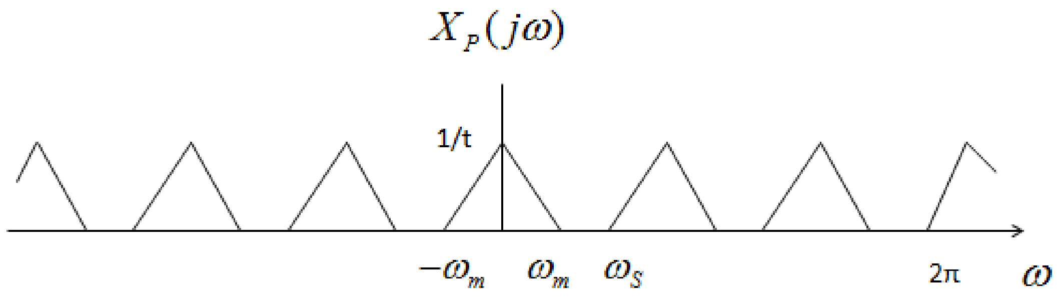

Figure 11

If the sampling frequency is ω_s, it is at least two times higher than the highest frequency of ω_m, ω_m is the original signal, and it is possible to reconstruct the signal from the samples, and this fact is Nyquist's Theorem, the Nyquist frequency is a sampling frequency $\omega_s=2\omega_m$. If the condition is not met, what will happen is that the information of the original signal will be lost. The duplications of the original signal's spectrum can overlap the original one, and the spectrum of original signal will have the bogus signal and some spectrum frequency will be lost, and Figure 10 displays the possible result. Furthermore, it is impossible to reconstruct the signal from its sample. So the phenomenon that the useful spectrum information is replaced by the bogus spectrum information is called aliasing. If the frequency is larger than $2\omega_n$, the signal information will be not lost, such as shown in Figure 11.

Fast Fourier Transform (FFT)

An FFT is a fast and efficient algorithm to compute the DFT. The DFT has a forward DFT and an IDFT as follows:

DFT:

$$X(k) = \sum_{n=0}^{N-1} x(n) W_N^{kn}\ , \qquad 0 \le k \le N\text{-}1; \tag{2.39}$$

IDFT:

$$x(n) = \frac{1}{N} \sum_{n=0}^{N-1} X(k) W_N^{-nk}\ , 0 \le n \le N\text{-}1 \tag{2.40}$$

The DFT computation is inefficient because it does not use the periodicity and symmetry properties of the Twiddle factor W_N . These properties are:

Periodicity property $W_N^{k+N} = W_N^k$ (2.41)

Symmetry property $W_N^{k+N/2} = -W_N^k$ (2.42)

Considering the computation $N = 2^v, W_8^3$ The N-point data FFT can be divided to 2 part N/2 – point data FFT f1(n), f2(n), according to the samples of x(n), these are,

$$\begin{aligned} f_1(n) &= X(2n) \\ f_2(n) &= X(2n+1) \end{aligned} \qquad n= 0,1,.., N/2 -1; \tag{2.43}$$

So the f1(n) and f2(n) are gained decimating x(n) through a factor of 2;

Decimation-in-time algorithm

$$\begin{aligned} X(k) &= \sum_{n=0}^{N-1} x(n) W_N^{kn} \\ &= \sum_{n=0}^{\frac{N}{2}-1} f_1(n) \bullet W_{N/2}^{kn} + \sum_{n=0}^{\frac{N}{2}} f_2(n) \bullet W \\ & F_1(k) + W_N^K F_2(k) \\ &= F_1(k) + W_N^K F_2(k) \end{aligned} \tag{2.44}$$

F1(k) and F2(k) are respectively the N/2 point DFTs of the f1(n) and f2(n).

F1(k) and F2(k) are periodic , and the period is N/2.

So F1(k+N/2)=F1(k), F2(k+N/2)=F2(k). Since the factor $W_N^{k+N/2} = -W_N^k$

$$X(k) = F_1(k) + W_N^k F_2(k)\text{, k=0,1,…,N/2-1} \tag{2.45}$$

$$X(k+\frac{N}{2}) = F_1(k) - W_N^k F_2(k)\text{, k=0,1,…N/2-1} \tag{2.46}$$

it can be found that the computation of F1(k) needs $(N/2)^2$ complex multiplications, and the same for the computation of F2(k). Moreover, computing $W_N{}^k F_2(k)$ needs the N/2 additional complex multiplications. So the total computation consume of X(k) is 2 $(N/2)^2$ +N/2= $N^2/2$ + N/2 complex multiplications. So at first, it can reduce the number of multiplications from N^2 to $N^2/2$ + N/2.

Then by computing N/4 point DFTs, it obtains the N/2 point F1(k) and F2(k)

$$F_1(k) = F\{f_1(2n)\} + W_{N/2}^k F\{f_1(2n+1)\}\text{, k =0, 1…, N/4-1; n= 0, 1,…, N/4-1} \tag{2.47}$$

$$F_1(k+\frac{N}{4}) = F\{f_1(2n)\} - W_{N/2}^k F\{f_1(2n+1)\}\text{, k =0, 1…, N/4-1; n= 0, 1,…, N/4-1} \tag{2.48}$$

$$F_2(k) = F\{f_2(2n)\} + W_{N/2}^k F\{f_2(2n+1)\}\text{, k =0, 1…, N/4-1; n= 0, 1,…, N/4-1} \tag{2.49}$$

$$F_2(k+\frac{N}{4}) = F\{f_2(2n)\} - W_{N/2}^k F\{f_2(2n+1)\}\text{, k =0, 1…, N/4-1; n= 0, 1,…, N/4-1} \tag{2.50}$$

The decimation of the sequence is repeated again and again, and the sequences can be reduced to one point sequences. The computations of the decimation are performed v= $V = \log_2 N$, when N= 2^N. Thus, the total computation of complex multiplications for the sequence is reduced to $(N/2)\log_2 N$.The computation of complex additions is $N\log_2 N$.

The example is N=8, gets the computation of N=8 DFTs. It can be observed that the process is performed through tree stages, the first step is to compute four two-point DFTs, the second step is to compute two four-point DFTS, then one eight-point DFT.

Figure 12 is 3 stages decimation-in-time FFT, and Figure 13 display its details of computation and Figure 14 shows basic computation of butterfly.

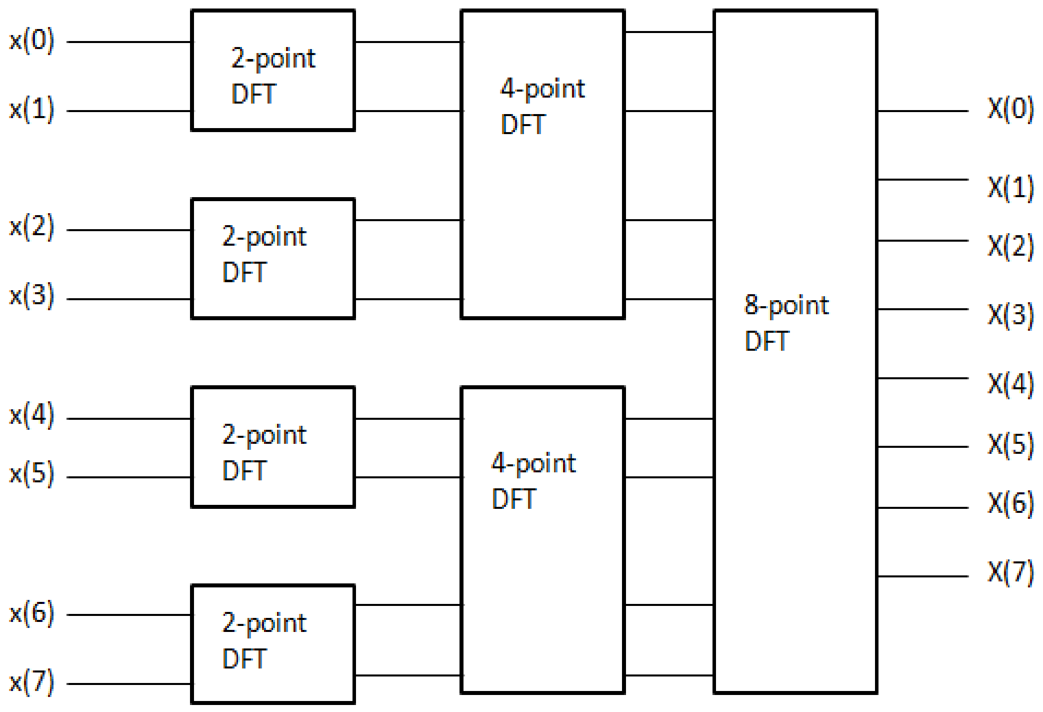

Figure 12 3-stages for 8-point FFT

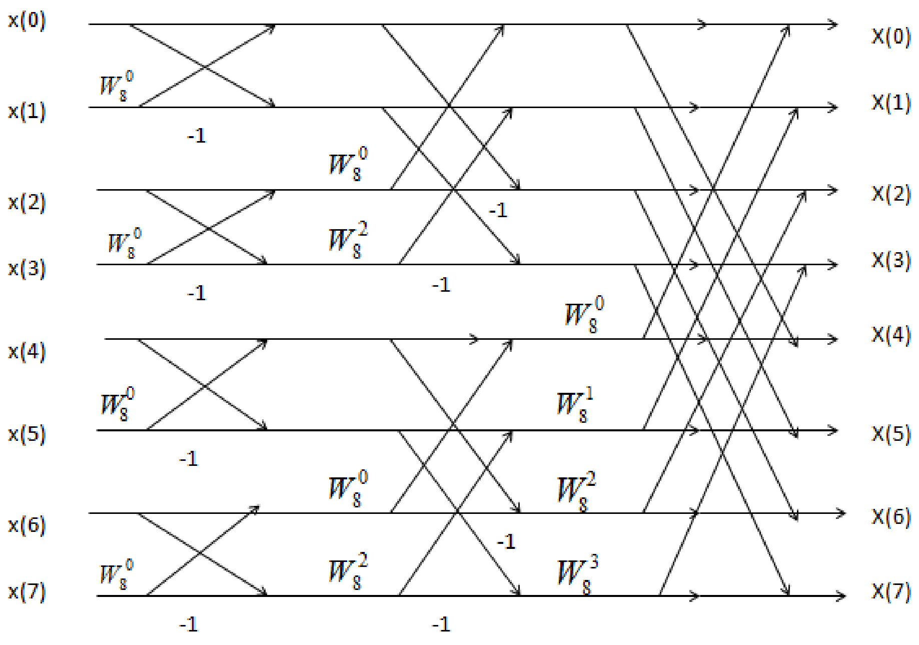

Figure 13 8-point butterfly computation in the algorithm.

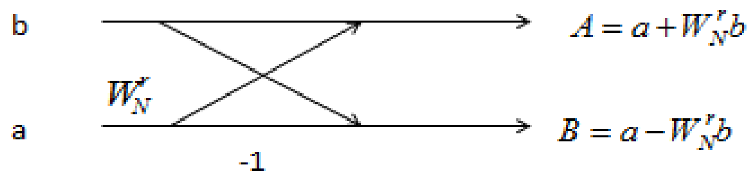

Figure 14 Basic butterfly computation in the decimation-in-time FFT algorithm

Decimation-in-frequency algorithm

Another important algorithm is the decimation-in-frequency algorithm, and it is gained by using the approach of divide-and-conquer. At the beginning, the formula of DFT two parts are spilt, one of which has the first N/2 data points and the second part has the last N/2 data points.

$$X(k) = \sum_{n=0}^{(N\ /\ 2)\ \ 1} x(n)Wn^{kn} + W_N^{Nk/2} \sum_{n=0}^{N\ \ (-\ /\ 2\ )\ \ 1} x(n+\frac{N}{2})Wn^{kn} \qquad (2.51)$$

Since $W_N^{kN/2} = (-1)^k$,

$$W_N^{kN/2} = (-1)^k \qquad (2.52)$$

Then, it can be split to even-numbered samples and odd-numbered samples.

$$X(2k) = \sum_{n=0}^{(N/2)-1} [x(n) + x(n+\frac{N}{2})]W_{N/2}^{kn} \qquad (2.53)$$

$$X(2k+1) = \sum_{n=0}^{(N/2)-1} [x(n) - x(n+\frac{N}{2})]W_n^n W_{N/2}^{kn} \qquad (2.54)$$

If $x_1(n) = x(n) + x(n+\frac{N}{2})$

$$x_2(n) = [x(n) - x(n+\frac{N}{2})]W_N^n$$

$$X(2k) = \sum_{n=0}^{(N/2)-1} x_1(n)W_{N/2}^{kn} \qquad (2.55)$$

$$X(2k+1)=\sum_{n=0}^{(N/2)-1} x_2(n)W_{N/2}^{kn} \tag{2.56}$$

For this algorithm, the FFT needs about $N\log_2 N$ complex additions and $(N/2)\log_2 N$ complex multiplications. The computation can be repeated by the decimation of two N/2-point X(2k) and X(2k+1), and it needs v=$\log_2 N$ stages to decimate, and each stage has N/2 butterflies.

Figure 15, 16, 17 displays the decimation-in-frequency FFT algorithm.

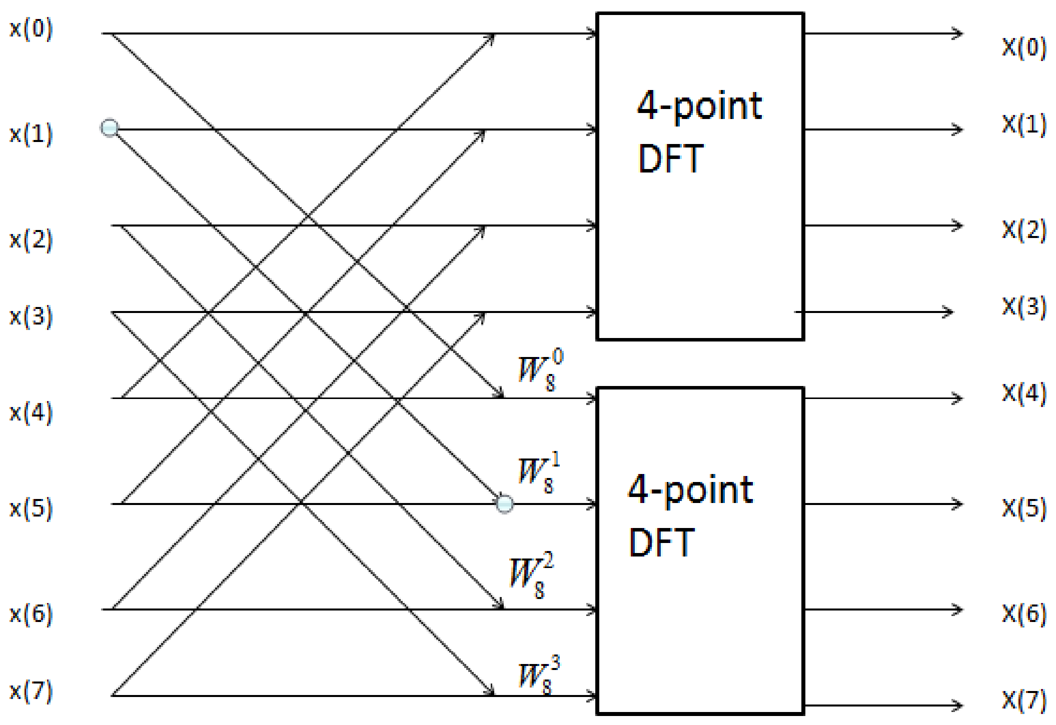

Figure 15 the first stage for the 8-piont the decimation-in-frequency FFT algorithm.

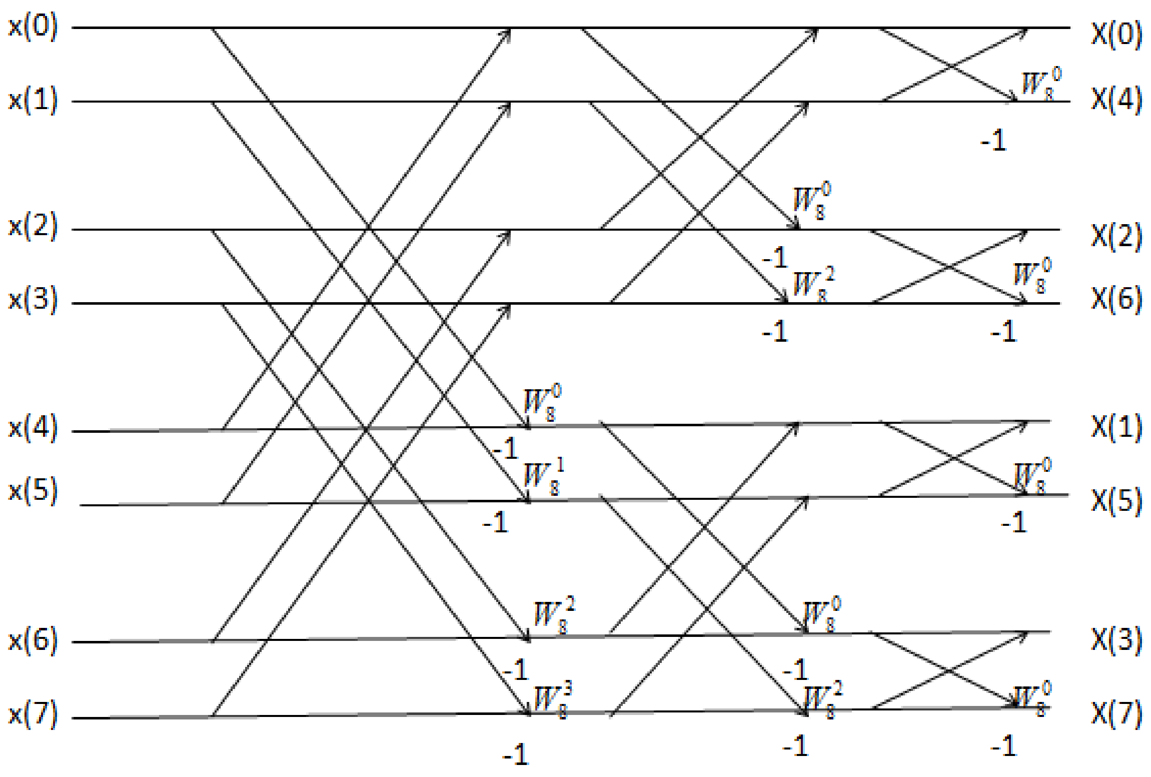

Figure16 8-piont the decimation-in-frequency FFT algorithm.

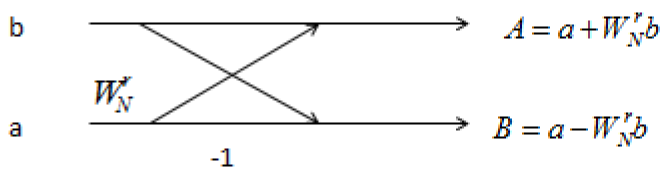

Figure 17 Basic computation in the decimation-in-time FFT algorithm

Chapter 3 The tools used for design.

Section 1 Microcontroller

Introduction to Microcontroller

A microcontroller is an integrated circuit similar to a small computer, containing memory, a processor core and input/output peripherals. In contrast to microprocessors designed for personal computers, they are often used for embedded applications.

Moreover, microcontrollers sometimes are designed into automatically controlled devices and products, such as office machines, engine control systems, and power tools. To make it lower cost to more devices and microcontrollers, the size and the calculation of the speed of the microcontroller are considered more. One type is designed for 4-bit words and has clock frequencies of 4 KHz. It has low power consumption and is used for retaining functionality when waiting for interrupts. The other type is for performance-critical roles, like digital signal processing. A microprocessor is the core of the microcontroller.

The first microprocessor was developed by Intel in 1971. it was a general purpose 4-bit Intel 4004 microprocessor, but because of the system cost, it was impossible to get the external chips to make it become a working system. Since the early 1970s, the capacity of microprocessors has increased a lot and it has followed Moore's law. During the 1990s, erasable and programmable ROM for microcontrollers, such as flash memory, appeared and occupied in the electronics market. With the help of electrical signals, this kind of microcontrollers can be programmed, erased and reprogrammed. Nowadays, besides general purpose microcontrollers, unique microcontrollers have started to be used in a variety of areas, such as automotive, communications and lighting, and some microcontrollers, like the PIC and the AVR, are becoming smaller, sleeker and more powerful. Microcontroller cost plummeted a lot, and some inexpensive 32-bit microcontrollers are $1 USD, and some cheap 8 bit microcontrollers are $0.25 USD.

Introduction to the Stm32f407

The Stm32 is a type of 32-bit microcontroller integrated circuit made by STMicroelectronics. The microprocessors of this family are based around a 32-bit ARM processor core, like the Cortex-M4F, the Cortex-M0，and the Cortex-M3. Internally, each microcontroller has certain parts, the processor core, flash memory, static RAM memory, a debugging interface, and various peripherals. The Stm32 F4-series is the STM32 microcontroller family that uses an ARM Cortex-M4 core. They also have the advantage that they support DSP and floating point instructions. It has a 64K CCM static RAM, a full duplex I2S, faster ADCs, and an improved real-time clock.

The Stm32f407, as shown in Figure 17, is designed for industrial and medical applications which require a high level of performance and integration.

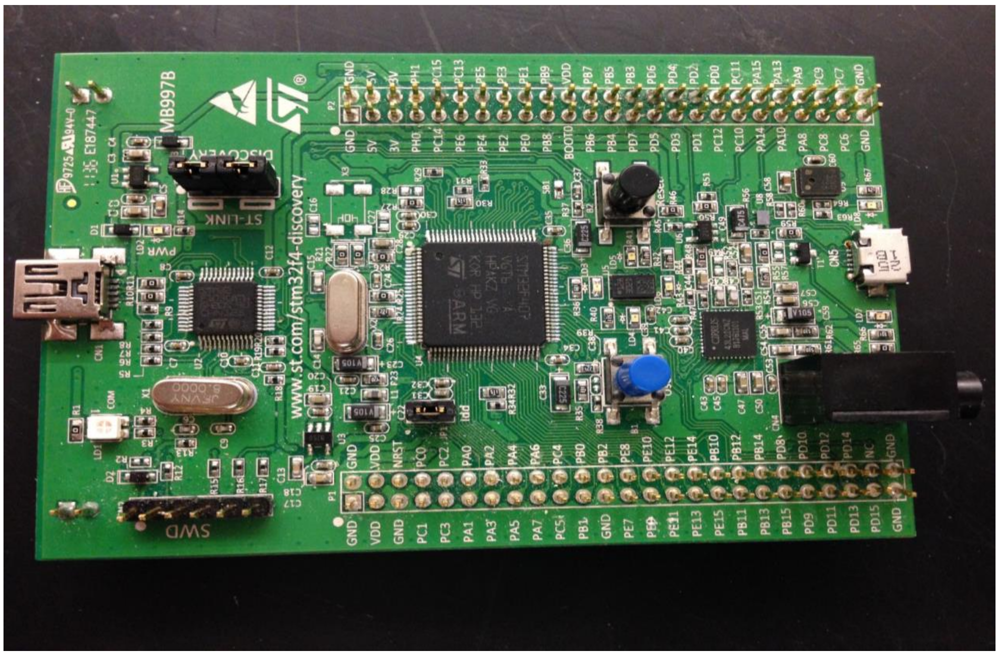

Figure 17 Stm32f407

The microprocessor of this microcontroller is the Cortex-M4 core running at 168 MHZ. At 168MHZ, with 0-waite states of the ST's ART Accelerator, it sends 210 DMIPS/566 Core Mark performances by Flash memory. Furthermore, the current consumption can be as low as 238µA/MHZ.

The connectivity of the STM32F407 is rich, since it can connect a CMOS camera sensor via an 8 to 14-bit parallel camera interface and can feature Ethernet with IEEE support.

It has 2 USB OTG; one of these is supported by HS. It has 15 communication interfaces, 6x USARTs, 3x SPI, 3x I2C, 2x CAN and SDIO). For the timers, it has 17 timers which are 16 and 32-bit having the ability to run at 168MHZ. Furthermore it uses a flexible static memory controller which can support SRAM, PSRAM, Compact Flash, NOR and NAND

Memories, and the memory range is extendable.

The STM32F407 product line has192 Kbytes of SRAM, from 512Kbytes to 1 M Byte of Flash and from 100 to 176 pins in the packages.

The architecture of the STM32F407

The 32-bit multilayer AHB bus is used in the main system, as shown in Figure 18, and it has 8 masters; they are the Cortex-M4 with an FPU core D-bus, I-bus and S-bus, DMA 1 memory bus, DMA2 memory bus, DMA2 peripheral bus, Ethernet DMA bus and USB OTG HS DMA bus. Furthermore, it has 7 slaves; they are the Internal Flash memory ICode bus, the Internal Flash memory DCode bus, the Main internal SRAM1 (112KB), the Auxiliary internal SRAM2(16KB), the AHB1 peripherals, the AHB2 peripherals and the FSMC.

Embedded Flash memory interface

For the flash memory interface of Stm32f407, as shown in Figure 19, it manages CPU AHB I-Code and D-CODE connects to the Flash memory. It has program and erase flash memory operations and also has mechanisms of read and write protection. By the help of instruction prefetch and cache lines, it can improve code execution.

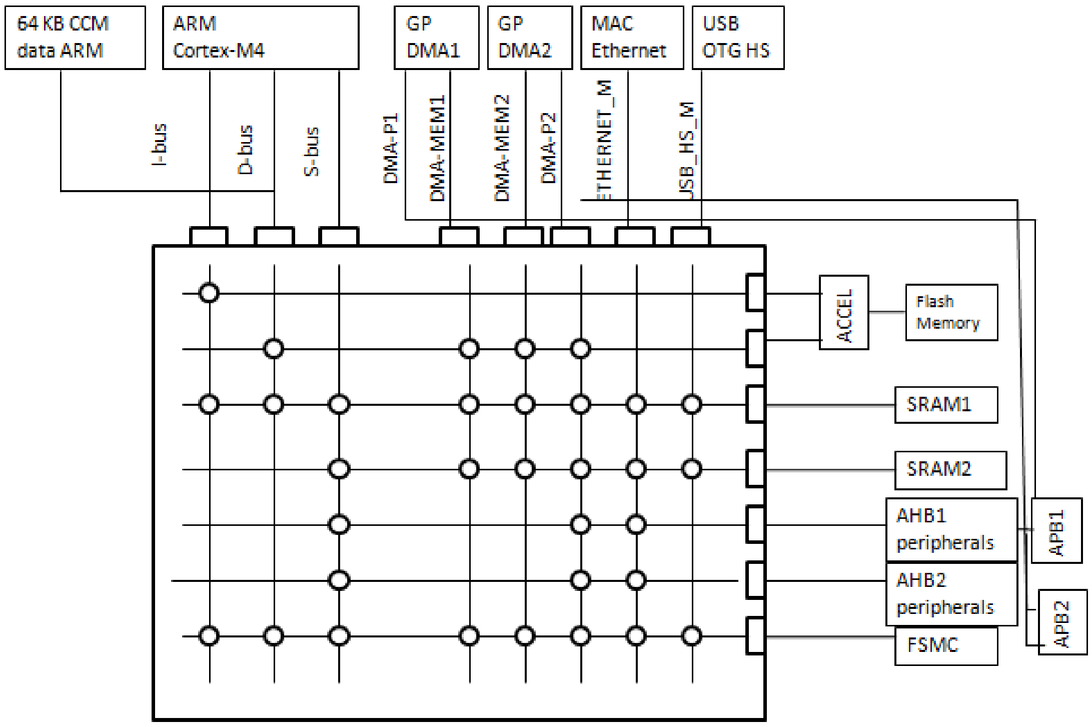

Figure 18 System architecture for STM32F407

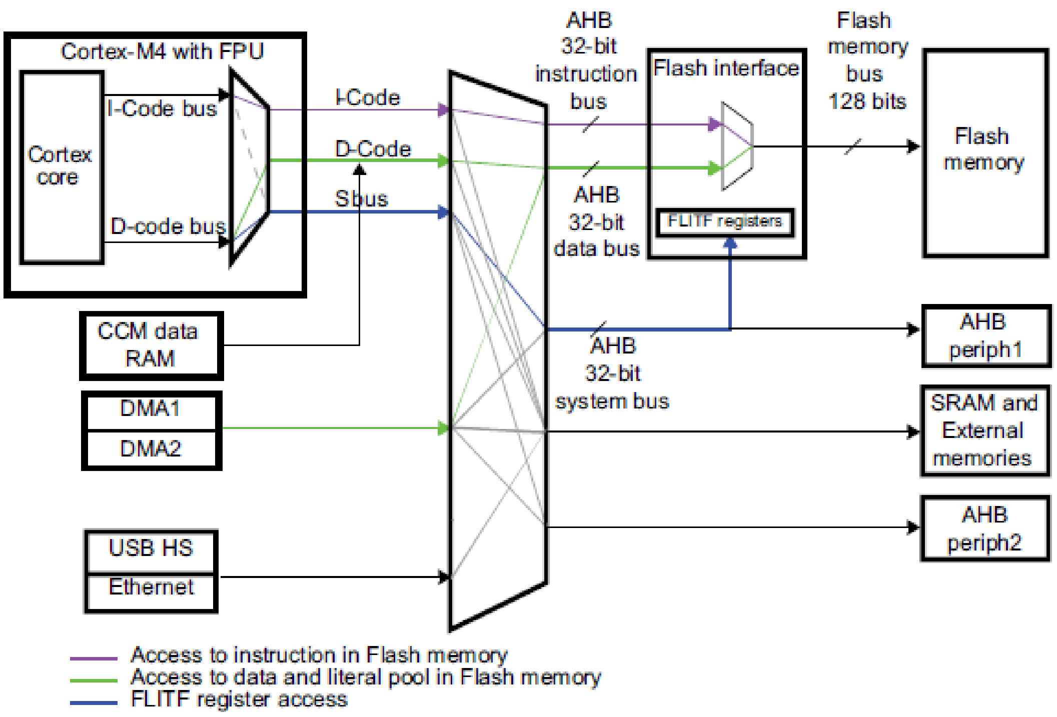

Figure 19 Flash memory interface connection inside system architecture

ADC of STM32F407

The function of the ADC is to successive approximation convert analog-to-digital, and it is 12 bit in the STM32F407. It has 19 multiplexed channels, 16 of them to measure signals from an external source, 2 of them to measure signals from inside the source and one for the VBAT channel. The channels have 4 conversion modes; they can be run in single, scan, continuous or discontinuous mode. After the conversion, the result of the ADC is stored in the 16-bit data register which is left or right-aligned.

For the ADC, it has an External trigger option and it can be used in both regular and injected conversions.

Multi ADC mode

For the stm32F407, it can use two ADCs or more, the Dual and Triple ADC modes can be used. DMA has three modes to request in ADC.

DMA mode1: the ADC-converted data as a half-word is transferred on each DMA request.

As an example: on the first request, ADC 1 data is transferred, and on the second request ADC2 data is transferred and so on.

In DMA mode 2; two ADC-converted data items can be seen as two half-words and they can be transferred as a word.

As an example: in Dual ADC mode, on the first request both ADC1 and ADC2 data can be transferred.

DMA mode 3: it is similar to DMA mode2; the difference in this type of DMA requests is that two ADC converted data packets as two bytes are transferred.

Timers of the STM32F407

TIM2 to TIM5: general purpose timers.

These two timers are made by a 16-bit auto reload counter that is driven by a programmable prescaler. They can be used for many purposes, like input capture, output compare, PWM generation, and one-pulse mode output. By the timer prescaler and controller prescaler, the waveform periods and pulse lengths can be modulated in several milliseconds at most.

Tim1&Tim8: Advanced-control timers.

These have almost the same function as a general purpose timer. Besides these functions, they have some advanced function. They can use external signals to the control timers and to interconnect several timers by a synchronization circuit.

They have a repetition counter to update the registers of the timer after the counter gives the number of cycles.

They can put the timer's output into a reset state or into a known state by breaking the input.

TIM6&TIM7 Basic timers

They are the timers that have a 16-bit auto reload upcounter and a 16-bit programmable prescaler.

For this purpose, they can be used for a time-base as generic timers. Furthermore, it can be used for a DAC; they can be used in the procedure of digital-to-analog conversion. They can drive this conversion by their trigger outputs.

Section 2 GUI

A GUI is a family of user interfaces that allow the users to easily communicate with electronic devices by the visual indicators and graphical icons. Usability is a design discipline for the GUI to make the design of a stored program efficient and easy of use. User-centered design is the discipline to make the visual language fit for the task well.

The visual widgets are manipulated by the user to make the interactions appropriate for the data they hold. The widgets can be chosen to support the actions when the user interacts with the information. The GUI is customized, because it has a flexible structure so that the interface is independent and is not connected to the application directly. The user can easily change the interface and get what they want. It has some applications, such as airline self-ticketing and check-in, automated teller machines, self-service checkouts in a retail store and the information kiosks in a public space.

Section3 VCP

USB

Universal Serial Bus (USB) is designed to make different peripherals be connected by a standardized interface. It provides a fast, expandable, low-cost, hot-pluggable, bi-directional hardware interface where the computer users can easily plug many peripheral devices to the USB and make the devices automatically configure. The USB can make the user connect a big range of peripheral devices by a single connector type, such as, mice, keyboards, printers, mass storage devices, scanners, telephones, digital still-image cameras, modems, audio devices, video cameras to a computer. USB is not mapped into computer I/O address space and does not use DMA channels and IRQ lines, so it does not consume system resources directly. Memory buffers of the USB system software are the only system resources consumed by a USB system.

VCP

COM ports provide an easy way for embedded systems and PCs to exchange information. The traditional way for a COM port to operate on a PC is by using an RS-232 serial port. Recent PCs sometimes skip the function of RS-232 because of the application of USB. A USB device can be used as a VCP that the application can access using the SerialPort class of .NET's with the right firmware.

When USB is used as a VCP, it is an interface that can give applications access to a USB device like a built-in-serial port. There are many functions of USB VCP that can be used as bridges to convert between RS-232. But sometimes a VCP does not need a serial interface. Some VCPs convert between a parallel interface and USB. Or a VCP just reads and stores data from a port of an on-chip analog to digital converter and sends it to a PC by USB.

Chapter 4 Design principle

From figure 20, the project is made up of 4 steps. They are ADC, a modified FFT algorithm, VCP and a GUI which are designed and connected together to test the transmissibility of a MEMS device.

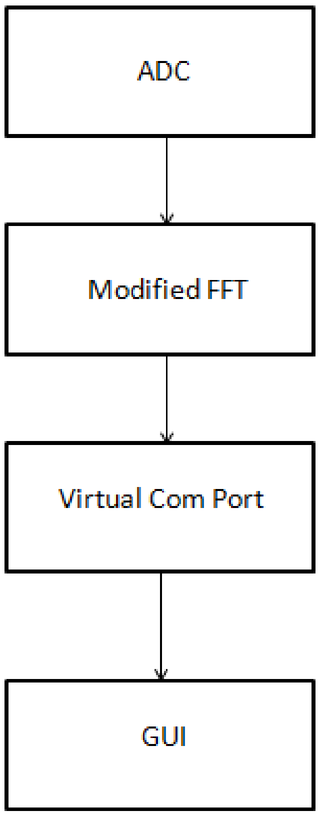

Figure 20 Process for the design

The first step is ADC. ADC will happen when the signal from the MEMS device is transmitted to the Stm32f407. The type of signal is an analog signal. Analog signals cannot be analyzed by the computer, because it has infinite resolution. ADC is a process that can convert continuous signals to digital numbers to represent the signal’s amplitude. Two ADC channels are used to transfer the input and output signals.

The second step is the use of a modified FFT algorithm to transfer the time domain of the signal to the frequency domain. An FFT is an algorithm to calculate the DFT quickly. The DFT is used to convert the function from time domain to frequency domain. Through the algorithm, it can calculate the value of the amplitude of output data and input data and then the ratio of the output to input will be computed. Since the existence of the calculation error and the limitation of the calculation, it will produce noise around the peak and also it will produce white noise over the entire frequency domain, so it will need to improve performance by digital filtering.

The third step is to set VCP. VCP is used to transfer the result from the microcontroller to the personal computer. The microcontroller has the function of the Virtual Com port; it can cause the USB2.0 to perform as a COM port, and it can get higher speed than an ordinary COM port.

The last step is the design of the GUI. After the PC receives the data from the microcontroller, the visual studio is used as a GUI in the project. After the data is transferred by the VCP, It uses Windows Forms to create and design the data display figure. It uses the thread way to simplify the program and to prove the real-time of the figure. At last the final result of the MEMS device will display in the PC correctly. And the transmissibility of the MEMS device will be displayed dynamically.

Figure 21 is the ADC structure of STM32f407.

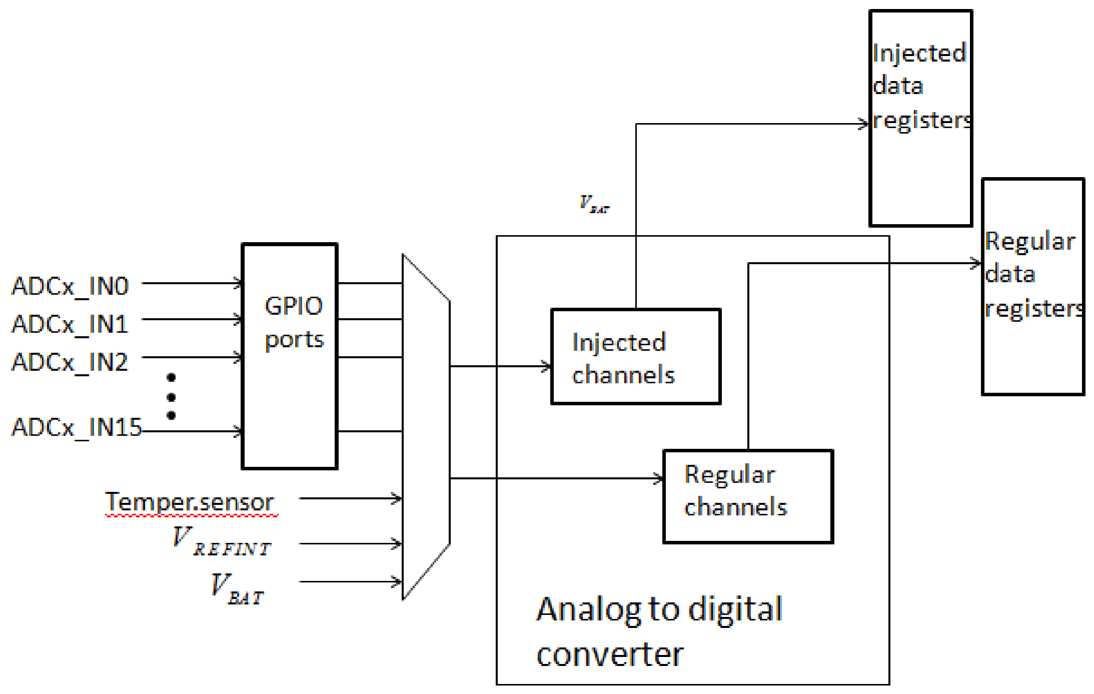

Figure 21 ADC Structure

Section1 The configuration of the ADC

Choosing ADC Channels

The test of transmissibility needs two signals to test and ratio, one is defined as the input signal and the other one is defined as the output signal. For the convenience of the configuration, the process needs the two ADC channels to convert the data at the same time. For the stm32f407, as shown in Figure 21, it has three ADC modes, ADC1, ADC2 and ADC3, and each of them has 16 channels that can transfer the data to the converter.

For the process, two channels of ADC 1 are selected to transfer the data.

Table 1 shows the configuration for each ADC.

	ADC1	ADC2	ADC3
Channel0	PA0	PA0	PA0
Channel1	PA1	PA1	PA1
Channel2	PA2	PA2	PA2
Channel3	PA3	PA3	PA3
Channel4	PA4	PA4	PF6
Channel5	PA5	PA5	PF7
Channel6	PA6	PA6	PF8
Channel7	PA7	PA7	PF9
Channel8	PB0	PB0	PF10
Channel9	PB1	PB1	
Channel10	PC0	PC0	PC0
Channel11	PC1	PC1	PC1
Channel12	PC2	PC2	PC2
Channel13	PC3	PC3	PC3
Channel14	PC4	PC4	
Channel15	PC5	PC5	
Channel16	Temp sensor		
Channel17	V_{REF}		

Table 1 ADC configuration

The mode ADC3 is selected in the process. For the channel selection, channel 12 and Channel 13 are selected and PC2 and PC3 are configured for these two channels, as shown in Table 1.

In the process ADC, DMA plays an important role for the transfer of the data. DMA allows the data transfer between memory and peripherals or memory and memory without the help of CPU action. It can provide higher speed to the microcontroller. The Stm32f407 has two DMA controllers and each of them has 8 streams. Each stream can be used to manage memory access requests from for one or more peripherals. Up to 8 channels can be used for each stream. For the DMA 1, there is no stream to use for the ADC.

And for DMA2 selection:

For the ADC1: Stream0 Channel0, Stream 4 Channel0

For the ADC2: Stream2 Channel1, Stream 3 Channel1

For the ADC3: Stream0 Channel2, Stream 1 Channel2

In the process, the DMA2 stream0 Channel2 is used.

Sampling frequency

Sampling frequency for the ADC is also important and the sampling time is programmable. The ADC samples the voltage of the input as ADCCLK cycles that will be modified. For this ADC process, it needs the sampling time as long as possible and to make the sampling rate as fast as possible, since the figure requires the frequency resolution as small as possible, and the detail of the figure can be observed easier.

So the calculation to get the sampling frequency for the process:

The APB2 bus is used for the process

The clock is 84MHz

The prescaler for the clock is 8,

Tcov= sampling time + 12cycles = 480 +12= 492cycles

So the ADC sampling frequency is 84MHz/8/492=21.3khz.

Moreover, a sampling frequency of 21.3 KHz cannot satisfy the requirements for the frequency resolution. Another way to reduce the sampling rate is needed. The way to reduce the sampling rate is to sample 8192 points each time, and get out one point of eight points to get 1024 for input signal channel, and other one point of eight points for output signal channel. And the sampling rate should be about 2.7 KHz at this condition, because the time to get 1 point becomes 8 times slower than before.

And for the ADC process, it does not use an external trigger, but use the software to trigger the ADC.

Working principle for ADC

For the process, 1024 points data will be used in the FFT one time to get the frequency time, and the ADC needs to sample 8192 points each time. The process uses the multichannel continuous conversion mode. For this mode, the process can configure the sequence of the ADC channels successively, and restarts at the first channel after the last channel conversion. So Channel 12 of ADC3 is configured to transfers one input data at first, after that, channel 13 of ADC3 is configured to transfer one output data, and the process is restarted by this way.

Figure 22 displays the working principle of the Multichannel of ADC. The buffer size for the DMA is 8192, with all the data from the converter stored in it, the odd 16-bits is for the data from Channel 12 and the even 16-bits is for the data from the Channel 13.

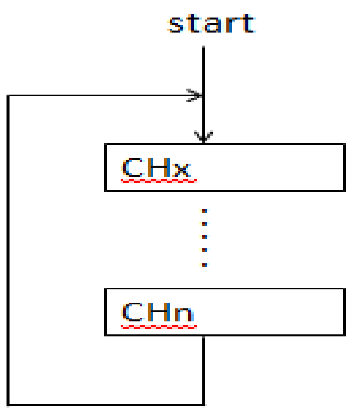

Figure 22 Working Principle for the Multichannel of ADC

The timer design

The Figure 23 displays the flow chart of the timer in the process of ADC.

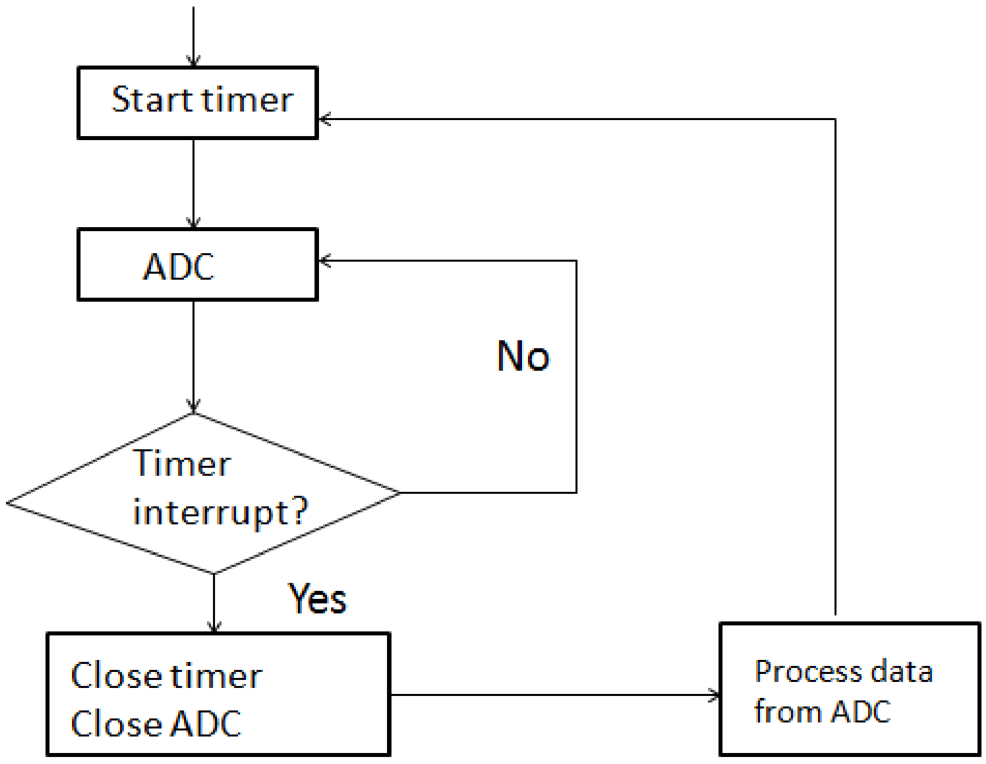

Figure 23 Timer in the process of ADC

For the timer design, the function of the timer is to trigger the interrupt where the FFT algorithm for the design is calculated. The timer used for the trigger in the design is TIM2. In the interrupt, the stm32f407 needs to close the function of ADC to ensure that the ADC does not work during the algorithm calculation. At the end of the interrupt, the function of the ADC will restart. The Timer period for the configuration is 65535 and its prescaler is 4199; so the period time is designed:

((1+TIM_Prescaler)/84M)*(1+TIM_Period)= 4200/84M*65536=3.28s

At each 3.28s, the program will enter into the timer interrupt.

Section2 Modified FFT algorithm

The first part for the Modified FFT algorithm is a 1024 points FFT calculation.

For the stm32f407, the Cortex-M4 core has the ability of FPU single precision, which can support all data types and data processing instructions of ARM single precision; so the microcontroller can be used for complex calculations like the FFT. Furthermore, it has a DSP library which has a rich set of DSP instruction to support complex calculations. For the process, the FFT can be solved by the DSP library of the Stm32f407, and it accomplishes this by way of a Radix-4 Decimation in Frequency algorithm. It is a floating point function process written in the C language. In the FFT process, one FFT is calculated and the other FFT is calculated next. Two 1024 point FFTs require high speed calculation ability from the microcontroller. Without the use of the DSP library, it would be too difficult to finish so complex of a calculation in 3s with the Stm32f407. Since DSP has a strong ability of computation, it can easily finish 1024 points FFT within 4s, and it can fit for the design of the timer. The DSP library can help the microcontroller reach this goal easily and its calculation is lower than 2s actually. The DSP library not only has the function for the FFT, it also has the function for getting the absolute value of the complex number, After the FFT, both ways get a 1024 complex number and all these complex numbers are sent to the DSP function to get the absolute value. And after these steps, the value from the output signal and the value from the input signal in the frequency domain are obtained by the microcontroller.

For the FFT used in this research, it needs to do a 1024 point one time FFT. To make the frequency resolution as small as possible, it samples 8192 points each time and gets out one point from each eight points each time. The frequency resolution in the frequency domain is related to the sampling frequency. The original sampling rate for the ADC is 21.3 kHz. As mentioned before that the sampling rate needs to be as low as possible, the sampling points for each ADC time are 8192 and it is 8 times than the points used for each channel FFT, thus the sampling frequency will be reduced by 8 times. So the sampling frequency for this time should be: 21300/8=2.7KHz and the frequency resolution will be2700Hz/1024=2.6Hz. This frequency resolution is enough to reflect the property of the transmissibility in the frequency domain.

Before, when the magnitude value is sent to the GUI, it will produce two kinds of noise. One kind is from the ADC process; it exists in the total frequency domain by the term of white noise. For this noise, an noise filter can be used to reduce it and it is easy to reduce.

The other kind is from the FFT calculation, since the FFT can't display infinite frequency points in the frequency domain because of the limitation of the calculation, and the energy of the point that does not exist in the domain frequency will affect the point around, it and this kind of noise will exist around the peak. For reducing this noise, an averaging filter is used to cancel the noise and it is designed in part of the GUI,

For the process of transmissibility, it focused more on the amplitude in the frequency domain. But phase shift also can be obtained during the process. The phase angle of the complex number is y=a+bi. A and b can be positive or negative, thus to get the phase angle of y, assume k= arctan(b/a)*180/ π; it has 4 different situations. If a>0, b>0, y=k; if a>0, b<0, y=k; If a<0, b>0, y=k+180; If a<0, b<0, y=k-180. Then, the phase shift is the phase difference between the phase angle of the input and output signals. When the figure is displayed, the change in the phase shift can be displayed dynamically also.

Section3 VCP Design

After downloading the VCP driver for the Stm32f407, USB2.0 can be used as a COM port to exchange the data between the microcontroller and the PC.

With the help of the VCP driver for the STM32, the function of VCP can be started. Actually, the core to operate VCP is similar with that of an ordinary COM port. The result of the calculation of the board is sent to the PC by USB. When the data is received by the PC, the data needs to be read and operated on later. For the transmission of VCP, one data needs to be broken into several bytes, since the unit transferred in the USB is one byte. Furthermore, to reduce the possible errors during the USB transmission or the storage, the checksum for the data that the PC received is important. For this step, after the data is unpacked, the sum of these bytes is gained and the result is appended as extra bytes. After all the bytes have been received by the PC, their sum will be computed again and compared with the sum computed in the microcontroller, and then all the data will be packed in bytes. If the result of the comparison is unequal, the error will be detected. This can prove the stability for the function of VCP. Furthermore, since more than 2100 bytes need to be transferred, the process size of the buffer for the USB transfer to the PC is set 4096 bytes, so it can prove that all of the useful bytes are transferred one time and make the process in real time. The speed of the USB transformation is tested by the software, and is about 200kb/s. It is much higher than the ordinary COM port. The high speed of transmission is a key step to prove the property of real time for the test of transmissibility.

Section 4 Design of GUI

The Visual Studio is used to display the transmissibility figure real time. Windows is the application programming interface to operate the data which can provide access to the Microsoft Windows interface elements through the way of packing the Windows application programming interface in managed code.

There are three parts for the code. They are Program.cs file, Form1.cs file and Form1.Designer.cs file. Program.cs is the starting point for the program and the code Form1 consists of Form1.cs and Form1.Designer.cs file. Form1.Designer.cs contains the code to design the figure which is generated when the components are dragged to the form using toolbox. Meantime, Form1.cs is the place where the logic function is designed by the code.

Figure 24 displays main steps of GUI

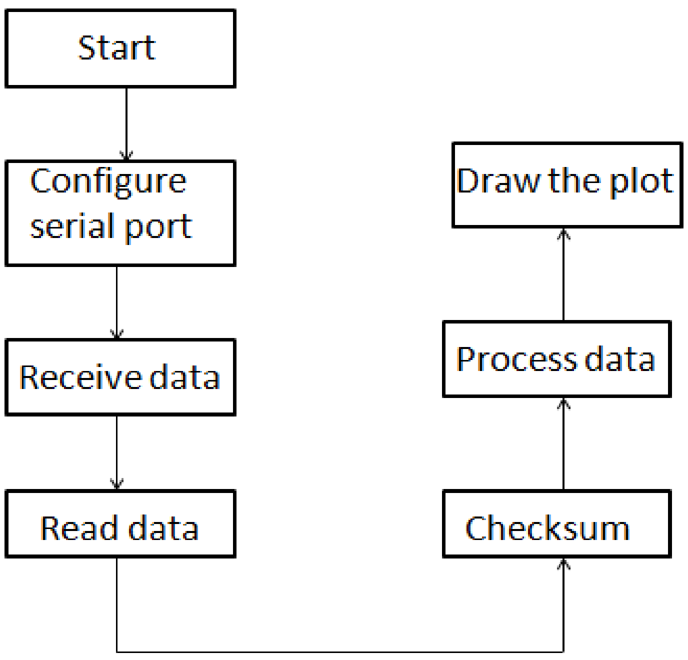

Figure 24 process of GUI

Process of GUI

The code is process oriented, as shown in Figure 24. At first, the serial port is configured. The class SerialPort is used for the data received and read. During the process, the data will be checked by way of checksum and processed using averaging to reduce noise in the figure drawn by the processed data.

Thread to draw the figure

The interface needs to display the frequency domain real time data dynamically and the data size is very big each time. It is possible to be stuck in the process when the interface updates the data. During the process, receiving data by Serialport will create a new thread, function of this.invoke can be seen as a delegate to update the interface in this thread. Threading is the ability to create an application that uses two or more thread executions. Thus one thread for the GUI process is to get the data transferred by the Virtual COM port at first and the second thread is to display the frequency domain real time data after unpacking the data. The function of the this.invoke is to call the method to update the points in the frequency domain.

Design of the Interface

For the Interface, a bitmap is used to draw the picture, as shown in Figure 25. For computer graphics, a bitmap stores a binary image in the domain which is a rectangle and the pixel in an image is either white or black. The bitmap consists of the figure of the magnitude and the figure of the phase shift. The data of the magnitude and the data of the phase shift will be sent to the bitmap, respectively. The GUI has the buttons to operate the autoscale function for the figure of the magnitude. The autoscale function is used to make the peak of the magnitude seen clearly by adjusting the scale of the value. It can be 2 times, 10 times and 25 times larger than the true value so that the figure can be observed easily. In the figure, the coordinate values need to be changed by the values of frequency resolution so that the frequency at the n points can be known. The frequency resolution is 2.6 Hz, obtained from the section of the Modified FFT and it is low enough to reflect the property of the frequency domain. When the frequency resolution is obtained, each interval in the frequency domain in the figure will be 2.6Hz. The first point in the x axis should be 2.6 Hz and the second point should be 5.2 Hz, and so on, and the largest frequency in the frequency domain of the figure for the value can be more than 1000 Hz. For the phase shift, the range of the phase is from -π to π.

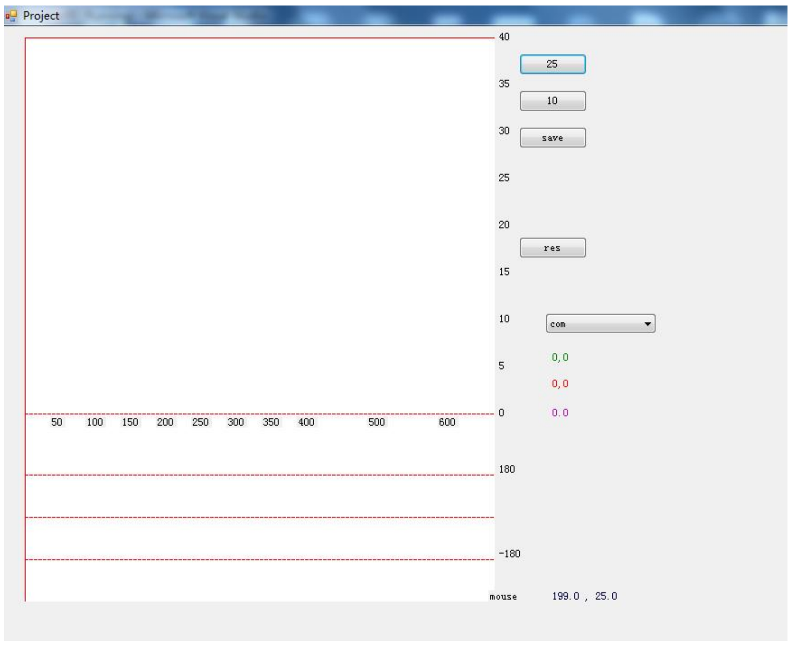

Figure 25 Design of the interface

Writing out the data

The value of the transmissibility in the frequency domain will be written out to a file. The class StreamWriter is used to write out data real time. Every 4 seconds, the data can be written out one time, and the data written out last time can be used to draw a professional figure in MATLAB or used for other work later.

Filtering noise

Furthermore, the GUI uses the averaging filtering to filter the noise of the peak, as shown in Figure 26.

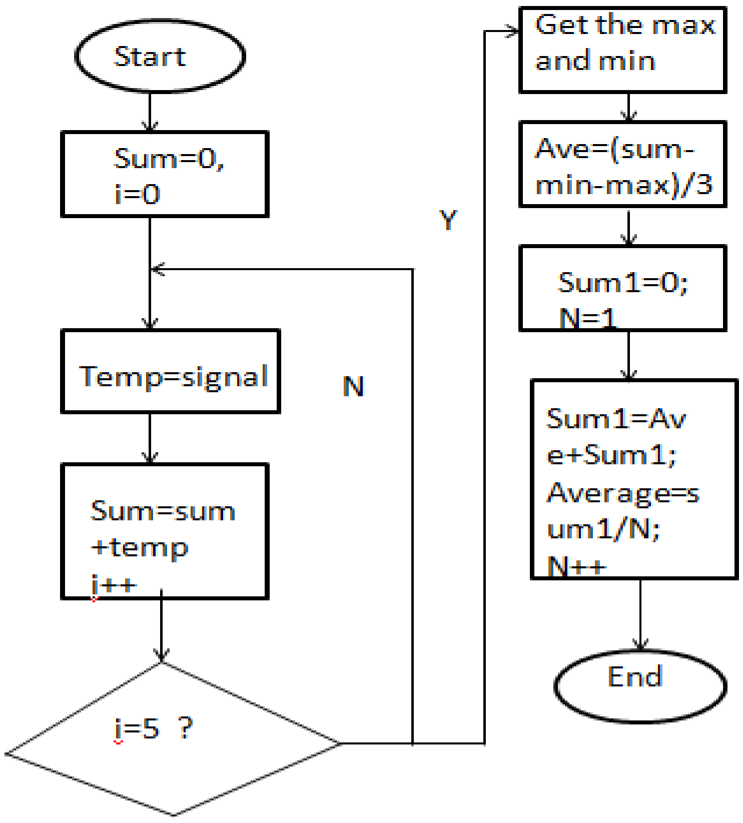

Figure 26 Flow chart of filtering noise

The method is easy to understand. First, after getting five times of data for each point in the frequency domain, the maximum and minimum values of this data will be calculated and canceled, and the average value of the remaining three values will be gotten. When the average of the next five data points is obtained in the same way as before, it will be added to the first average and the new average value can be gotten. Then when the average of the third five data comes in, it will be added to the previous sum, and the new average is divided by three to get the new overall average value. For the N average, it will be added to the previous sum and then the new average value can be gotten, and the previous sum is N times the value of the previous average. The white noise exists randomly, the technique for canceling the maximum and minimum values can eliminate the sudden noise in the frequency domain and averaging can reduce the white noise little by little meantime.

Reset function

In the research, the reset function is also set by the design. If the environment is changed, after the reset button is pressed in the interface, the figure will restart again. The function of reset can help the figure display the change clearly.

Chapter 5 Test

Section 1 Test by function generator

Test the correctness of ADC

To prove the correctness of the ADC configuration, an oscilloscope is used to test it and the DAC process needs to be configured as well. It is the reverse process of ADC, since it can convert a digital data to analog data again. Two ADC channels and two DAC channels are used to recover a certain frequency wave. In the test of the configuration of ADC, only one channel of ADC connects the input signal, which is from a function generator. When 1000 points were sampled, the 50Hz wave can be recovered successfully. It means the function of ADC runs well. Figure 27 displays the function generator and oscilloscope to test ADC and DAC.

Figure 27 Oscilloscope and Function generator.

Test the correctness of FFT

The function generator is also used to check the correctness and stability of the FFT. This time, only one signal is sent by the function generator. The input is a sinusoidal wave which has a certain frequency. So after the signal is processed by the design, besides the DC component, it will appear as a high peak at a certain frequency with energy at other frequencies being relatively low. After the check, the FFT can be used in the next step.

Simulate mechanical system

The function generator system is relatively simple compared with a mechanical system and they will have the same feature with the phenomenon when testing the resonant frequency, since a function generator can simulate the resonant frequency of a mechanical system. If the fault is found during the test with a function generator, it is easy to correct.

Since for the design, one ADC input is for the Output signal and the other ADC input is for the Input signal, two signals need to be produced by the function generator. Because to find the mechanical resonance is to find the ratio of output and input, so for the function generator, the circuit is connected to create the condition that the two signals have the same frequency, but different amplitudes. After these two signals are processed by the design, the peak in a certain frequency should be observed in the figure. To make the signals the same frequency, these signals should be made by the same function generator. By using a circuit, the amplitude of the signal A is 15 times greater than the amplitude of signal B, and they have the same frequency. For an example, if the function generator is set to a certain frequency signal, such as 199 Hz, and the output and input of the stm32f07 will be set to 199 Hz but with different amplitudes.

Figure 28 is the circuit to help to test the correctness of the design using the function generator. There are 15 identical resistors connected in series in the circuit, the signal B is tested after the first resistance and the signal A is tested after the other resistors, and the signal from function generator is set at the same position, so the signal A should be 15 times bigger than Signal B. In the simulation by function generator, signal A can be seen as output signal and signal B can be seen as input signal.

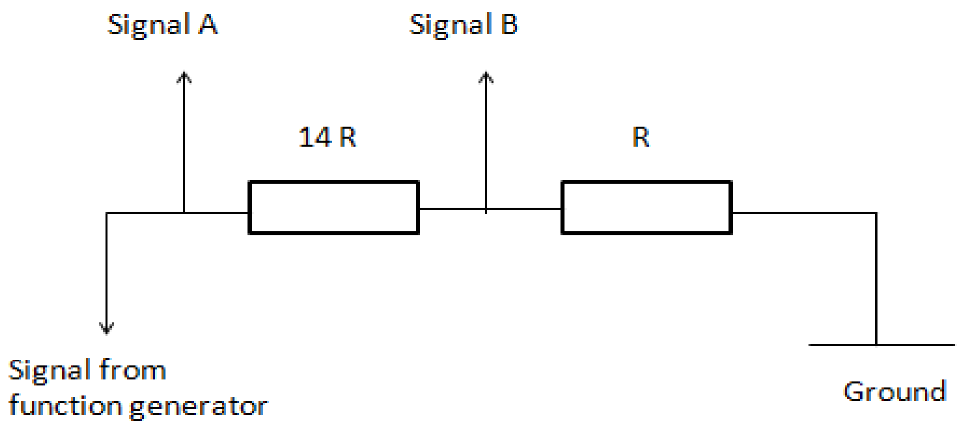

Figure 28 Circuit to verify the accuracy of the design by using function generator

In Figure 29, the peak is at 199 Hz, the amplitude is at 15, but in the other points of the domain frequency, noise clearly exists.

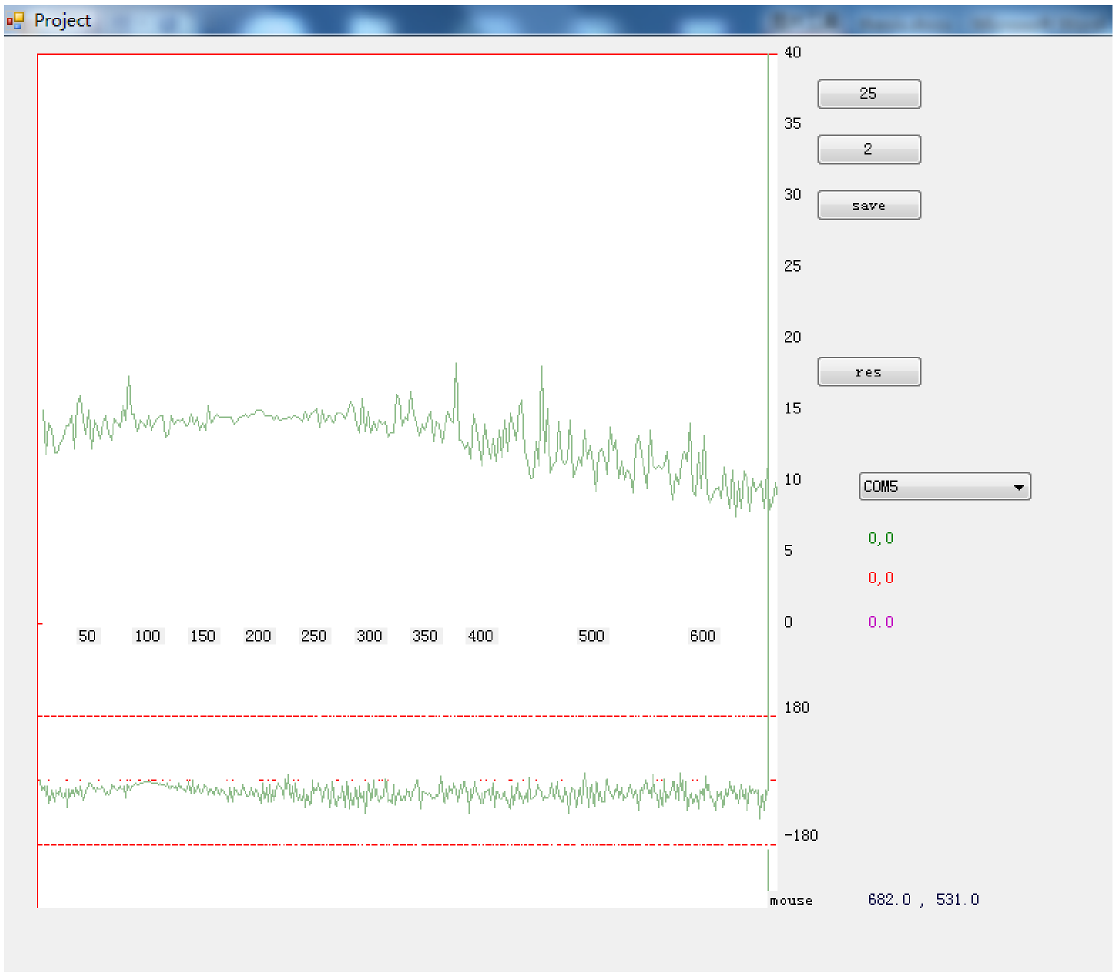

Figure 29 The Original Result

During the test, since the noise around the input information or output information is much smaller than mechanical resonance, but the ratio is close to or a little smaller than the mechanical resonance, so the noise in the transmissibility figure can be eliminated by eliminating the noise in the input information and the output information. If the input value or output value is less than a set threshold, the ratio of the output to input can be set to 1, which means that the input value equals the output value if there is no interference from the noise. This threshold should not be too high to mask the useful information for the transmissibility. Through adopting this thresholding technique, all the noise around mechanical resonance is eliminated. For real mechanical system, the input or output information has no big difference with other spots noise, so the averaging filter introduced in GUI section will make more contribution to the noise filter of the graph.

In the figure 30, the result is good by using the thresholding technique.

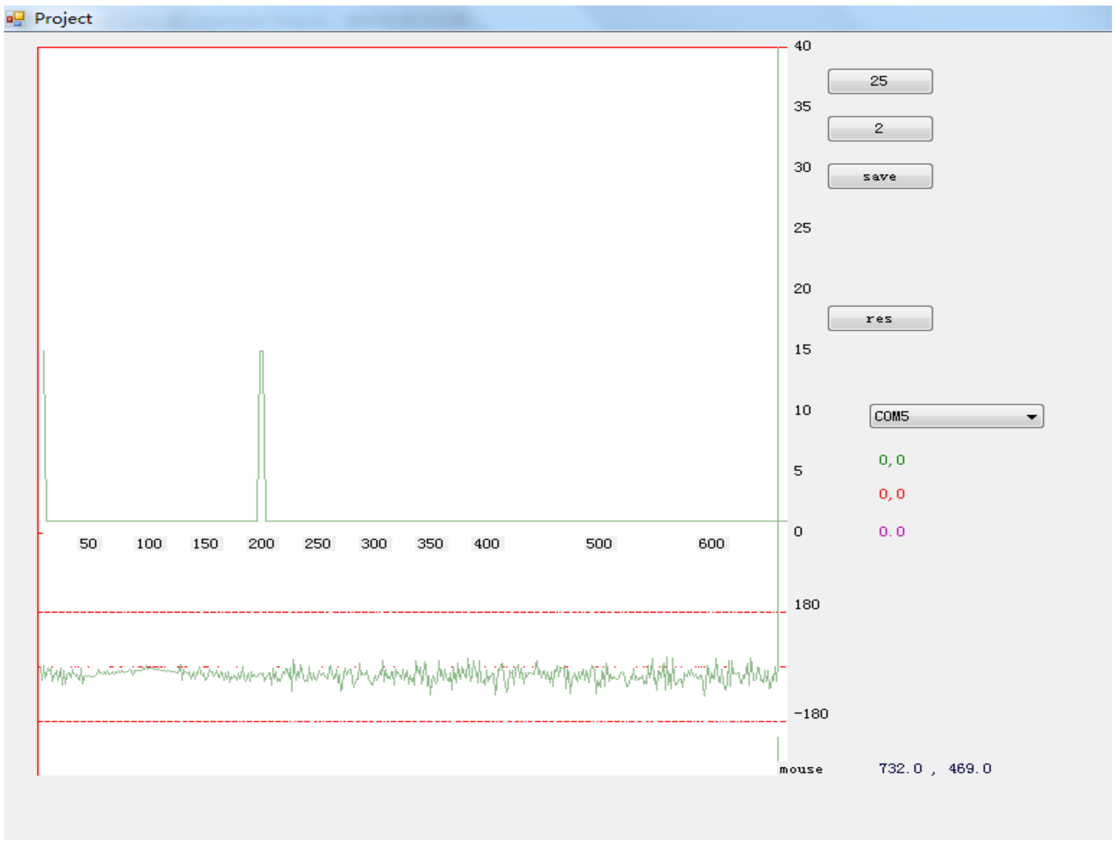

Figure 30 Result with noise filtering

Section 2 Assembling

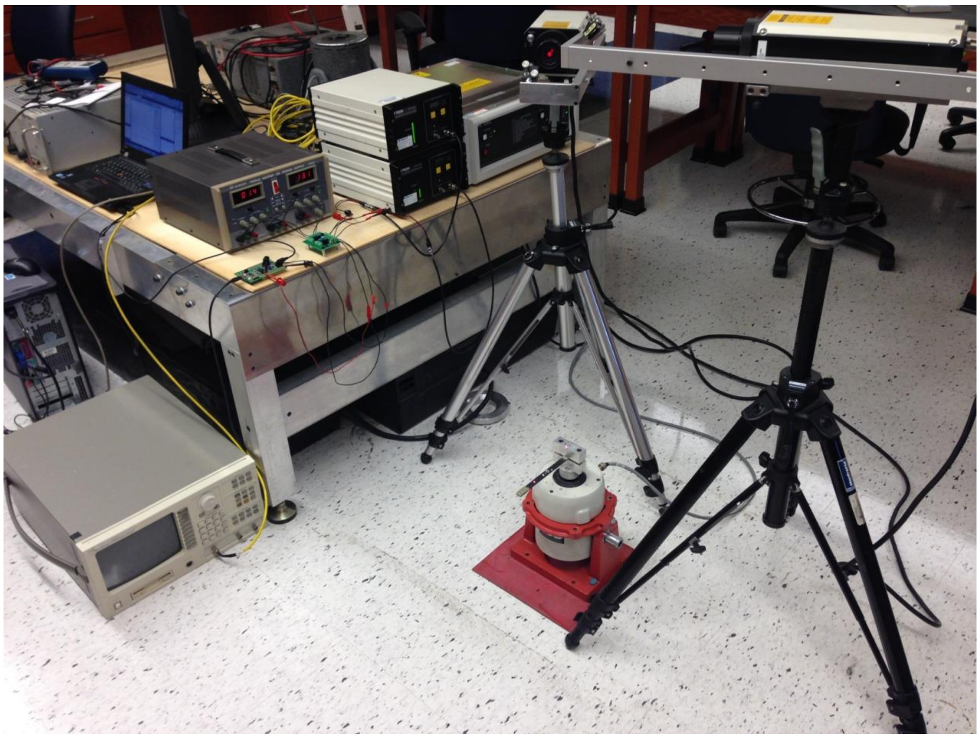

Figure 31 Test system

For the mechanical system test, the core of the design is to test the transmissibility of a MEMS device. Motion transmissibility of three samples is measured in the test. As shown in Figure 31, an analyzer, a laser interferometer displacement measurement system, and an electromechanical shakerare used for testing. An Hp35665A dynamic signal analyzer is used to generate the excitation signals for the electromechanical shaker, such as a sine wave for the shaker, and it can analyze the frequency response. The shaker is connected to shake the sample and excite mechanical resonance. Two Polytec laser vibrometers are used for measuring the displacement of the sample on the vibration system. The laser system needs to measure the output and input displacement. Input is the displacement of the holder, and output is the displacement of the sample. Analyzing the ratio of these two displacements, the transmissibility of the system can be obtained. Then the signal gotten from laser systems will transfer to a circuit to change the range of the voltage from -8V to 8V to 0 to 3V with the help of a power supply, as shown in Figure 35, because it is the range of voltage that can be received by the Stm32f407. Then the new signal is transferred to the Stm32f407to compute the transmissibility to compare it with the results from the analyzer.

Polytec Laser Vibrometer System

The Polytec laser vibrometer is used to measure the displacement in a non-contact way for this experiment, as shown in Figure 32. The fringe counter principle is used for the displacement measurement for the system. The system uses two Polytec OFV 353 lasers which are held by two Manfrotto tripods, and point to the desired locations. Two lasers can measure the displacement of the two different locations respectively .The system also has two controllers for each laser, which provide power to the system and help to convert the laser signal into an electrical signal. The output of each Polytec laser vibrometer system is an analog signal which is proportional to the vibration amplitude in the measured location. Then the signal can be observed on the oscilloscope or sent to a data acquisition system to process, such as the HP 35665A dynamic signal analyzer.

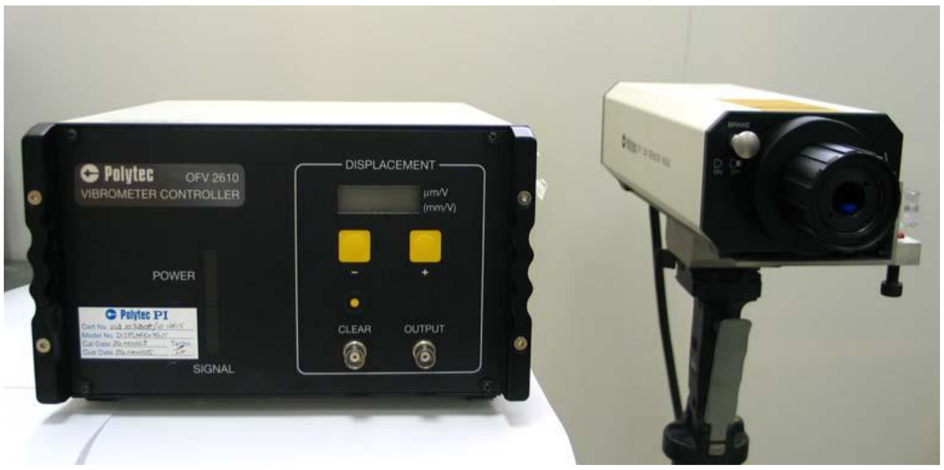

Figure 32 Polytec Laser Vibrometer System

Measurement Range	**Full Scale Output(Peak to Peak)**	**Resolution**	**Max Vibration Frequency**	**Max. Velocity**	**Max. Acceleration**
µm/V	Mm	Mm	kHz	m/s	G
20	0.32	0.08	20	1.6	20.000
80	1.3	0.32	20	1.6	20.000
320	5.2	1.3	20	1.6	20.000
1280	20.5	5	20	1.6	20.000
5120	82	20	20	1.6	20.000

Table 2 Parameters of Ploytec laser Vibrometer System

Electromechanical shaker

The electromechanical shaker used in the experiment consists of a V408 Shaker and a PA 500L Amplifier. Vertical vibration is provided by the shaker and a set of fixtures that are attached on the shaker and are ready for the transmissibility test, as shown in Figure 33. The system also has a vibration control system which can generate time-domain signals containing the required characteristics of frequency-domain and sends the signals to the amplifier. The amplifier, as shown in Figure 34, will make the signal larger and then the shaker will be driven by the powerful amplified signal with a larger voltage and current. After the signal is processed by the controller of the Polytec laser vibrometer, it will be transferred to the frequency-domain in the analyzer by the FFT transform.

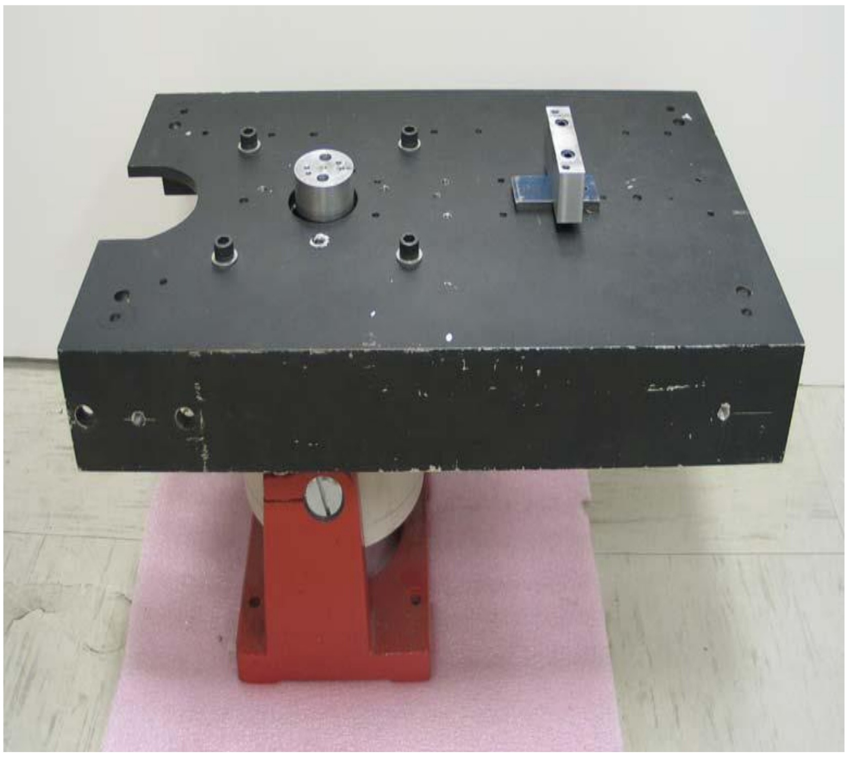

Figure33 V408 Shaker and Fixture Table

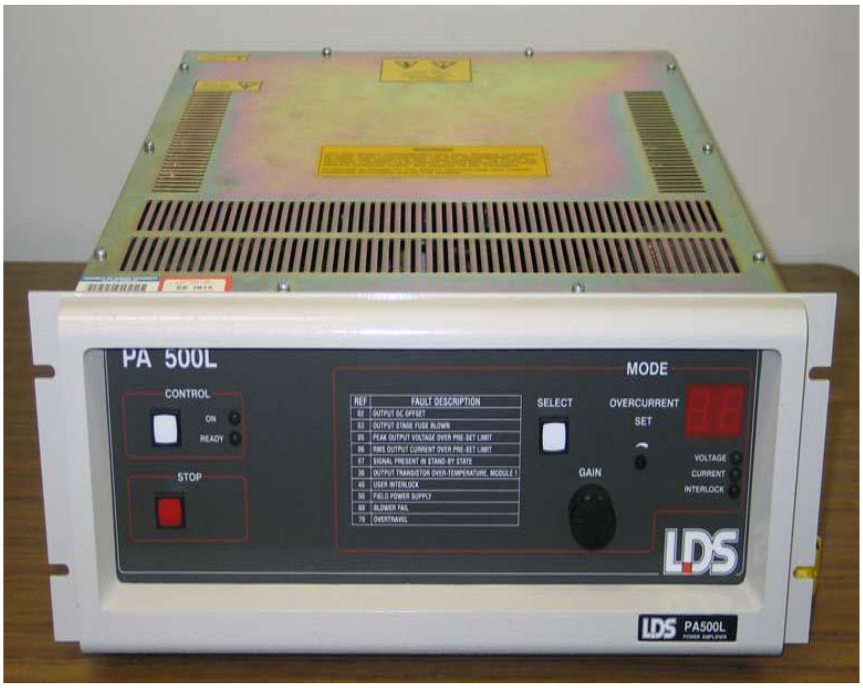

Figure 34 PA500L

Figure 35 and 36 give the detail of the circuit used in the project.

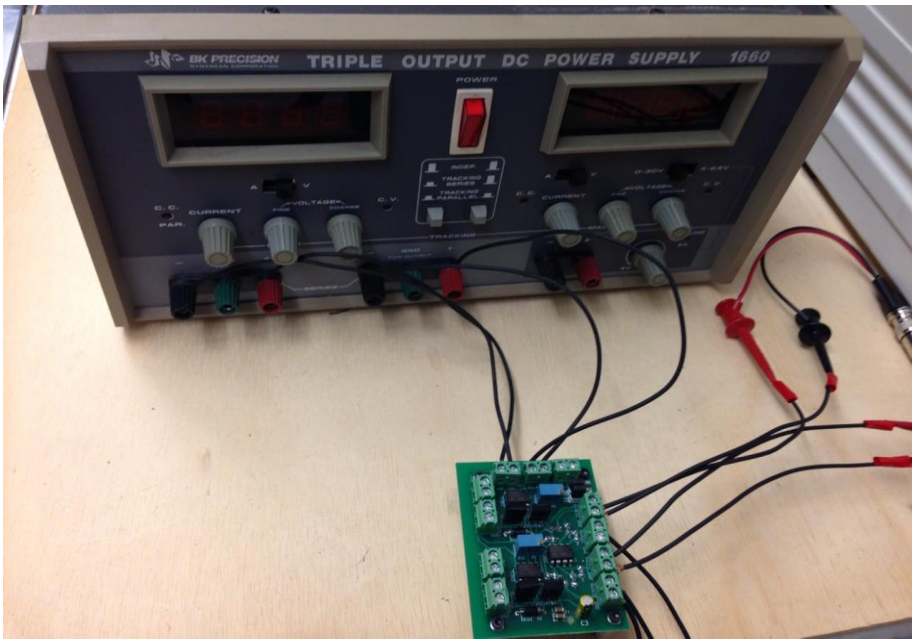

Figure 35 Power supply and circuit to change the range of voltage of input signal.

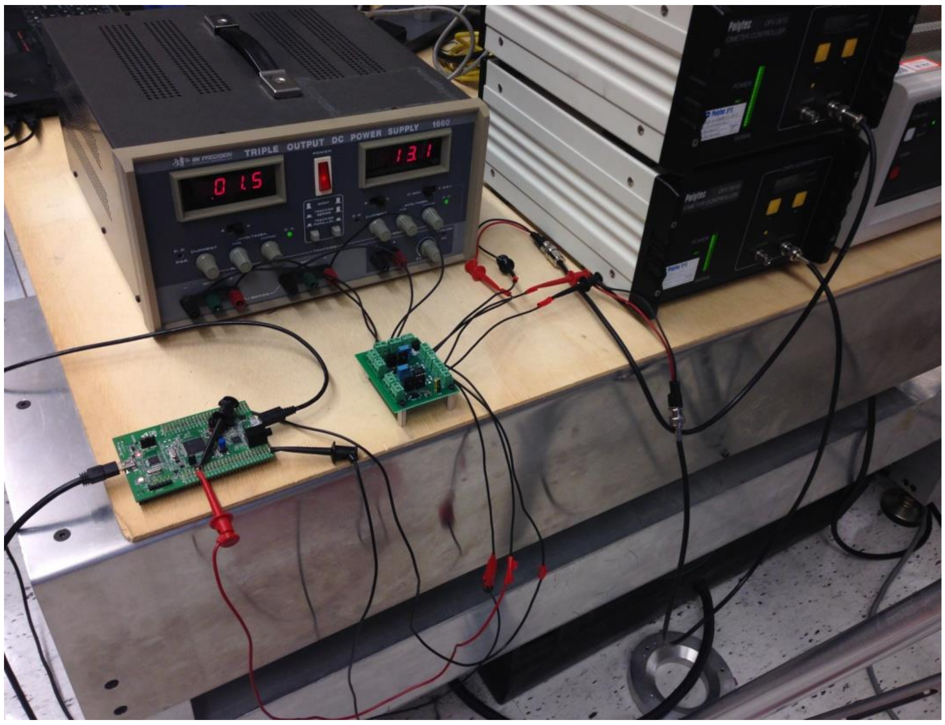

Figure 36 Circuit of design

Section 3 Compared with the results from analyzer

The analyzer is used to produce the transmissibility figure on the PC. Three different samples with different materials and different locations are used to test the correctness of the design, as shown in Figures 39, 40, 41. Furthermore, the results after the FFT are also written to a file by the GUI and used by MATLAB to draw a more professional figure. Then the figures of the analyzer also will be exported and compared with the figures of MATLAB. During the first time, the shaker is not used, but the laser system is used. The peak also appears at 23Hz, as shown in Figure 37. It means the vibration system has its original frequency response without any excitation; the frequency response results are due to the fixture itself. Figure 38 is the figure made by MATLAB. Using MATLAB to get the figure is more professional and can be compared with the figure from the analyzer clearly. Then three different samples are used to test and verify the results of the design. They are put in the fixture and use the laser system to measure the different locations of the sample.

Figure 37 and 38 display the transmissibility of the fixture itself without of the help of the shaker.

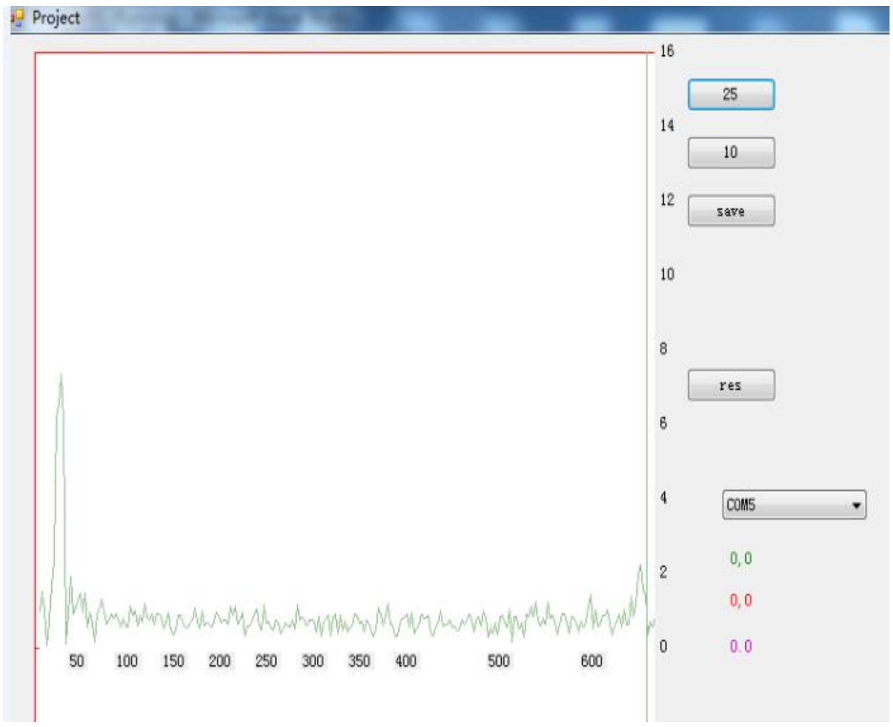

Figure 37 One Peak at 23Hz in my interface.

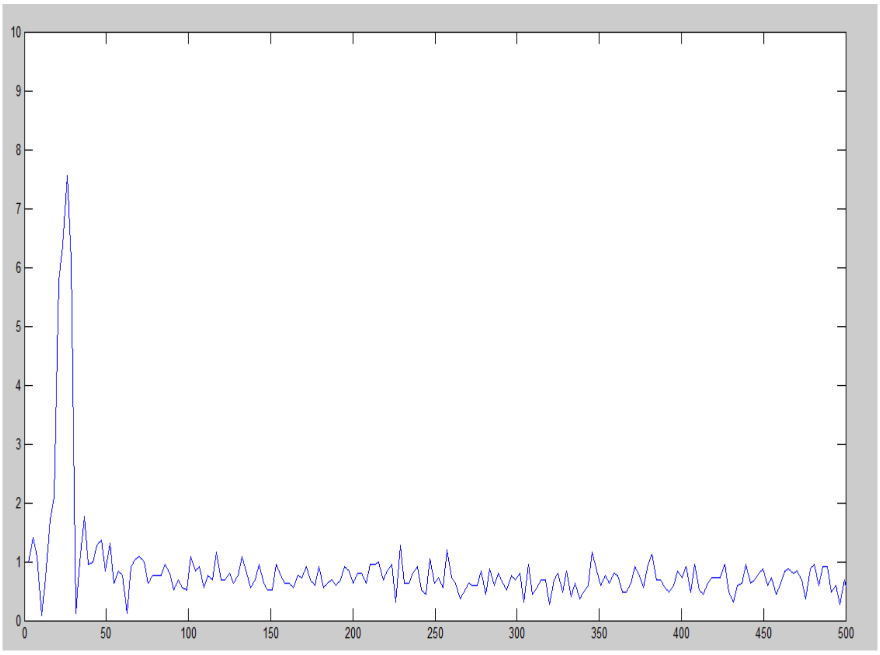

Figure 38 One peak at 23Hz displayed in MATLAB

Figure 39, 40, 41 are the samples measured in the project.

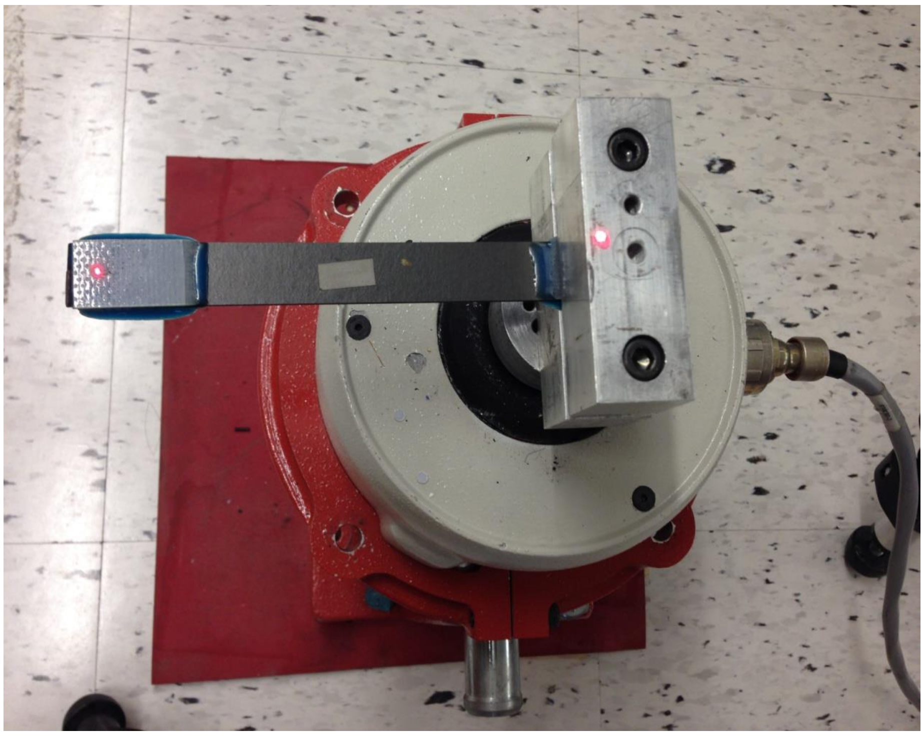

Figure 39 The first sample to test

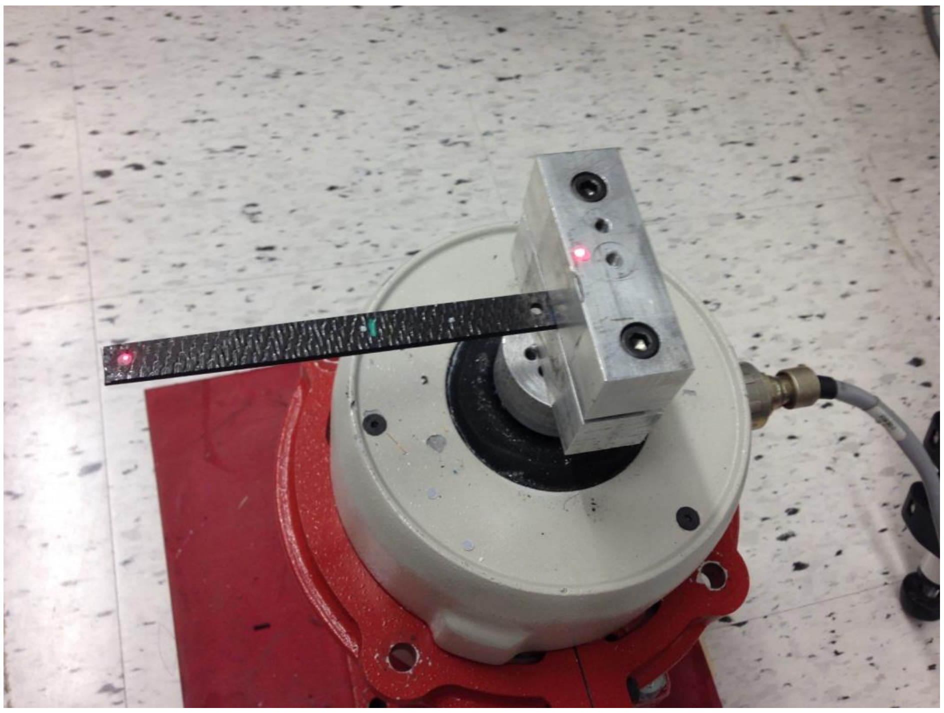

Figure40 The second sample to test

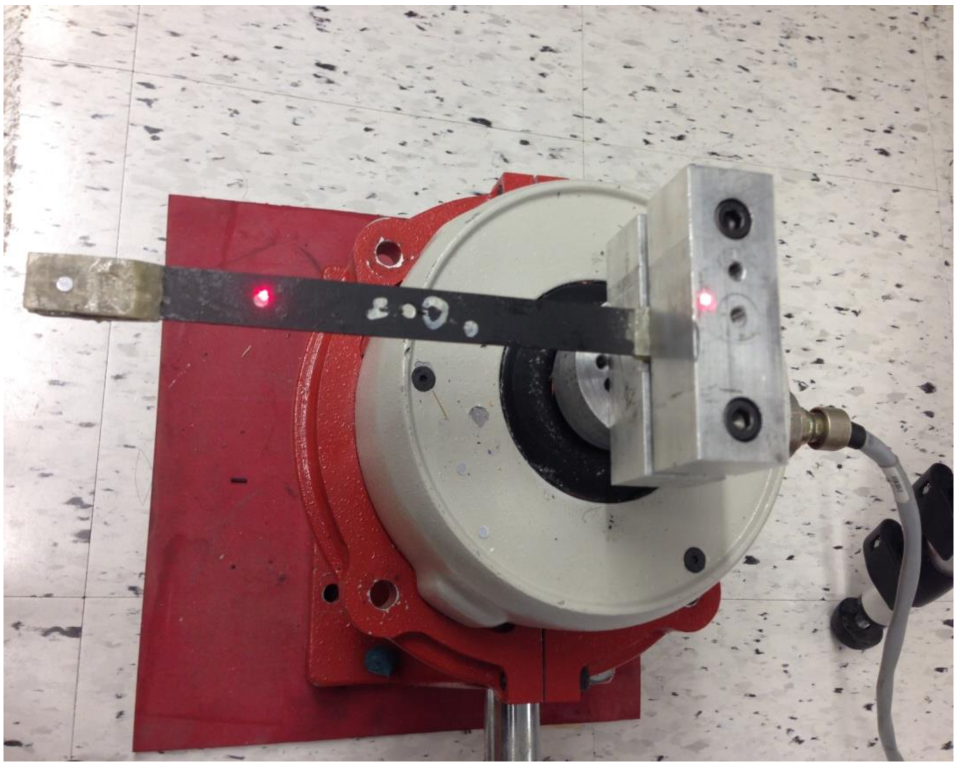

Figure41 The third sample to test

For the first sample, two peaks appear in the figure.

As shown in Figure 42, one peak is at 53 Hz, the other one is at 59 Hz, and the high peak is close to 13 and the other one is close to 8. The shaker is used to excite the sample, and the laser system is used to measure the transmissibility.

For the figure from the analyzer, as shown in Figure 43, it also has two peaks, and the high peak is close to 10, the other one is close to 6.5. And from comparing the two figures, the frequencies of the peaks are the same.

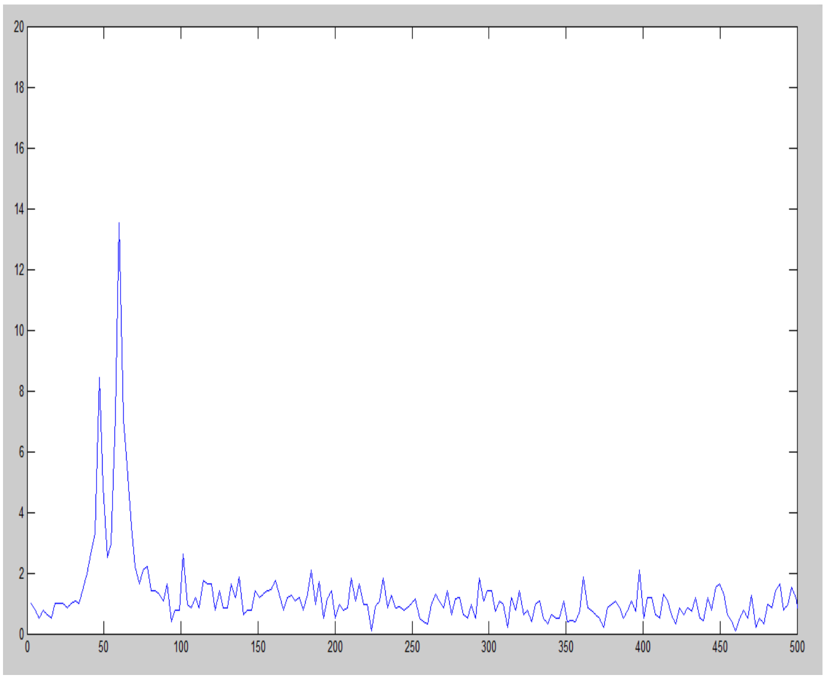

Figure 42 Two peaks at 53Hz and 59Hz displayed in MATLAB through the data of the design

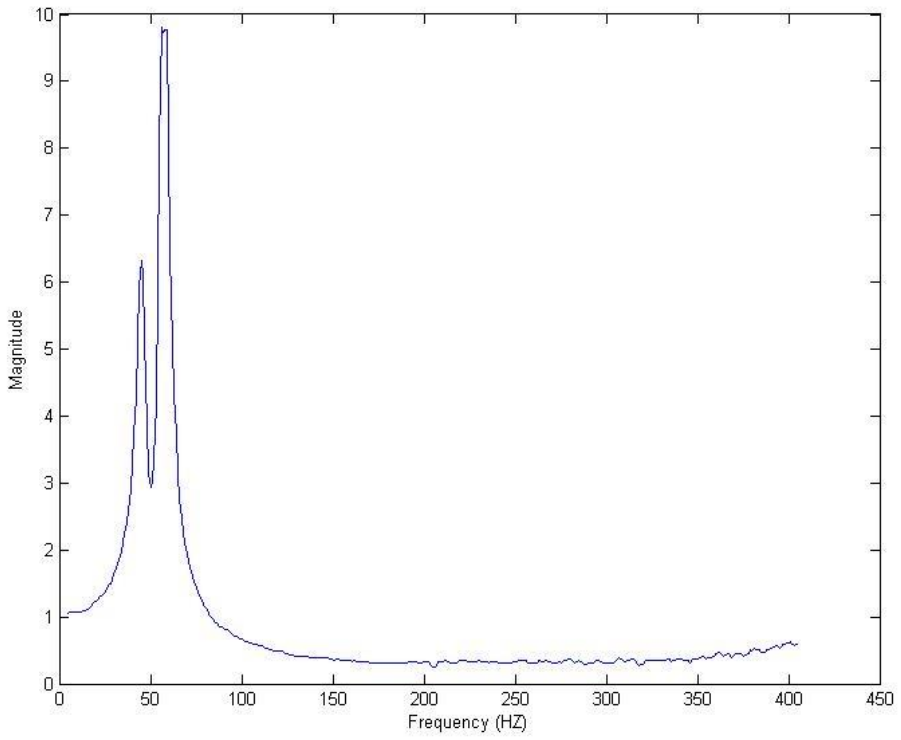

Figure 43 Two peaks at 53Hz and 59Hz from Hp35665 analyzer.

For the second sample, there is peak at 135 Hz and its amplitude is close to 100, as shown in Figure 44. The figure from analyzer also has a peak with an amplitude close to 100 at 135 Hz, as shown in Figure 45.

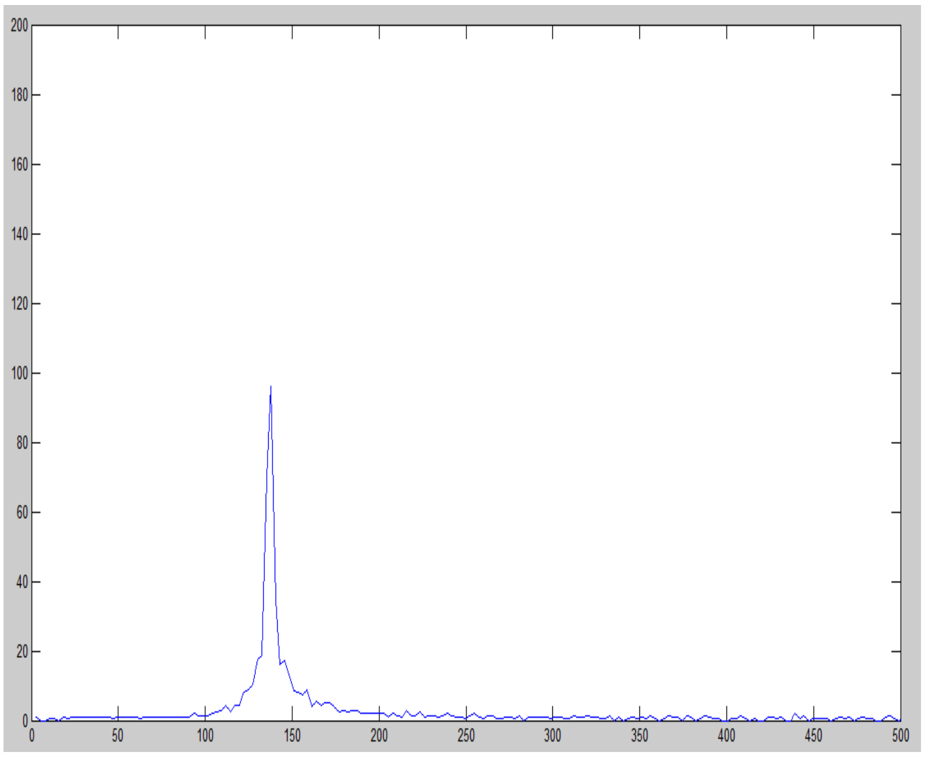

Figure 44 The peak at 135Hz displayed by MATLAB

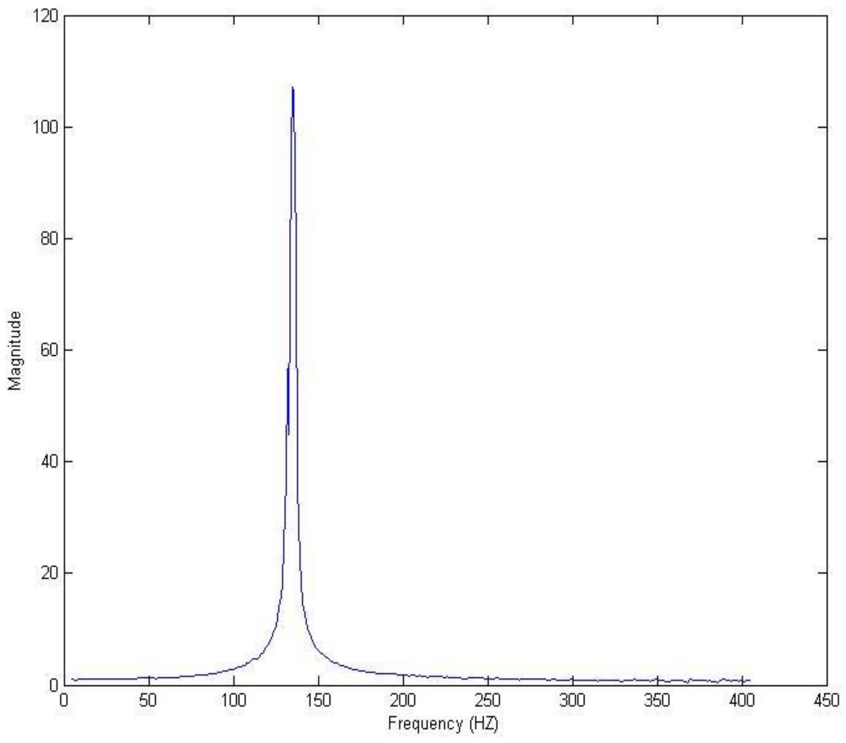

Figure 45 The peak at 135Hz from Hp35665 Analyzer

For Figure 46, the new fixture is placed, and the different material is tested. For this test, the peak is at 233 Hz, the amplitude is close to 20. The result from analyzer also has a peak close to 20 at 233 Hz, as shown in Figure 47.

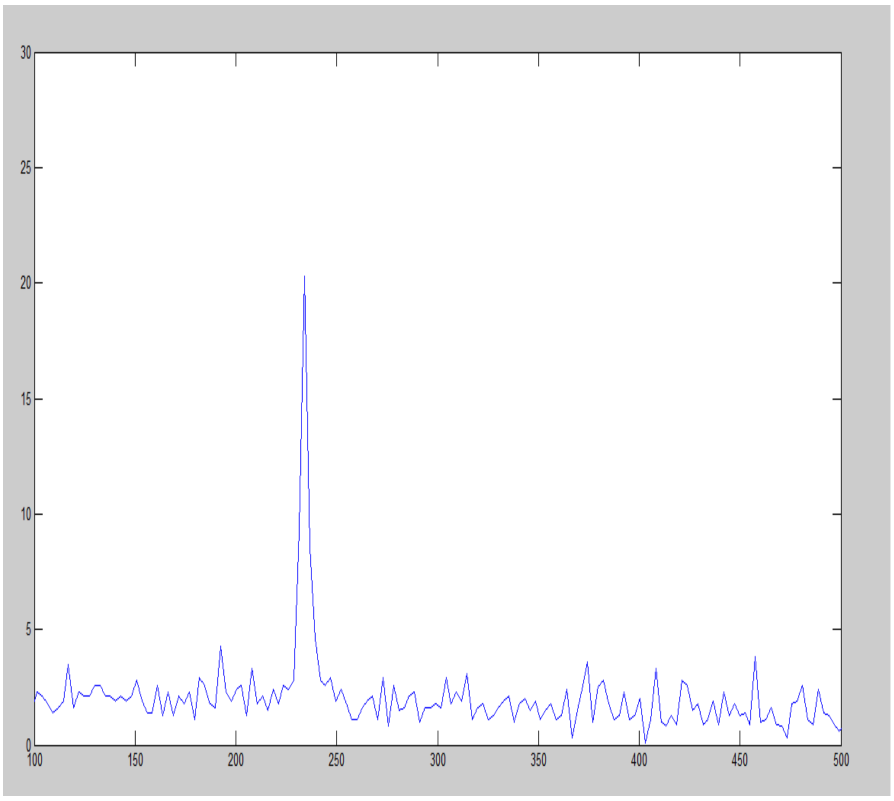

Figure 46 The peak at 233Hz displayed in MATLAB

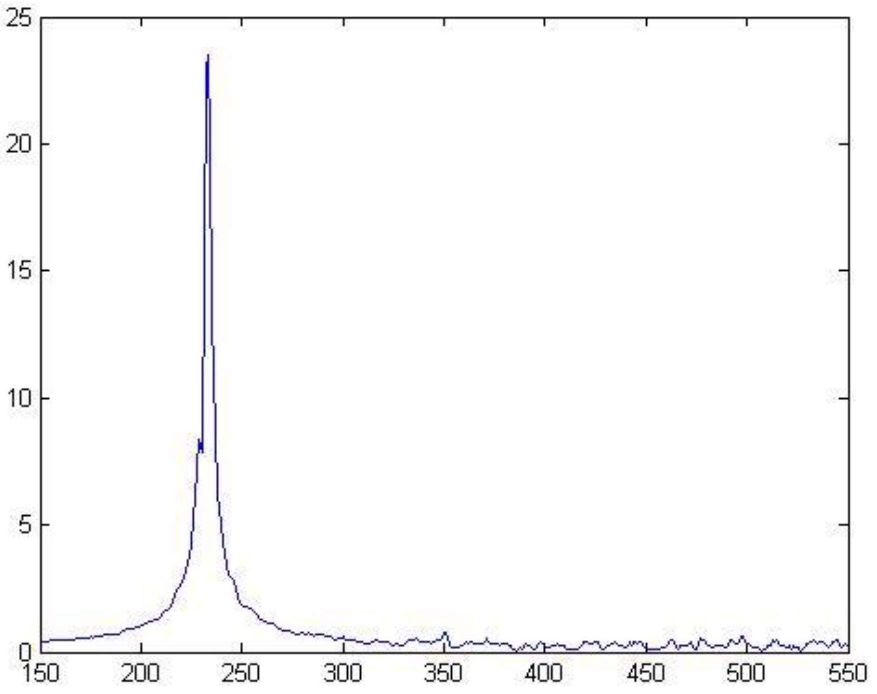

Figure 47 The figure from Hp35665 analyzer at 233Hz

Chapter 6 Conclusion and future work.

In conclusion, the design shows the high performance of the STM32f407, includes its computation ability and its DSP ability. The modified FFT algorithm is correct verified when it is used to calculate the FFT in the project. The VCP works well and the data is transmitted stably. The algorithm used after receiving data in the GUI successfully reduces the noise effect during the transmission. The speed to display the figure is good enough to observe the transmissibility. Through the averaging filter, the data received by the GUI can be refreshed 20s one time. Without averaging filter, the period is close to 4s. Furthermore, the data displayed in the GUI can be collected and written out to analyze later. 3 different samples are used to test the transmissibilities and the results of the design are uniform with the data from the HP 35665 analyzer.

For future work, the design can be used in different applications to test the response frequency. Not only for the MEMS device, but other useful applications can use this technique to observe the resulting phenomenon, such as LC circuits and film deposition. Furthermore, it can be embedded in some applications to perform its function.

For higher required calculation, the STM32f407 can be replaced by an FPGA, which has better performance and faster speed, since it is focused more on parallel execution. For an MCU, it needs more cycles to process in order to finish more work, but for the FPGA, it doesn't have this limit. The FPGA has a lot of FSM running all the time, it is only space limited and is not time limited, and bigger an FPGA means a higher ability of calculation. With the use of an FPGA, it can do a more precise FFT, and it can get better resolution. Furthermore,

more data and a better algorithm can be run in the FPGA, and the stability of the design can be improved. With the high performance of calculation of the FPGA, its ability of real-time will be better and the data received by the GUI can be refreshed faster.

For the noise present in the design, the design uses an averaging filter to reduce the noise effect. But sometimes, if the transmissibility signal is very weak, the result won’t be good, because the value of the noise will close to the value of the signal. For future work, to prove its stability, better filter techniques can be tried. A hardware filter can be used in the front of the design, or better filter algorithms can be used in the MCU or GUI, such as adaptive-filtering, which can be used to reduce the noise effect during the process.

For the GUI design, the design can be designed more humanized and more functions can be added so that it will be more easily used in future work.

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Appendix C code and C# code

(1) Main program for Stm32f407(ADC, DMA set)

```
#include "stm32f4xx.h"

#include "usbd_cdc_core.h"

#include "usbd_cdc_vcp.h"

#include "usbd_usr.h"

#include "usbd_desc.h"

#include "time.h"

#include "stm32f4_discovery.h"

#include <stdio.h>

#include <math.h>

#include <stdlib.h>

#include "arm_math.h"

#include "usbd_cdc_vcp.h"

#define ADC3_DR_ADDRESS          ((uint32_t)0x4001224C)
```

```
#define ADC1_DR_ADDRESS          ((uint32_t)0x4001204C)

__IO uint16_t ADC3ConvertedValue[8192];

float32_t Value1[2048];

float32_t Value2[2048];

void TimerInit(void);

void ADC3_CH12(void);

void TIM6_Config(void);

extern void FFT1_DAC(void);

__IO uint32_t ADC3ConvertedVoltage = 0;

__ALIGN_BEGIN USB_OTG_CORE_HANDLE        USB_OTG_dev __ALIGN_END ;

/* Extern variables ---------------------------------------------------------*/

extern uint16_t VCP_DataTx(uint8_t* Buf, uint32_t Len);

/* Private function prototypes -----------------------------------------------*/

void Delay(__IO uint32_t nTick);

void Delay1( unsigned);
```

```
int main(void)

{

      uint32_t i = 0;

          USBD_Init(&USB_OTG_dev,

#ifdef USE_USB_OTG_HS

    USB_OTG_HS_CORE_ID,

#else

    USB_OTG_FS_CORE_ID,

#endif

    &USR_desc,

    &USBD_CDC_cb,

    &USR_cb);

        ADC3_CH12();

  TimerInit();

    void ADC3_CH12(void)
```

```
{

  ADC_InitTypeDef         ADC_InitStructure;

  ADC_CommonInitTypeDef ADC_CommonInitStructure;

  DMA_InitTypeDef         DMA_InitStructure;

  GPIO_InitTypeDef        GPIO_InitStructure;

  /* Enable ADC3, DMA2 and GPIO clocks ****************************************/

  RCC_AHB1PeriphClockCmd(RCC_AHB1Periph_DMA2 | RCC_AHB1Periph_GPIOC, ENABLE);

  RCC_APB2PeriphClockCmd(RCC_APB2Periph_ADC1, ENABLE);



  /* DMA2 Stream0 channel0 configuration **************************************/

  DMA_InitStructure.DMA_Channel = DMA_Channel_0;

  DMA_InitStructure.DMA_PeripheralBaseAddr = (uint32_t)ADC1_DR_ADDRESS;

  DMA_InitStructure.DMA_Memory0BaseAddr = (uint32_t)&ADC3ConvertedValue;

  DMA_InitStructure.DMA_DIR = DMA_DIR_PeripheralToMemory;

  DMA_InitStructure.DMA_BufferSize = 8192;
  DMA_InitStructure.DMA_PeripheralInc = DMA_PeripheralInc_Disable;

  DMA_InitStructure.DMA_MemoryInc = DMA_MemoryInc_Enable;

  DMA_InitStructure.DMA_PeripheralDataSize = DMA_PeripheralDataSize_HalfWord;

  DMA_InitStructure.DMA_MemoryDataSize = DMA_MemoryDataSize_HalfWord;

  DMA_InitStructure.DMA_Mode = DMA_Mode_Circular;

  DMA_InitStructure.DMA_Priority = DMA_Priority_High;

  DMA_InitStructure.DMA_FIFOMode = DMA_FIFOMode_Disable;

  DMA_InitStructure.DMA_FIFOThreshold = DMA_FIFOThreshold_HalfFull;

  DMA_InitStructure.DMA_MemoryBurst = DMA_MemoryBurst_Single;

  DMA_InitStructure.DMA_PeripheralBurst = DMA_PeripheralBurst_Single;

  DMA_Init(DMA2_Stream0, &DMA_InitStructure);

  DMA_Cmd(DMA2_Stream0, ENABLE);

/* Configure ADC3 Channel12 pin as analog input ******************************/

  GPIO_InitStructure.GPIO_Pin = GPIO_Pin_2|GPIO_Pin_3;

  GPIO_InitStructure.GPIO_Mode = GPIO_Mode_AN;
  GPIO_InitStructure.GPIO_PuPd = GPIO_PuPd_NOPULL ;

  GPIO_Init(GPIOC, &GPIO_InitStructure);


  /*                    ADC                    Common                    Init
*********************************************************/

  ADC_CommonInitStructure.ADC_Mode = ADC_Mode_Independent;

  ADC_CommonInitStructure.ADC_Prescaler = ADC_Prescaler_Div8;

  ADC_CommonInitStructure.ADC_DMAAccessMode = ADC_DMAAccessMode_Disabled;

  ADC_CommonInitStructure.ADC_TwoSamplingDelay = ADC_TwoSamplingDelay_5Cycles;

  ADC_CommonInit(&ADC_CommonInitStructure);


  /*                                  ADC3                                  Init
***************************************************************/

  ADC_InitStructure.ADC_Resolution = ADC_Resolution_12b;

  ADC_InitStructure.ADC_ScanConvMode = ENABLE;

  ADC_InitStructure.ADC_ContinuousConvMode = ENABLE;
  ADC_InitStructure.ADC_ExternalTrigConvEdge = ADC_ExternalTrigConvEdge_None;

  ADC_InitStructure.ADC_DataAlign = ADC_DataAlign_Right;

  ADC_InitStructure.ADC_NbrOfConversion = 2;

  ADC_Init(ADC1, &ADC_InitStructure);

  /* ADC3 regular channel12 configuration *************************************/

  ADC_RegularChannelConfig(ADC1, ADC_Channel_12, 1, ADC_SampleTime_480Cycles);

  ADC_RegularChannelConfig(ADC1, ADC_Channel_13, 2, ADC_SampleTime_480Cycles);

 /* Enable DMA request after last transfer (Single-ADC mode) */

  ADC_DMARequestAfterLastTransferCmd(ADC1, ENABLE);



  /* Enable ADC3 DMA */

  ADC_DMACmd(ADC1, ENABLE);



  /* Enable ADC3 */

  ADC_Cmd(ADC1, ENABLE);
```

```
        ADC_SoftwareStartConv(ADC1);

        }

void Delay(__IO uint32_t nTick)

{

    for(; nTick != 0; nTick--);

}



void Delay1(unsigned x)

{

  unsigned i;

  for (i=0;i<x;i++){;}

}



#ifdef USE_FULL_ASSERT

/**********************************************************************
```

```
****

* Function Name    : assert_failed

* Description        : Reports the name of the source file and the source line number

*                              where the assert_param error has occurred.

* Input                   : - file: pointer to the source file name

*                              - line: assert_param error line source number

* Output                 : None

* Return                 : None

void assert_failed(uint8_t* file, uint32_t line)

{

   /* User can add his own implementation to report the file name and line number,

       ex: printf("Wrong parameters value: file %s on line %d\r\n", file, line) */



   /* Infinite loop */

   while (1)
```

```
    {}

}

#endif
```

(2) Timer set

```
void TimerInit()

{

        NVIC_InitTypeDef NVIC_InitStructure;

        GPIO_InitTypeDef GPIO_InitStructure;

        TIM_TimeBaseInitTypeDef TIM_TimeBaseStructure;

        RCC_APB1PeriphClockCmd(RCC_APB1Periph_TIM2, ENABLE);

        RCC_AHB1PeriphClockCmd(RCC_AHB1Periph_GPIOD, ENABLE);

        NVIC_InitStructure.NVIC_IRQChannel = TIM2_IRQn;

        NVIC_InitStructure.NVIC_IRQChannelPreemptionPriority = 0;

        NVIC_InitStructure.NVIC_IRQChannelSubPriority = 1;
```

```
    NVIC_InitStructure.NVIC_IRQChannelCmd = ENABLE;

    NVIC_Init(&NVIC_InitStructure);

    TIM_TimeBaseStructure.TIM_Period = 65535;

    TIM_TimeBaseStructure.TIM_Prescaler = 4199;

    TIM_TimeBaseStructure.TIM_ClockDivision = TIM_CKD_DIV1;

    TIM_TimeBaseStructure.TIM_CounterMode = TIM_CounterMode_Up;

    TIM_TimeBaseStructure.TIM_RepetitionCounter = 0;

    TIM_TimeBaseInit(TIM2, &TIM_TimeBaseStructure);

    TIM_ITConfig(TIM2, TIM_IT_Update, ENABLE);

    TIM_Cmd(TIM2, ENABLE);



        GPIO_InitStructure.GPIO_Pin = GPIO_Pin_12 | GPIO_Pin_13| GPIO_Pin_14|
GPIO_Pin_15;

    GPIO_InitStructure.GPIO_Mode = GPIO_Mode_OUT;

    GPIO_InitStructure.GPIO_OType = GPIO_OType_PP;
```

```
        GPIO_InitStructure.GPIO_Speed = GPIO_Speed_100MHz;

        GPIO_InitStructure.GPIO_PuPd = GPIO_PuPd_NOPULL;

        GPIO_Init(GPIOD, &GPIO_InitStructure);

      }
```

(3) Timer IRQHandler

```
void TIM2_IRQHandler(void)

{

  Delay1(1000000);

        if (TIM_GetITStatus(TIM2, TIM_IT_Update) == SET)

        {

                        int i=0;

                        int k1=0;

                        int k2=0;

                        TIM2->CR1 &= (uint16_t)~TIM_CR1_CEN;
```

```
              ADC1->CR2 &=~(1 << 1);

              ADC_DiscModeCmd(ADC1,DISABLE);

              for(i=0;i<8192;i++)

           {

           if(i%8==0)

          {

               Value1[k1]=ADC3ConvertedValue[i];

              Value1[k1+1]=0;

              k1=k1+2;

                Value2[k2]=ADC3ConvertedValue[i+1];

              Value2[k2+1]=0;

              k2=k2+2;

            }

   }

         FFT_DAC();
```

```
}

      ADC1->CR2 |= (1 << 1);

      ADC_SoftwareStartConv(ADC1);

     TIM2->CR1 |= TIM_CR1_CEN;

     TIM_ClearITPendingBit(TIM2,    TIM_FLAG_Update);

}
```

(4) Using FFT to get transmissibility result and sending the data to GUI by VCP.

```
void FFT_DAC(void)

{

    arm_status status;

    arm_cfft_radix4_instance_f32 S;

    float32_t maxValue;

   unsigned    int sum=0;
unsigned int sum1=0;

int shift=0;

  int i=0;

int h=0;

            for(h=0;h<32;h=h+2)

              {

             Value3[h]= h;

          Value3[h+1]= 0;}

             arm_cfft_radix4_init_f32(&S, fftSize,     ifftFlag, doBitReverse);

            arm_cfft_radix4_f32(&S, Value1);

            arm_cmplx_mag_f32(Value1, testOutput1,     fftSize);

          arm_cfft_radix4_f32(&S, Value2);

          calculating the magnitude at each bin */

          arm_cmplx_mag_f32(Value2, testOutput2,     fftSize);

        VCP_DataTx (0, 255);
VCP_DataTx (0, 255);

VCP_DataTx (0, 255);

testOutput1[0]=testOutput1[0]/2;

testOutput2[0]=testOutput2[0]/2;

y=testOutput1[b]/testOutput2[b];

y1[b]=y;

y2[b]=(y3[b]*kkk1+ y1[b])/(kkk1+1);

  y3[b]=y2[b];

  k1 = y;

  y=y>>8;

  k2=y;

 y=y>>8;

 k3=y;

  VCP_DataTx (0, (uint32_t)k3);

 VCP_DataTx (0, (uint32_t)k2);
```

```
                VCP_DataTx (0, (uint32_t)k1);

               sum=sum+k1+k2+k3;

}

  kkk1=kkk1+1;

  k4=sum;

sum=sum>>8;

k5=sum;

sum =sum>>8;

k6=sum;

VCP_DataTx (0, (uint32_t)k6);

VCP_DataTx (0, (uint32_t)k5);

VCP_DataTx (0, (uint32_t)k4);

sum=0;

for(w=0; w<512; w++)

        {
          phase1[w]    = (atan(Value1[2*w+1]/Value1[2*w]))*180/(4*atan(1));

          phase2[w]= (atan(Value2[2*w+1]/Value2[2*w]))*180/(4*atan(1));

           if (Value1[2*w]<0)

{

             if(Value1[2*w+1]>0)

             phase1[w]= phase1[w]+180;

             else phase1[w]=phase1[w]-180;}

           if (Value2[2*w]<0)

    {

             if(Value2[2*w+1]>0)

             phase2[w]= phase2[w]+180;

             else phase2[w]=phase2[w]-180;

}

             phase3[w]=phase1[w]-phase2[w];

             phase[w]=(phase3[w]+360)/4;
```

```
            shift=phase[w];

            k7=shift;

            shift=shift>>8;

           k8=shift;

           VCP_DataTx(0, (uint32_t)k8);

          VCP_DataTx(0,    (uint32_t)k7);

          sum1= sum1+k7+k8;

    }

         k9=sum1;

        sum1=sum1>>8;

        k10=sum1;

        sum1 =sum1>>8;

        k11=sum1;

        VCP_DataTx (0, (uint32_t)k11);

        VCP_DataTx (0, (uint32_t)k10);
    VCP_DataTx (0, (uint32_t)k9);

    sum1=0;

}
```

(5) Programme in Visual Studio

```
using System;

using System.Collections.Generic;

using System.ComponentModel;

using System.Data;

using System.Drawing;

using System.Text;

using System.Windows.Forms;

using System.Drawing.Imaging;

using System.IO.Ports;

using System.Threading;

using System.Collections;

using System.IO;

namespace OwnCarPC

{

     public partial class Form1 : Form
```

```
{
    private    object obj = new object();
    private bool Listening = false;
    private Graphics mapGraphics;
    private Bitmap bufMap;
    private Color backColor = Color.White;
    private Color nowColor = Color.Red;
    private Color targetColor = Color.DarkSeaGreen;
    private Color broderColor = Color.Red;
    private int p = 0;
    private int[] table;
    private int t = 0;
    private int[] numbers;
   private int targetX = 50;
   private int targetY = 50;
  private float nowX = 0;
 private float nowY = 0;
  private float oldY = 0;
    private float oldX = 0;
    int[]oldM= new int[1000];
    private float Y = 0;
    private SerialPort scom;
```

```
        private Boolean uartOpen = false;

        private double detDist = 0;

        private double detDeg = 0;

        int kk = -10;

       int we = 0;

       int LL = 0;

       int[]    nowN= new int[1000];

       int[]    nowM= new int[1000];

       int[] nowL = new int[1000];

       int[] oldL = new int[1000];

       int ss=100;

       int kkk3 = 0;

     int[] nowP = new int[612];

     int[] oldP= new int[612];

   int[] olds = new int[612];

   int[,]ten = new int[600, 600];

   int[]    nowt = new int[2540];

   int[] nowk = new int[3000];

            int[] nows = new int[3000];

            int[] nowe = new int[3000];

            int[] olde = new int[3000];

            int[] nowf = new int[3000];
            int[] oldf = new int[3000];
          private ArrayList detDistList = new ArrayList();
          private ArrayList detDegList = new ArrayList();
         private    int change =10;
public int counter = 0;
     int num = 0;
          int res = 0;
          int l=0;
          int kkk1 = -10;
          ushort sum;
          int phase = 0;
          int[] average = new int[1000];
          int[] average1 = new int[1000];
          int kk4 = 0;
     int k = 0;
          public Form1()
          {     InitializeComponent();
                bufMap = new Bitmap(520, 600);
              mapGraphics = Graphics.FromImage(bufMap);//初始化
              ResetMap();
              this.uartComboBox.DropDownStyle = ComboBoxStyle.DropDownList;
            }
```

```
private void button2_Click(object sender, EventArgs e)
        {
      change = 10;
        }


        private void button3_Click(object sender, EventArgs e)
        {
            Form2 showForm = new Form2();
            showForm.Show();
          }
        private void button4_Click(object sender, EventArgs e)
        {
            //Form2 showForm = new Form2();
            // showForm.Show();
            res=1;
        }
     public void ResetMap()
        {
            int i = 0;
            mapGraphics.DrawRectangle(new Pen(broderColor), 0, 0, 1999, 1999);
            mapGraphics.FillRectangle(new SolidBrush(backColor), 1, 1, 1998, 1998);
            while (i < 520)
            {
                mapGraphics.DrawLine(new Pen(broderColor), i, 510, i + 3, 510);
                mapGraphics.DrawLine(new Pen(broderColor), i, 555, i + 3, 555);
                mapGraphics.DrawLine(new Pen(broderColor), i, 465, i + 3, 465);
                mapGraphics.DrawLine(new Pen(broderColor), i, 400, i + 3, 400);
               // mapGraphics.DrawLine(new Pen(broderColor), 249, i, 249, i + 3);
                i += 5;
            }
            while (i < 2000)
            {
                mapGraphics.DrawLine(new Pen(broderColor), i, 999, i + 3, 999);
                mapGraphics.DrawLine(new Pen(broderColor), 999, i, 999, i + 3);
                i += 10;
            }
          nowX = 0;
            Y = 0;
            pictureBoxMap.Image = bufMap;
        }
private void pictureBoxMap_MouseClick(object sender, MouseEventArgs e)
        {
            ResetMap();
            targetX = e.X;
```

```
                    targetY = -(e.Y-99);

                    mapGraphics.FillEllipse(new SolidBrush(targetColor),e.X,e.Y,6,6);

                    targetPointLabel.Text = targetX.ToString() +".0 , "+ targetY.ToString()+".0";

                    distLabel.Text =

Math.Sqrt((long)(targetX-nowX)*(targetX-nowX)+(long)(targetY-Y)*(targetY-Y)).ToString();

                    pictureBoxMap.Image = bufMap;



                    nowX = 0;

                    Y = 0;

                    oldX = 0;

                    oldY = 0;

                      }

  private void pictureBoxMap_MouseMove(object sender, MouseEventArgs e)

          {

                   cursorPointLabel.Text = Convert.ToString(e.X) + ".0 , " +

Convert.ToString(e.Y)+".0";

                    }

 private void Form1_Load(object sender, EventArgs e)

          {

                    Control.CheckForIllegalCrossThreadCalls = false;

                    string[] sNames = getUARTNames();

                    uartComboBox.Items.Clear();
```

```
                  uartComboBox.Items.Add("com");
                  if (sNames.Length == 0)
                  {
                      MessageBox.Show("no com");
                         }
                  else
                  {
                      //uartComboBox.Items.Clear();
                      for (int i = 0; i < sNames.Length; i++)
                      {
                          uartComboBox.Items.Add(sNames[i]);
                      }
                  }
                  uartComboBox.SelectedIndex = 0;
        }
         private void saveButton_Click(object sender, EventArgs e)
              {
                  SaveFileDialog dig = new SaveFileDialog();
                  dig.RestoreDirectory = true;
                  if (dig.ShowDialog() == DialogResult.OK)
                  {
                      bufMap.Save(dig.FileName + ".jpg", ImageFormat.Jpeg);
```

```
                }

            }

            private string []getUARTNames()

            {

                string[] names=null;

                try

                {

                    names = SerialPort.GetPortNames();

                }

                catch (Exception ex)

                {

                    MessageBox.Show(“fault:"+ex.Message.ToString());



                }

                return names;

            }

        private void uartComboBox_SelectedIndexChanged(object sender, EventArgs e)

            {

             if(uartComboBox.SelectedIndex!=0)

                {

                    try

                    {
```

```
                    scom = new SerialPort(uartComboBox.Text.ToString(),
                                                                                      115200,
Parity.Odd,
                                                                                        8,
StopBits.One);
                    scom.RtsEnable = false;
                    scom.ReadTimeout = 3300;
                    if (!scom.IsOpen)
                    {
                        scom.Open();
                        uartOpen = true;
                        //scom.DataReceived+=new
SerialDataReceivedEventHandler(scomDataReceived);
                        scom.DataReceived += new
SerialDataReceivedEventHandler(scom_DataReceived);
                        MessageBox.Show(uartComboBox.Text.ToString() + "open
successfully");
                    }
                }
                catch (Exception ex)
                {
                    uartOpen = false;
```

```
                              MessageBox.Show("open abnormal" + ex.Message.ToString());
                         }
           }
               public          delegate void UpdateTxt();
            public UpdateTxt update;
          public void ThreadMethodTxt()
            {
                 if (this.InvokeRequired)
                 {
                 this.Invoke(new EventHandler(updatepoint));
                            return;
                 }
            }

            private void scom_DataReceived(object sender, SerialDataReceivedEventArgs e)
                  {
                     try
                       {
                            Thread.Sleep(500);
                          Listening = true;
                          byte[] scomBuf = new byte[150000];
                          scom.Read(scomBuf, 0, scomBuf.Length);
```

```
                    int s = 0;

                      int f = 0;

                     int w = 0;

                     int b = 0;

                     int k1 = 0;

                     int k2 = 0;

                     int ee = 0;

                     int kk2 = 0;

            if (scomBuf[0] == 255)

                     {



                        if (scomBuf[i+1] == 255)

                            {

                                if (scomBuf[1] == 255)

                                {

                                    if (scomBuf[2] == 255)

                                    for (int a = 0; a < 1537; a = a + 3)

                               {

nowN[k] = (scomBuf[5 + a ] + (scomBuf[4 + a ] << 8) + (scomBuf[3 + a ] << 16))*change;

                          s = nowN[512];

                          nowt[a] = scomBuf[3 + a ];

                         nowt[a + 1] = scomBuf[4 + a ];
```

```
  nowt[a + 2] = scomBuf[5 + a ];
    if(kk>-1)
          {
          nowN[k] = 0;
  ten[kkk3, k] = nowN[k];
            }
        if ((kkk3+1)%5==0)      //max and min
    {int max, min, sum3=0;
      max = ten[kkk3-4,k];
        min = ten[kkk3-4,k];
          for (int ii = 0; ii < 5; ii++)
            {
                if (max < ten[kkk3-4+ii, k])
              max = ten[kkk3-4+ii, k];
              else
            {
              if (min > ten[kkk3-4+ii, k])
          min = ten[kkk3-4+ii, k];
}
  sum3 = ten[kkk3-4+ii,k] + sum3;
                }
average[k] = (sum3 - max-min ) / 3;
```

```
                    average[k] = (average1[k] * kk4 + average[k])/(kk4+1);

                              }

                              k = k + 1;

                         }

                   for (int a = 1536; a < 2560; a = a + 2)

           {

               nows[b] = (scomBuf[a    + 7]+ (scomBuf[a+6]<<8;

               nowk[a] = scomBuf[6 + a ];

               nowk[a + 1] = scomBuf[7 + a ];

                 if (kk> -1)

             {

                   nowP[b] = (oldP[b] * kk + nows[b]) / (kk + 1); }

                   b = b + 1;

      }

  nowP[512] = (scomBuf[2568]) + (scomBuf[2567] << 8) + (scomBuf[2566] << 16);

             for (int c = 0; c< 1536; c++)

               {

       f = f + nowt[c];

}

      f= f* change;

      k1=nowP[512];

      for (int c = 1536; c < 2560; c++)
```

```
                        {
                              k2 = k2 + nowk[c];
}
                      if (s == f)
              {
                        if (k1 == k2)
                          {
                                  this.Invoke(new EventHandler(updatepoint));
                                  Thread.Sleep(4500);
                        }
          }
                              if ((kkk3 + 1) % 5 == 0)
                                  {
                              kk4 = kk4 + 1;
                              }
                                  if(kk>-1)
                                                          {
                              kkk3 = kkk3 + 1;
                                }
                            kk = kk + 1;
                            l = l + 1;
                            k = 0;
```

```
                    f = 0;

                    b = 0;

                    k2 = 0;

                    we = 0;

                    if (res == 1)

                            {

                              kk = -5;

                              res=0;

                             }



                             }

                         }

                      }

                 }

             catch (Exception ex)

             {

           MessageBox.Show("com port abnormal:" + ex.Message.ToString());

             }



        }

private void Form1_FormClosed(object sender, FormClosedEventArgs e)

        {
                if ((uartOpen = true)&&(scom!=null))

                {

                        scom.Close();

                }

        }

private void updatepoint1(object sender, EventArgs e)

          {    }



          private void updatepoint(object sender, EventArgs e)

        {

            // lock(obj)

                {



                              try

                              {



                average[512] = 1280;

                average[513] = 1280;

                      for (int i = 0; i < 512; i++)

                                {

                              mapGraphics.DrawLine(new Pen(backColor), 2*(i +

                                          2), 400-average1[i], 2*(i + 3), 400 - average1[i+1]);
```

```
                                    }
                                   for (int i = 0; i < 512; i++)
                                    {
                                         mapGraphics.DrawLine(new Pen(targetColor), 2*(i+2),
400-average[i], 2*(i+3), 400 - average[i+1]);
                                            average1[i] = average[i];
                                    }
                                  for (int i = 0; i < 512; i++)
                                    {
                               mapGraphics.DrawLine(new Pen(backColor), i +
                                 2, 600-oldP[i], i + 3,600 - oldP[i+1]);
                                    }
                                 for (int i = 0; i < 512; i++)
                                    {
                                           mapGraphics.DrawLine(new Pen(targetColor), i + 2,
    600-nowP[i], i + 3, 600 - nowP[i+1]);
                                oldP[i] = nowP[i];
                                    }
                              StreamWriter sw = File.AppendText("F:\\TestTxt.txt");
                               for (LL = 0; LL < 514; LL++)
                                  {
                             sw.Write(average[LL] + " ");
```

```
                    }
                    sw.Flush();
                   sw.Close();
                   pictureBoxMap.Image = bufMap;
                        }
                       catch (Exception ex)
                       {
                               MessageBox.Show("com port abnormal:" +
ex.Message.ToString());
                       }
                 }


}
        public ArrayList detDistList1
              {
                      set { detDistList=value;}
                      get{return detDistList;}
              }
              public ArrayList detDegList1
              {
                      set { detDegList = value; }
                      get { return detDegList; }
```

```
                }

            }

        }
```