Project 1： USRP

Author	Name：	Student ID:

Introduction

In this lab, we will realize the frequency modulation radio base on USRP. In telecommunications and signal processing, frequency modulation (FM) is the encoding of information in a carrier wave by varying the instantaneous frequency of the wave. In analog frequency modulation, such as FM radio broadcasting of an audio signal representing voice or music, the instantaneous frequency deviation, the difference between the frequency of the carrier and its center frequency, is proportional to the modulating signal. Frequency modulation is widely used for FM radio broadcasting. It is also used in telemetry, radar, seismic prospecting, and monitoring newborns for seizures via EEG, two-way radio systems, music synthesis, magnetic tape-recording systems and some video-transmission systems. In radio transmission, an advantage of frequency modulation is that it has a larger signal-to-noise ratio and therefore rejects radio frequency interference better than an equal power amplitude modulation (AM) signal. For this reason, most music is broadcast over FM radio.

Lab results & Analysis：

1. Theoretical Basis

a) FM

Firstly, the signal of FM can be expressed as the following equation. Instantaneous carrier frequency is varied linearly with the message signal.

$$f_{FM(t)} = A_c \cos[2\pi f_c t + 2\pi k_c \int_{-\infty}^{t} m_{(\tau)} d\tau]$$

Secondly, in the remodulation module, we adopt the method of orthogonal modulation to obtain the real part and the imaginary part signals, and finally generate the following FM signal as. The math process is shown below and figure 1 shows the program flow.

FM signal:

$$s_{(t)} = A_c \cos\left[2\pi f_c t + 2\pi k \int m_{(\tau)} d\tau\right]$$

Expand by formula:

$$s_{(t)} = A_c\, cos\left(2\pi k \int m_{(\tau)} d\tau\right) cos(2\pi f_c t) - A_c\, sin\left(2\pi k \int m_{(\tau)} d\tau\right) \sin(2\pi f_c t)$$

Decided into two parts:

$$s_{I(t)} = A_c\, cos\left(2\pi k \int m_{(\tau)} d\tau\right) \quad s_{Q(t)} = A_c\, sin\left(2\pi k \int m_{(\tau)} d\tau\right)$$

Result:

$$s_{(t)} = s_{I(t)} \cos(2\pi f_c t) - s_{Q(t)} \sin(2\pi f_c t)$$

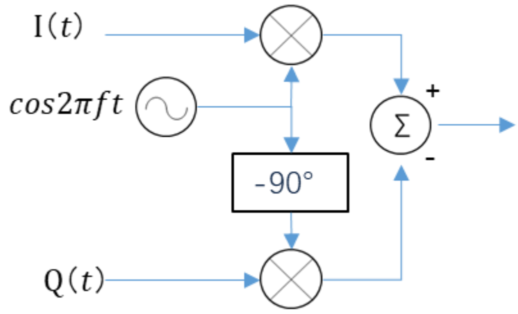


Figure 1 Modulation theory

Then, USRP receiver get the I-signal $s_{I(t)} = A_c \cos\left(2\pi k_c \int_{-\infty}^{t} m(\tau)\, d\tau\right)$ and Q-signal $s_{Q(t)} = A_c \sin\left(2\pi k_c \int_{-\infty}^{t} m(\tau)\, d\tau\right)$. And then compound them to a complex signal $s(t) = s_{I(t)} + js_{Q(t)}$. The key to the demodulation module of the received signal is the positive and arctangent method. From this equation $2\pi k \int m_{(nT)} d\tau = atan(\frac{s_{Q(nT)}}{s_{I(nT)}})$ we can recover the message $m_{(nT)} = \frac{1}{2\pi k}\frac{d}{dT}atan(\frac{s_{Q(nT)}}{s_{I(nT)}})$.

b) USRP

USRP (Universal Software Radio Peripheral) is designed to enable ordinary computers to function like high-bandwidth Software Radio devices. Essentially, it acts as the digital baseband and IF part of a radio communication system. Different USRP devices are used at both the transmitting and receiving ends.

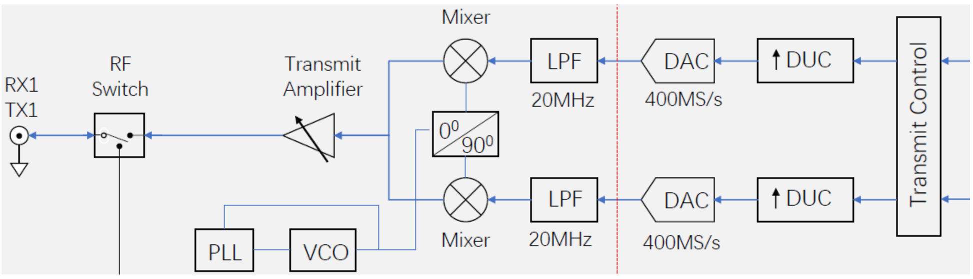


Figure 2 USRP transmitter

In the transmitter, baseband data up-converse to intermediate frequency. The digital signal is converted to analog. The LPF makes the signal smoother. The signal generated by the crystal oscillator modulates the analog signal to the specified frequency point.

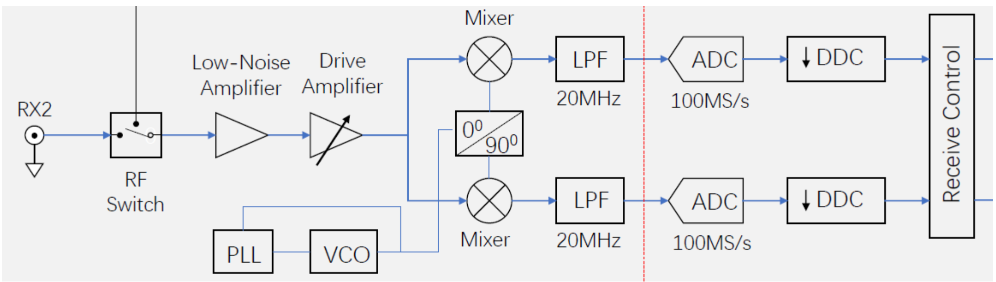


Figure 3 USRP receiver

On the other hand, in the receiver, after low noise amplification, the signal is down converted to the intermediate frequency. The LPF makes the signal smoother. Received signal down-converse to intermediate frequency. The analog signal is converted to digital signal. The signal is converted from the intermediate frequency down to the baseband.

2. Block Diagram

The overall program is designed to read data, demodulate, respond to signals, and play music. Therefore, the producer - consumer cycle is adopted to design the program. We firstly present a program that doesn't require DLL module. This shows the producer loop, which reads the IQ information from the file into the queue.

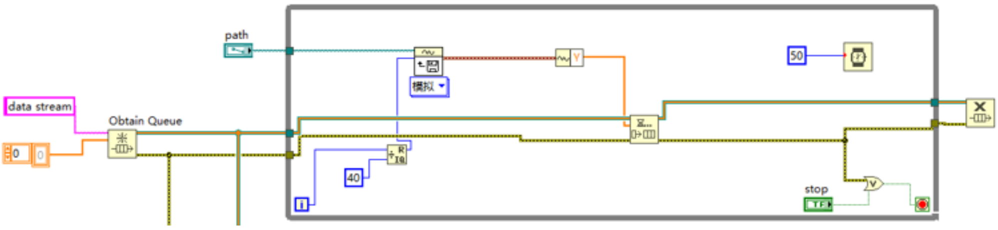


Figure 4 Producer cycle

The consumer loop is responsible for processing the data and restoring the signal. The first module, as shown in figure 5, is to read out the data in the queue as the real and imaginary parts. And

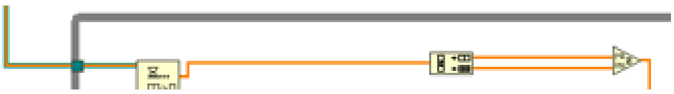


Figure 5 Fetch RX data

Figure 6 shows the demodulation module, which in turn is Unwrap Phase, Differential and Down Sampling.

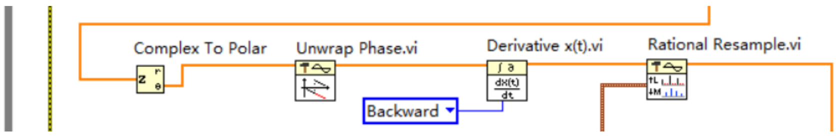


Figure 6 Demodulation

Finally, the recovered signal reaches the player through normalization, which is shown in figure 7. And the overall program is shown in the figure 8.

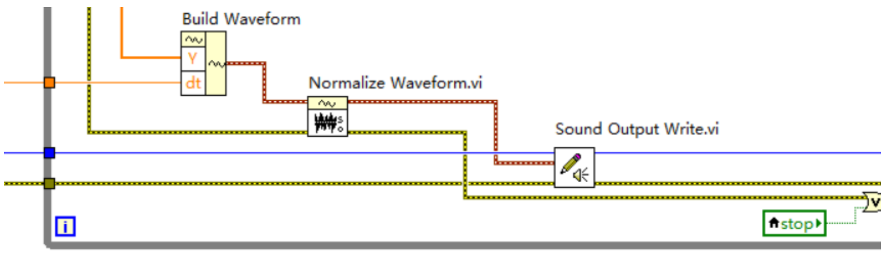


Figure 7 Play sound module

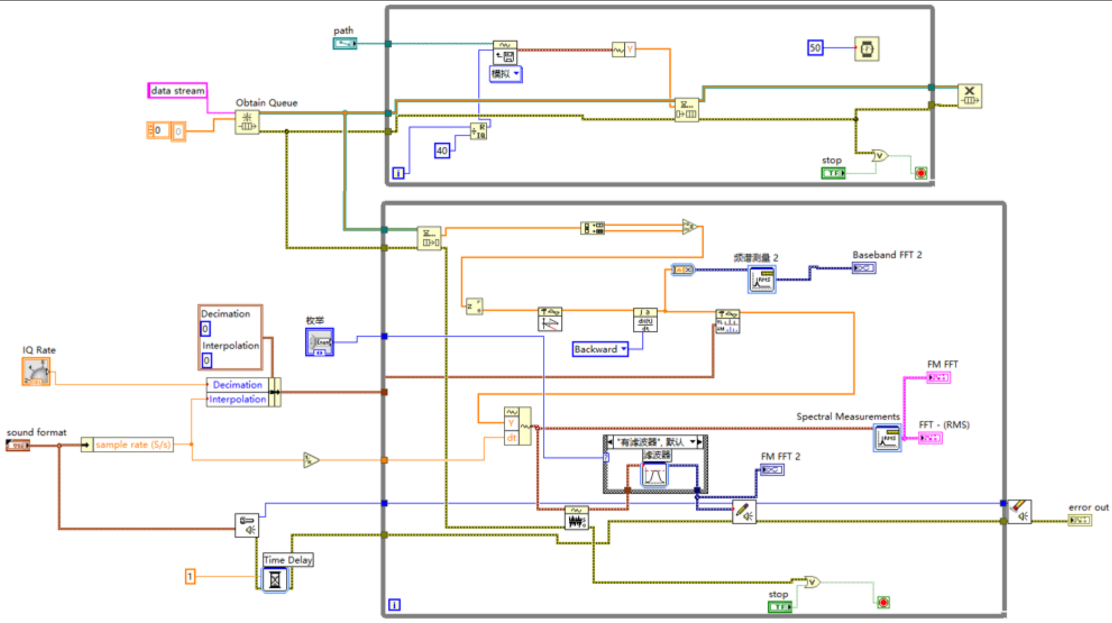
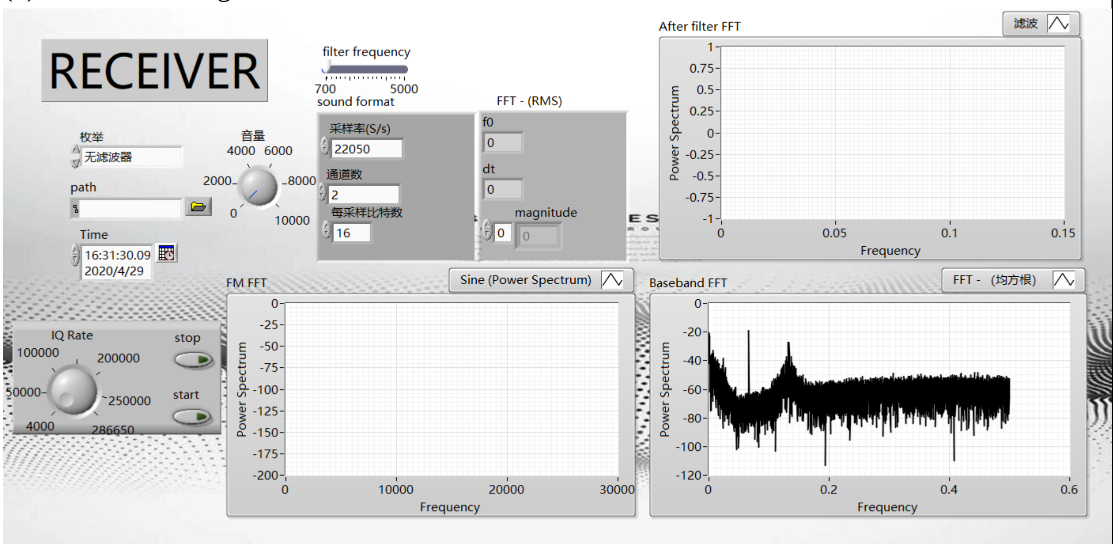


Figure 8 Overall program diagram

● Lab results

(1) interfacial design

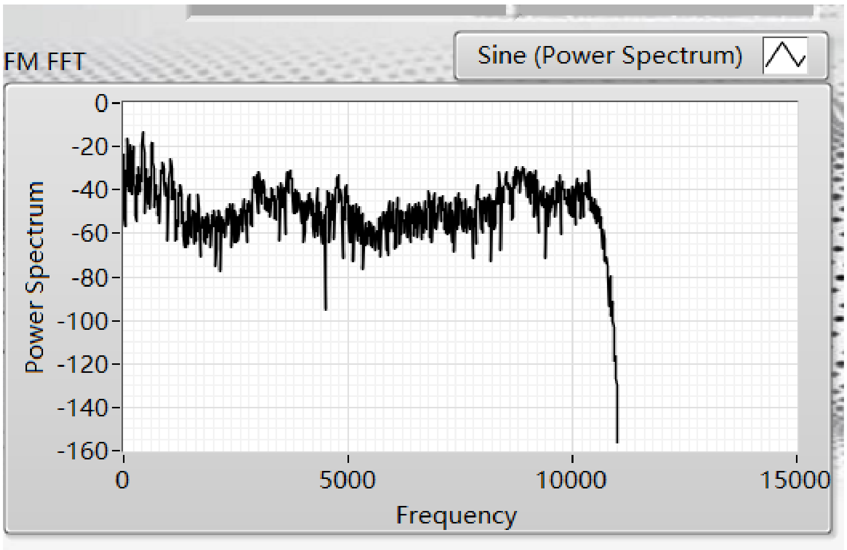


For an ordinary FM radio, the basic functions that should have are turning on and off, frequency adjustment and volume adjustment. These basic requirements were met in our interface design. We also added a time display and unique background images to enhance the user experience.

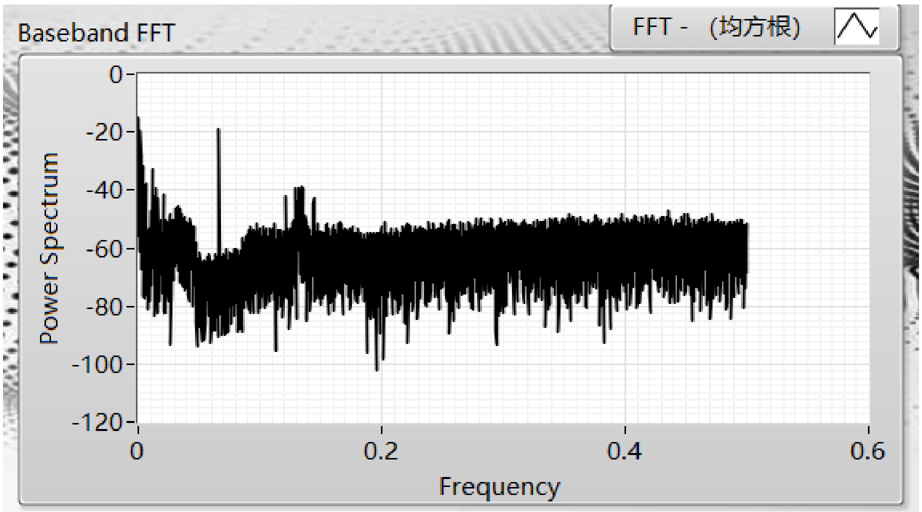
(2) Graphs

Under the best conditions we tested, we obtained the full audio of the song and the following results.

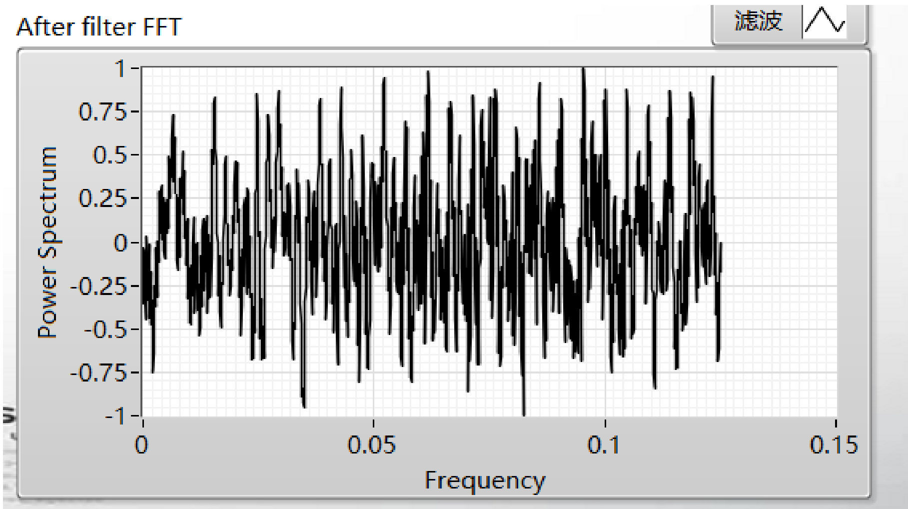
a. FFT （fast Fourier transform）result of FM



b. FFT （fast Fourier transform）result of baseband



c. FFT （fast Fourier transform）result of the signal after filter

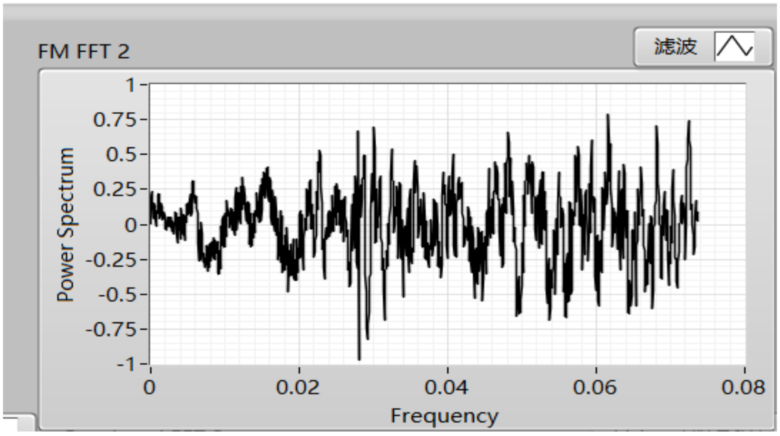


- **Analysis**

(1) Decimation

In our first attempt, IQ rate was set to decimation, not the interpolation. So we cannot use the given data:286650. The following conclusions are drawn from the repeated experiments on the numerical characteristics of the decimation

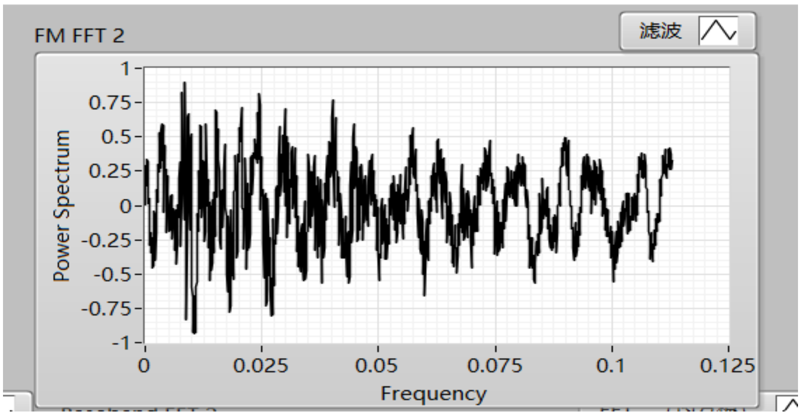
a. If decimation is small like decimation=4000



FFT （fast Fourier transform）result of the signal after filter

The output audio playback speed is very fast, the voice part of the audio is similar to the female voice or children's voice, the voice is sharp.

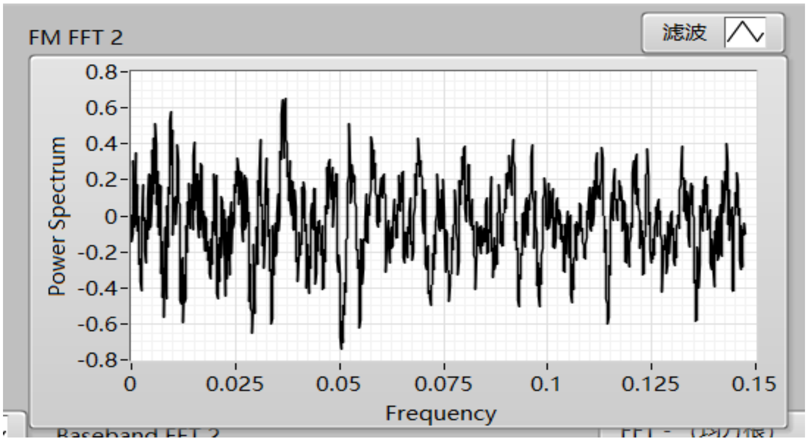
b. If decimation is meddle like decimation=6000



FFT （fast Fourier transform）result of the signal after filter

The output audio playback speed is normal, the voice part of the audio and the original song singer sound more consistent.

c. If decimation is large like decimation=8000



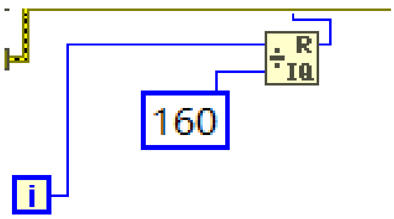
FFT （fast Fourier transform）result of the signal after filter

The speed of the output audio is slowed down. The voice part of the audio is similar to the bass, and individual sounds cannot be distinguished.

Through the above analysis of controlling single variable, we draw the following conclusions: First, the larger the decimation, the lower the pitch. Secondly, the decimation value and the audio playback speed are similar to the inverse relationship, the larger the decimation value, the slower the audio playback speed

(2) Song integrity

After a lot of trial and error, we found that the key to determining whether a song will play completely is to debug this parameter in the input.



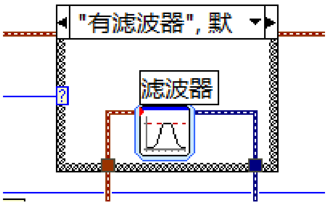
a. If we use the default value which is 40 here, it only gives out the early part of the music which is “this winter”.

b. If we use the debugged value which is 160 here, it can give out the whole parts of the music which is “this winter” to “subway”.

c. If we use the bigger value like 170, the system will report an error:

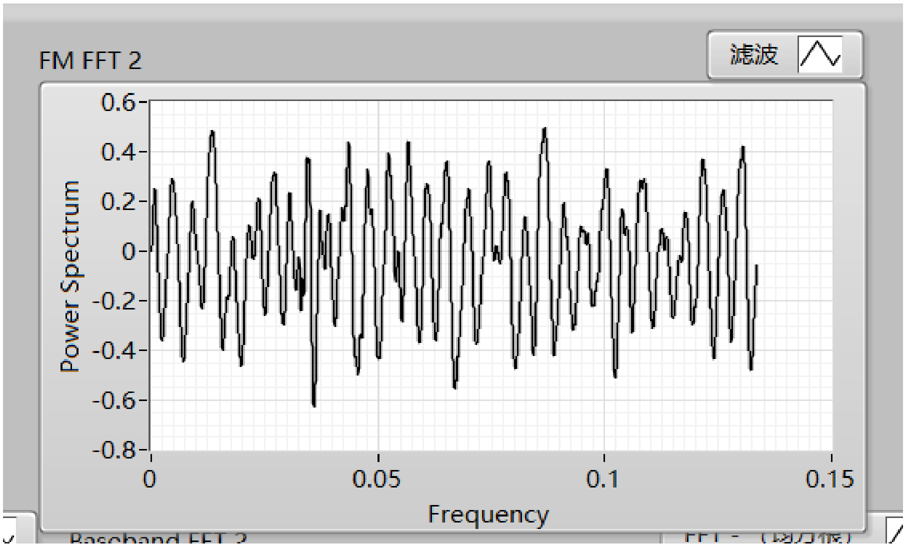
It is because this value is too large that even after the data is completely read the loop still not end.

(3) Filter



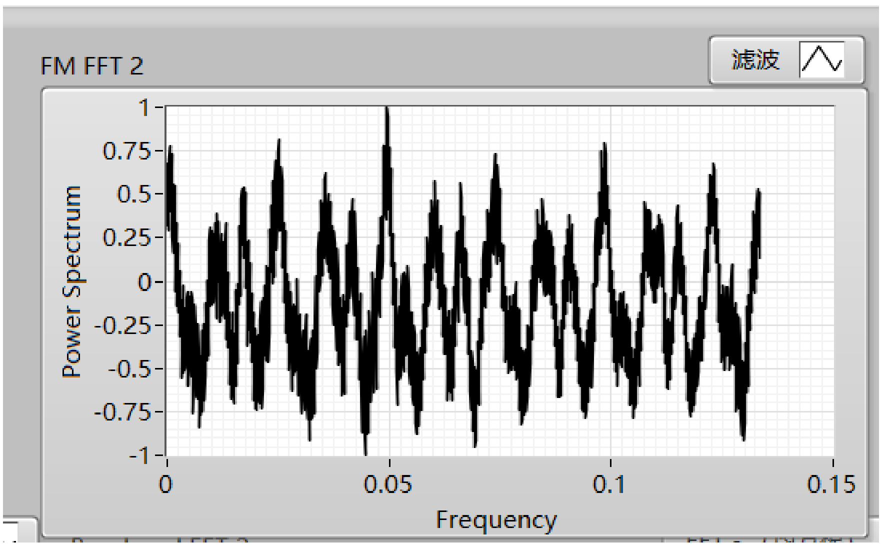
Considering that the frequency of male voice in human voice is between 64 and 523Hz, we set up a bandpass filter with corresponding frequency

a. with filter



FFT （fast Fourier transform）result of the signal after filter

b. without filter



FFT （fast Fourier transform）result of the signal after filter

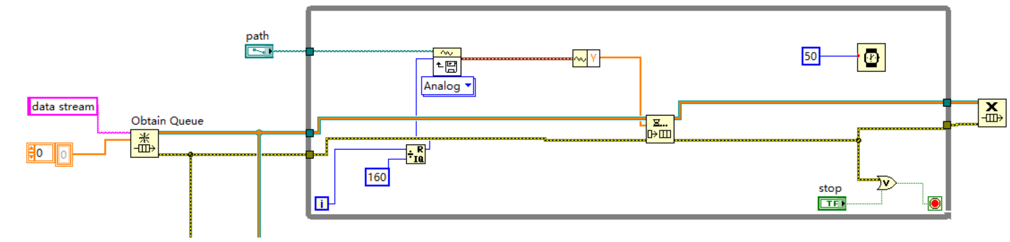
By comparing the above images with the corresponding audio files, we draw the following conclusions：

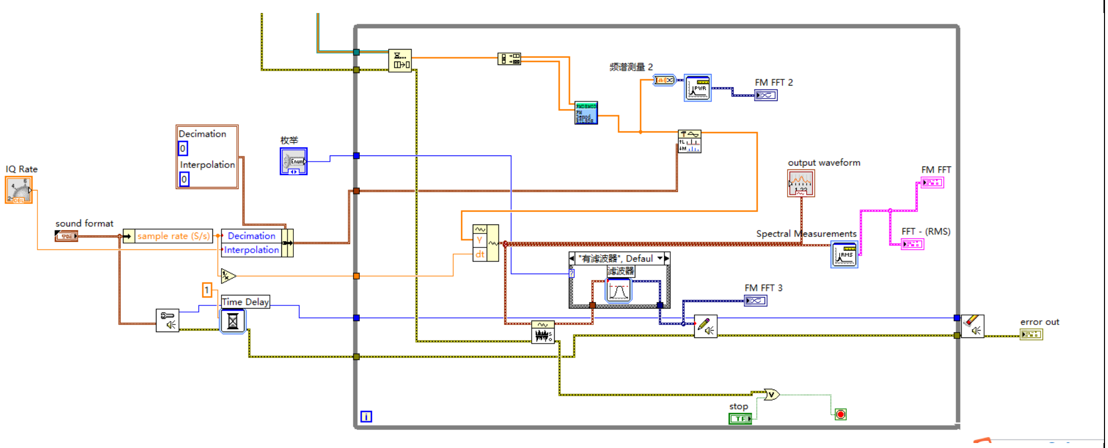
First of all, the filter is really good at eliminating the audio bottom noise. By comparing the sounds before and after adding the filter, we can easily find that the voice part with the filter is more clear, the syllables are less disturbed, and it is easier to distinguish.

Second, with the addition of a filter, the acoustic details of the instrument in the background are reduced, and the pop of a soda can be heard.

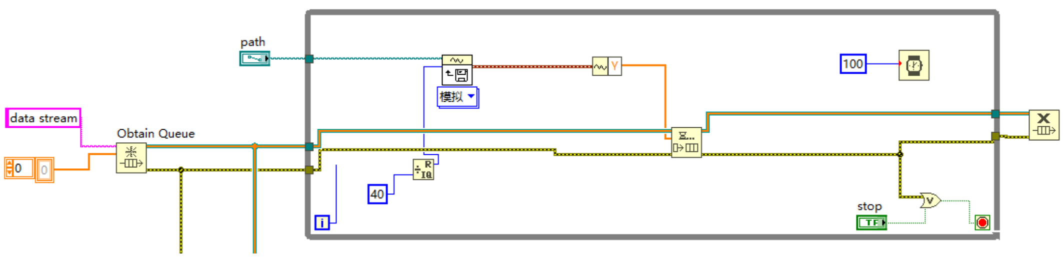
The advanced research

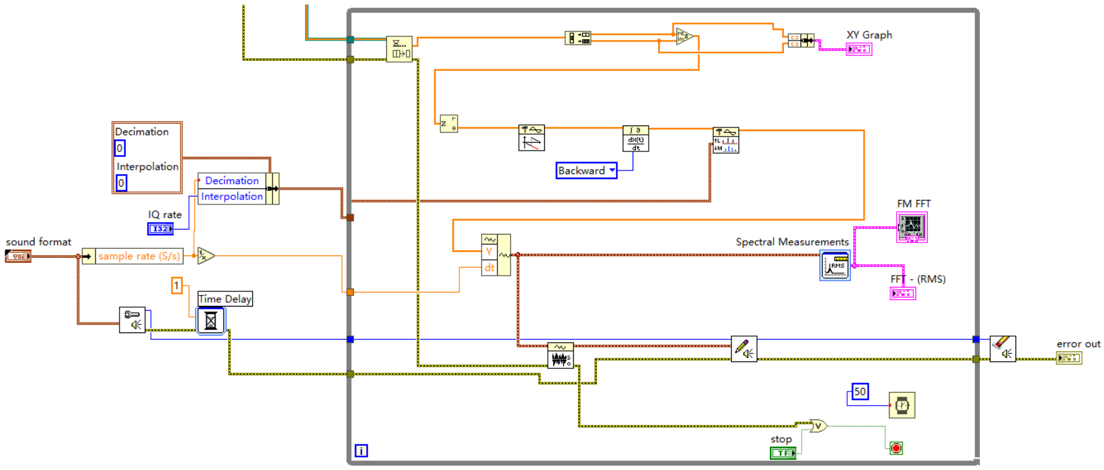
1.Program flowchart





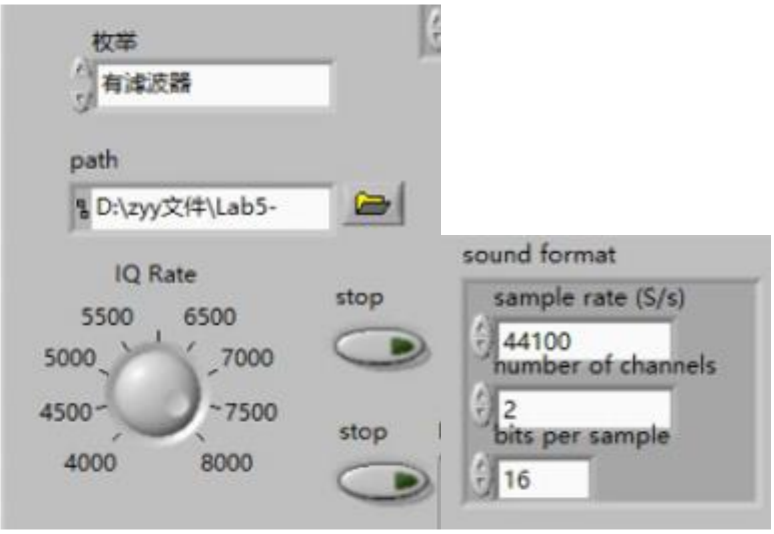
Add DLL modulation



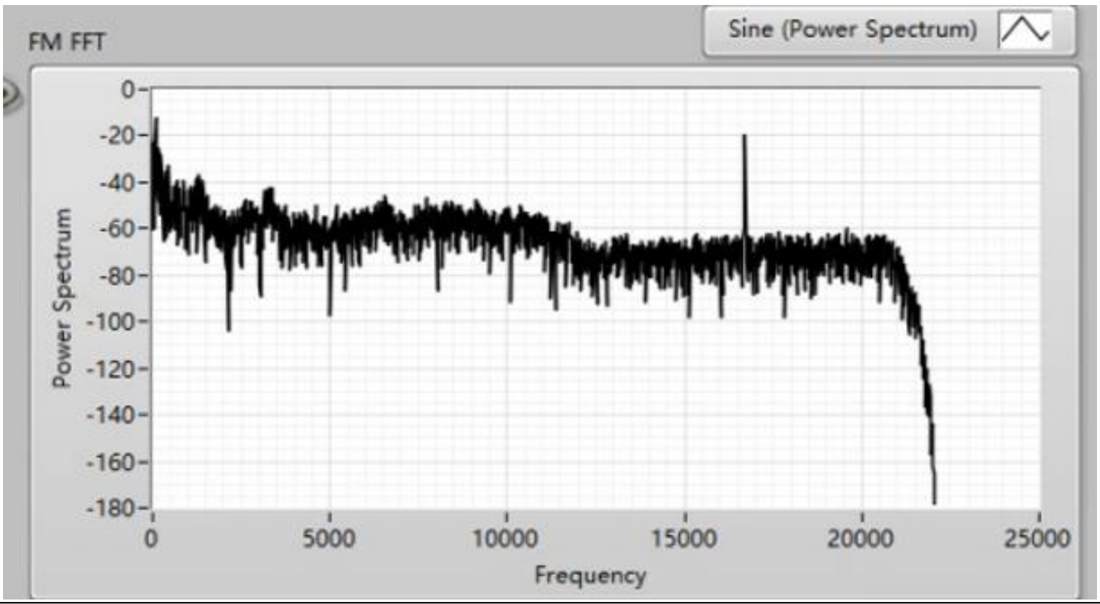


Without DLL

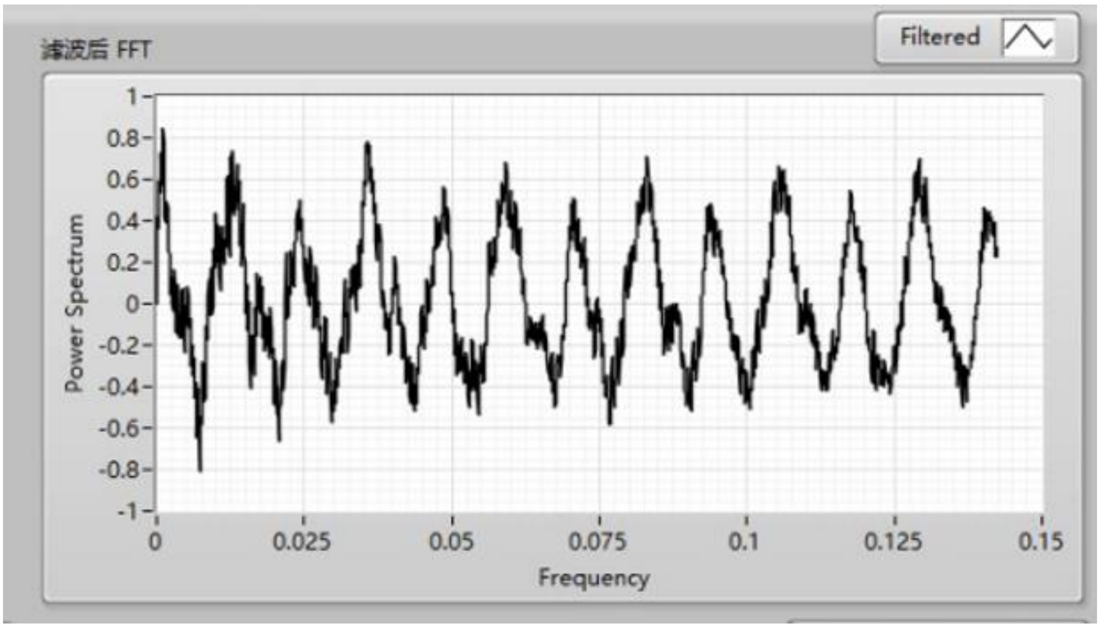
The results presented



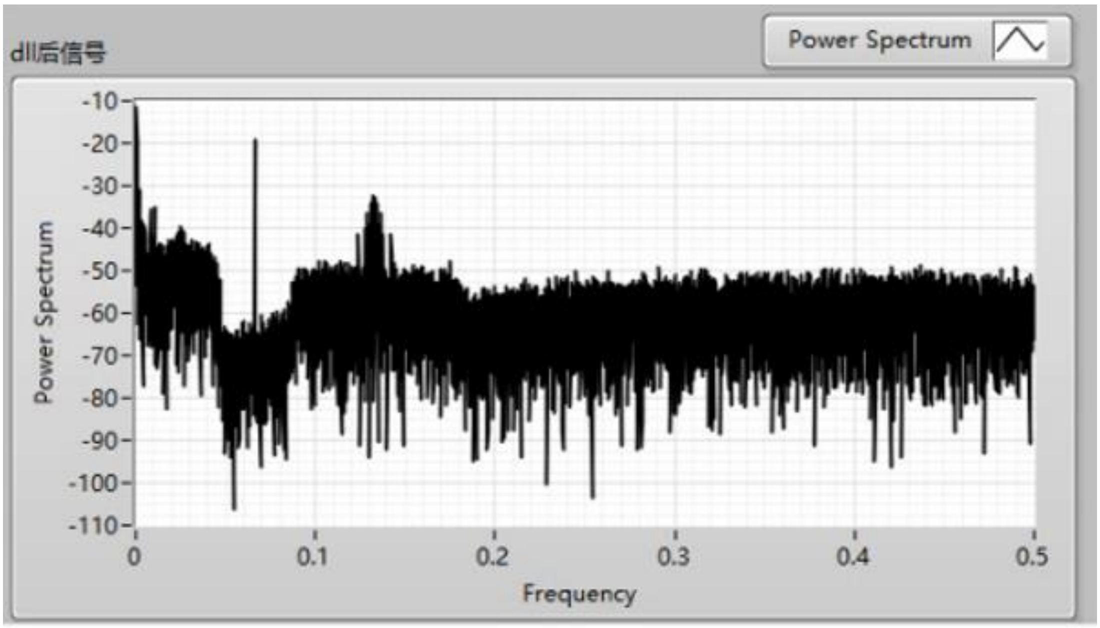
This is the data input part of the system



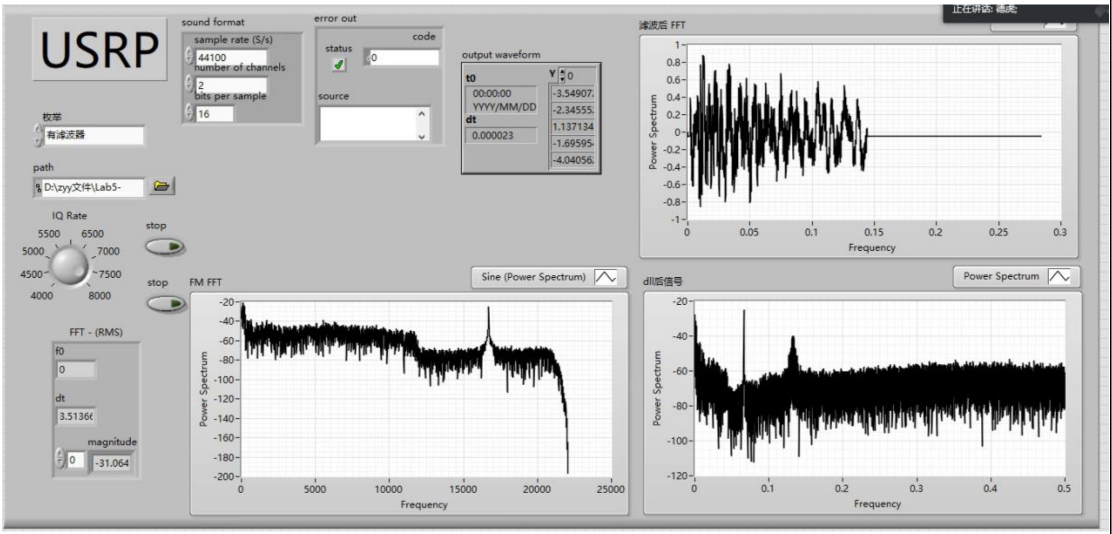
This is the FFT of the sampled signal



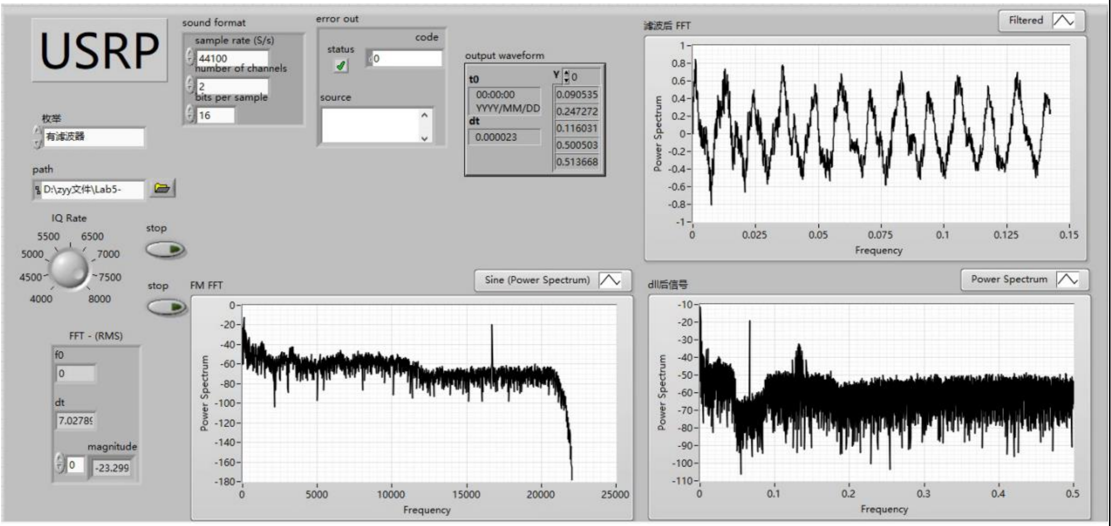
This is the FFT of the filtered signal



This is the signal after the DLL operation

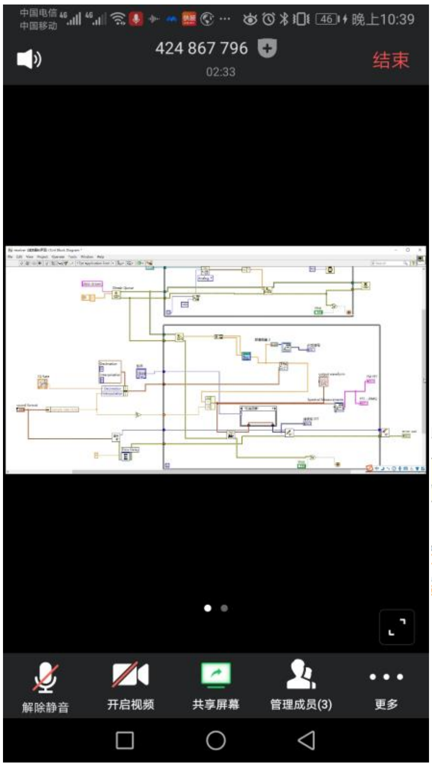


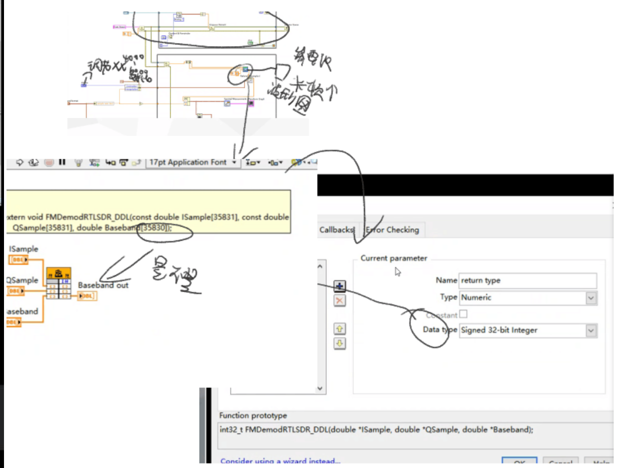
This is the waveform diagram at the beginning of the program after adding the DLL



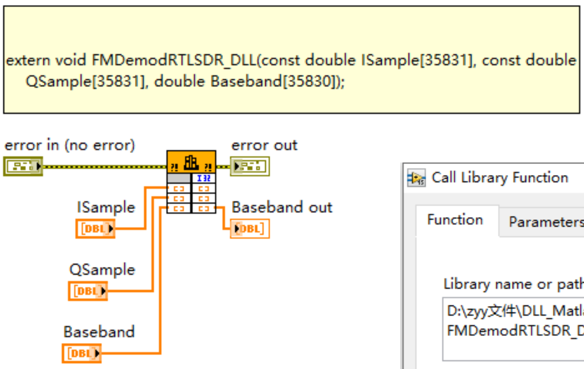
This is the adjusted waveform after adding the DLL

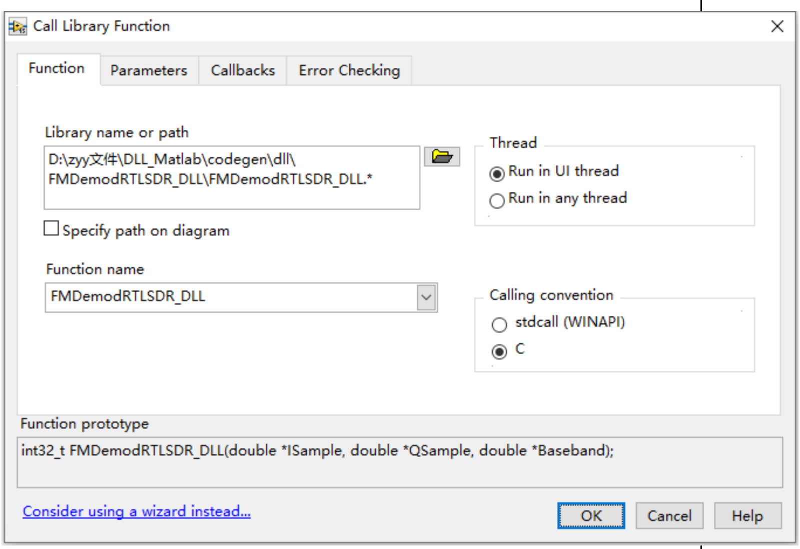
These two images, the first one is the problem we encountered after joining the DLL, the second one is the result of running our program standard.We can hear the first video sound playing continuously, and the second video sound is disconnected, similar to a phenomenon of caton.

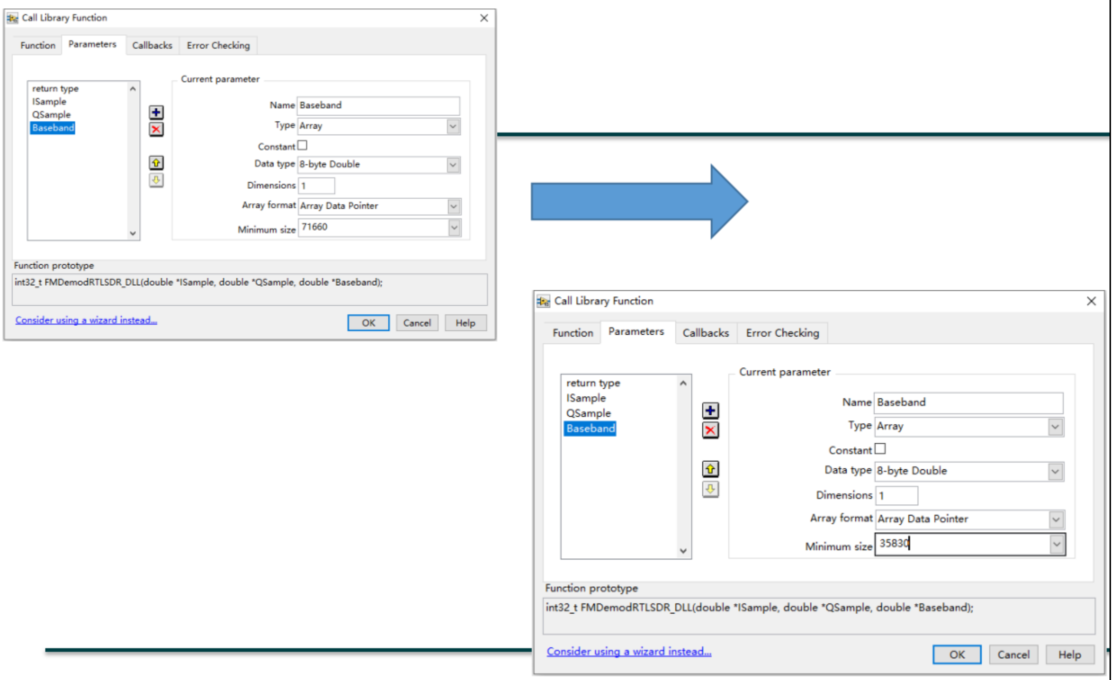




When we first encountered this problem, we also felt that it was very difficult. We approached the truth little by little by consulting classmates, reviewing notes and holding meetings.



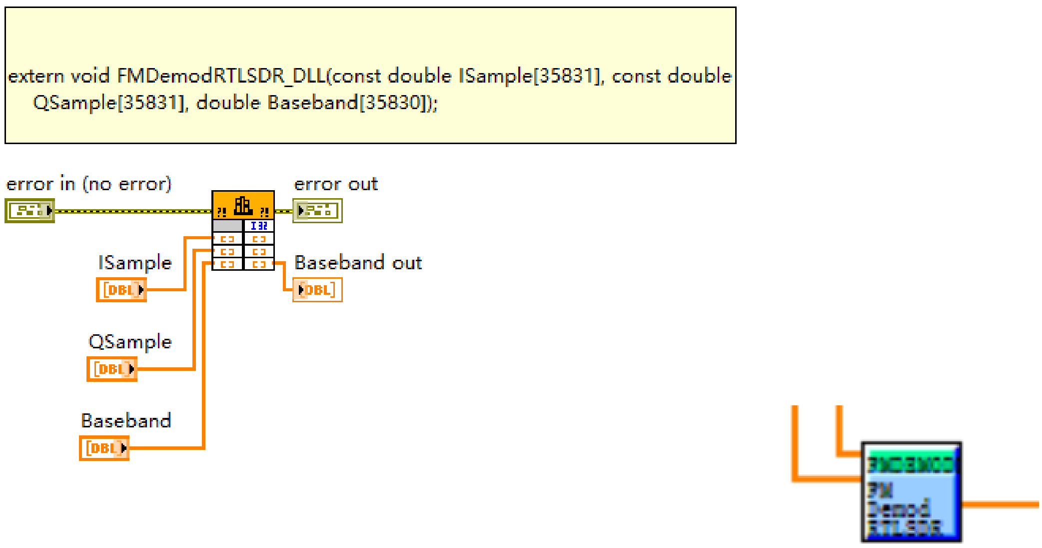




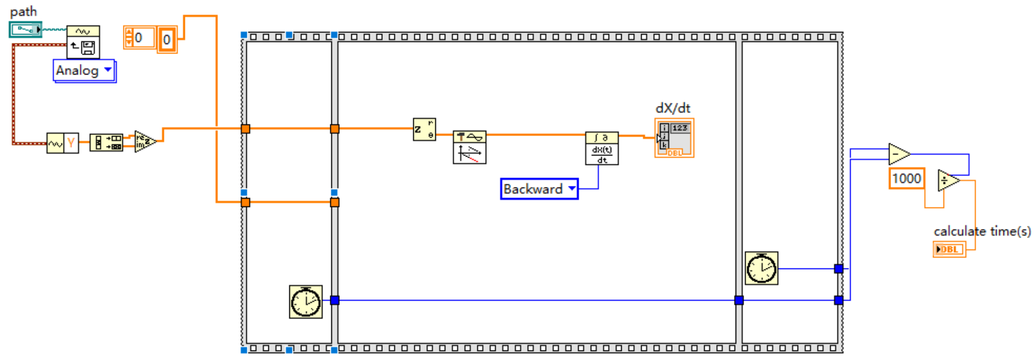
After the above adjustment, we can play the audio normally

3. DLL

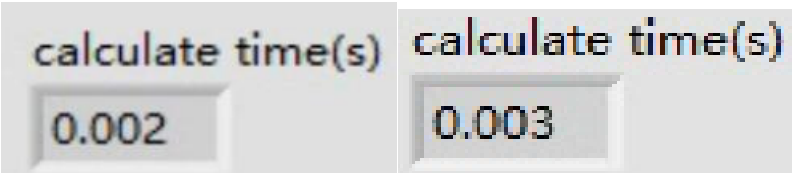
When no DLL is used, we can hear the sound of the recovery is shorter.
After using the DLL, you can reply to the sound in full.



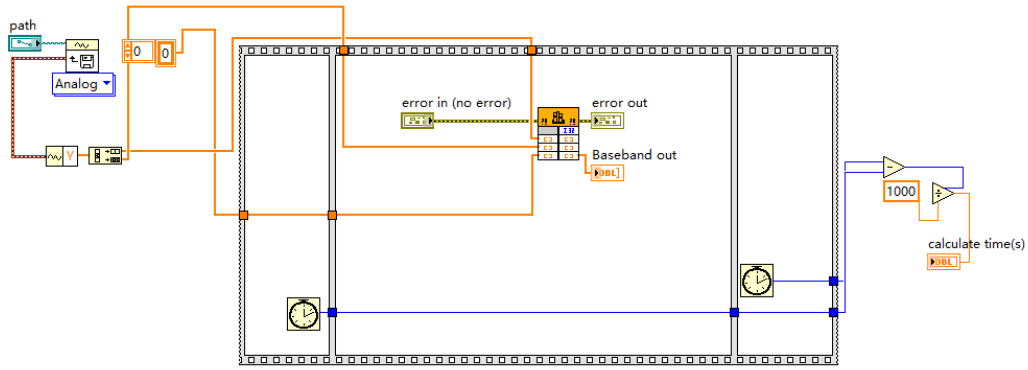
This is the DLL part



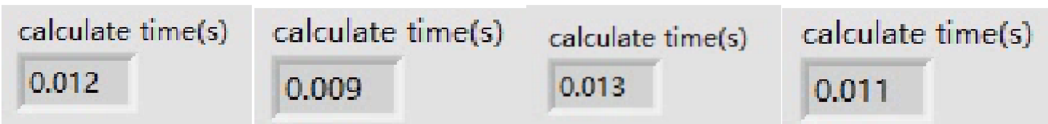
A system that tests the running time of a program（without DLL）



We can see that the running time of the DLL program is between 0.002 and 0.003



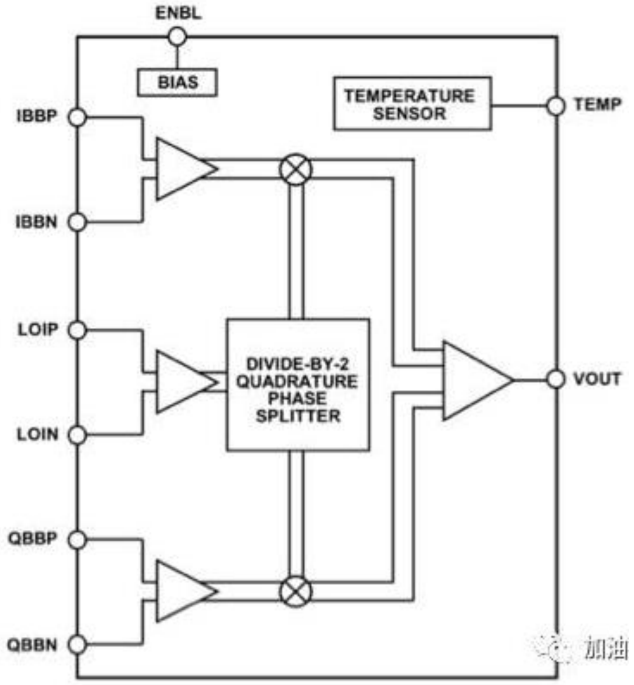
A system that tests the running time of a program（with DLL）



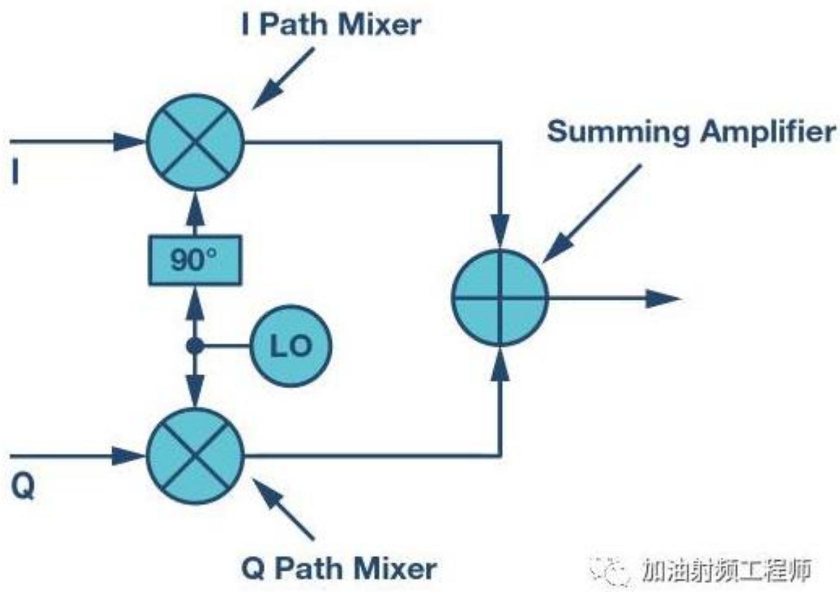
We can see that the DLL program run time is about 0.001

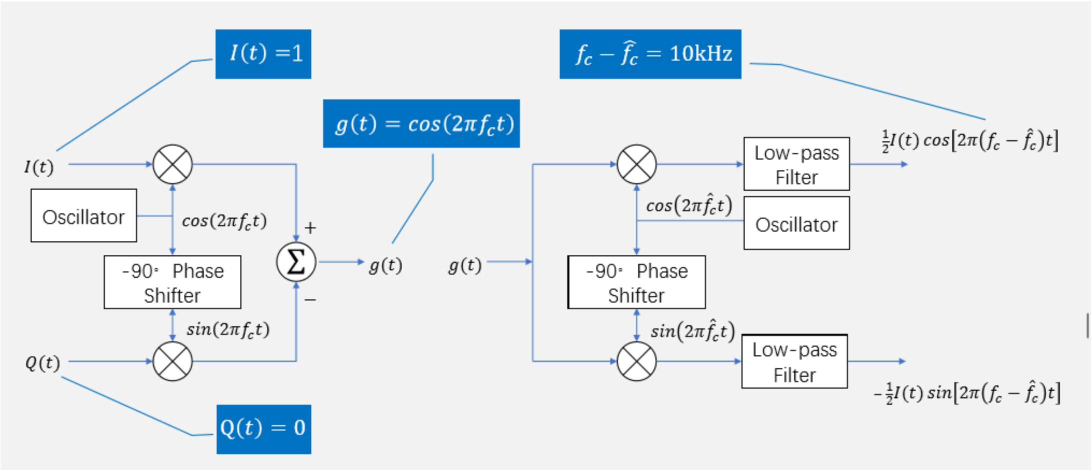
Therefore, the use of DLL modulation can greatly save the program running time on the basis of extending the playback time.This is one of the things we learned as we went through the process (more on that later)

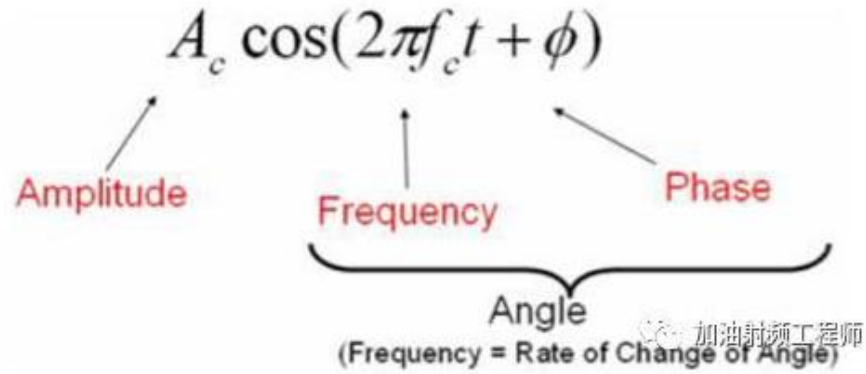
4. IQ Rate



Above is a block diagram of an orthogonal modulator from ADI.The local oscillator signal is divided into two signals with a phase difference of 90 degrees by means of the normal cross phase shift power divider. After mixing with the signals of road I and road Q roadbed respectively, the RF signal is obtained by adding them together.A more simplified block diagram is shown below.

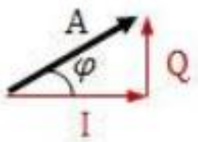






$$\cos(\alpha+\beta)=\cos(\alpha)\cos(\beta)-\sin(\alpha)\sin(\beta)$$

$$A\cos(2\pi f_c t+\varphi)=A\cos(2\pi f_c t)\cos(\varphi)-A\sin(2\pi f_c t)\sin(\varphi)$$

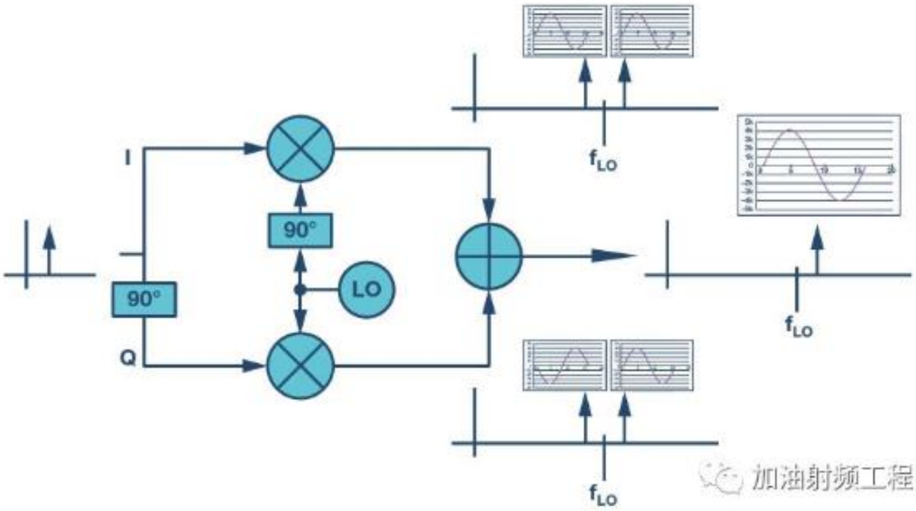


$$I=A\cos(\varphi)$$
$$Q=A\sin(\varphi)$$

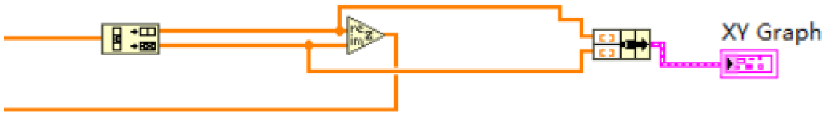
$$A\cos(2\pi f_c t+\varphi)=I\cos(2\pi f_c t)-Q\sin(2\pi f_c t)$$

where I is the amplitude of the in-phase carrier
Q is the amplitude of the quadrature-phase carrier

The first advantage of IQ signals is that they simplify the modulation processA sinusoidal signal has three variables, amplitude, frequency and phase.Modulation is the modulation of the amplitude, frequency or phase of a sinusoidal signal.With the IQ signal, you just have to change the amplitude of the IQ signal to achieve these modulation.

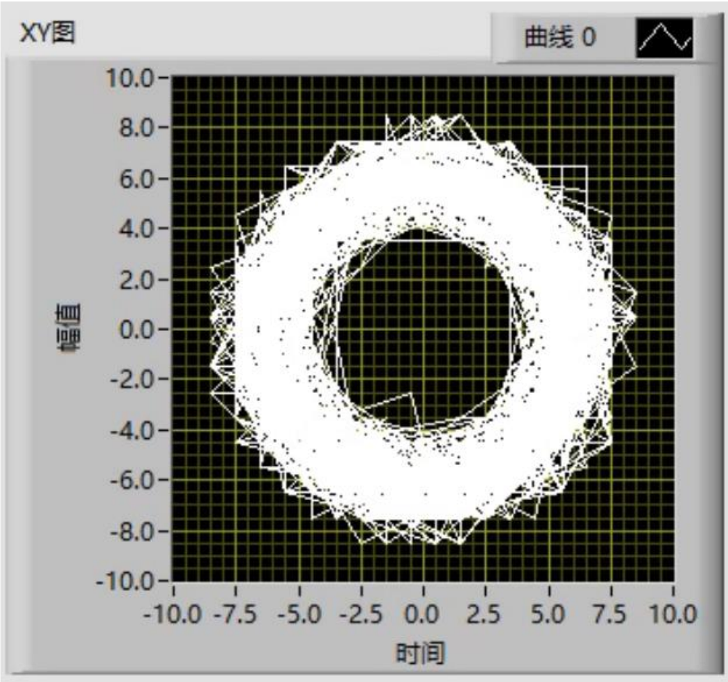


The second advantage is that the filter suppression degree can be reducedFrom the time domain analysis: suppose the input signal is $\sin(2\pi f_1 t)$, the local oscillator signal is $\sin(2\pi f_2 t)$, then the output signal is $\sin(2\pi f_1 t)*\cos(2\pi f_2 t)+\cos(2\pi f_1 t)*\sin(2\pi f_2 t)=\sin(2\pi(f_1+f_2)t)$, there is only one sideband signal, the other sideband cancel.

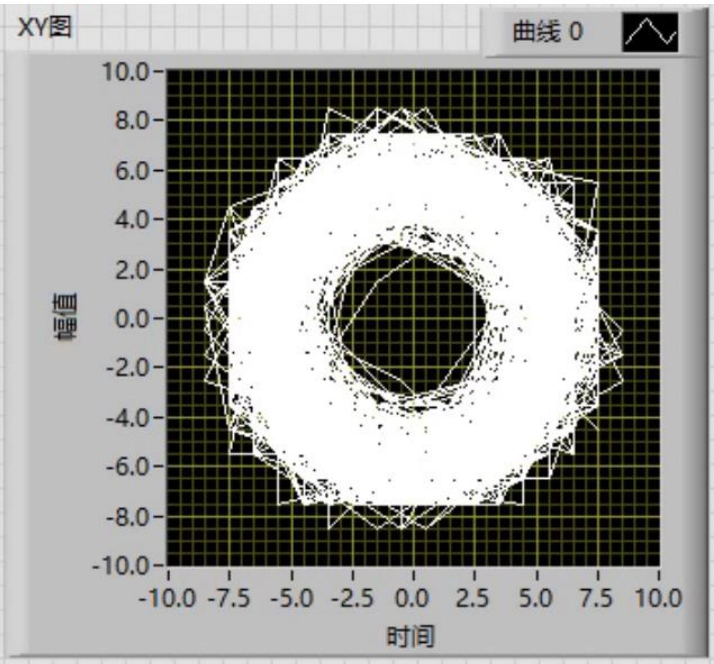


Let's draw the x-y phase diagram

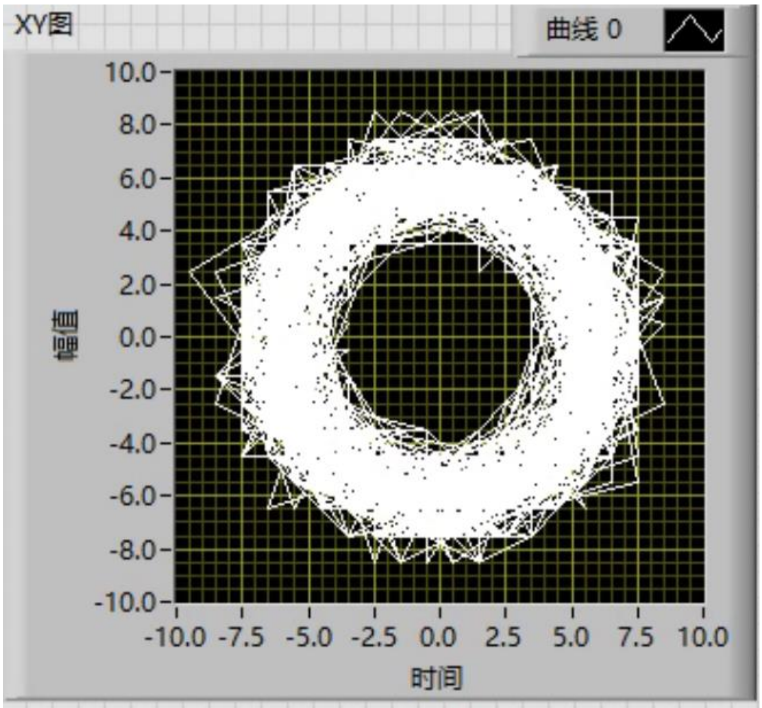
Iq=5000（Due to the difference between the IQ rate of the program and the standard answer, the result of the program is correct.）



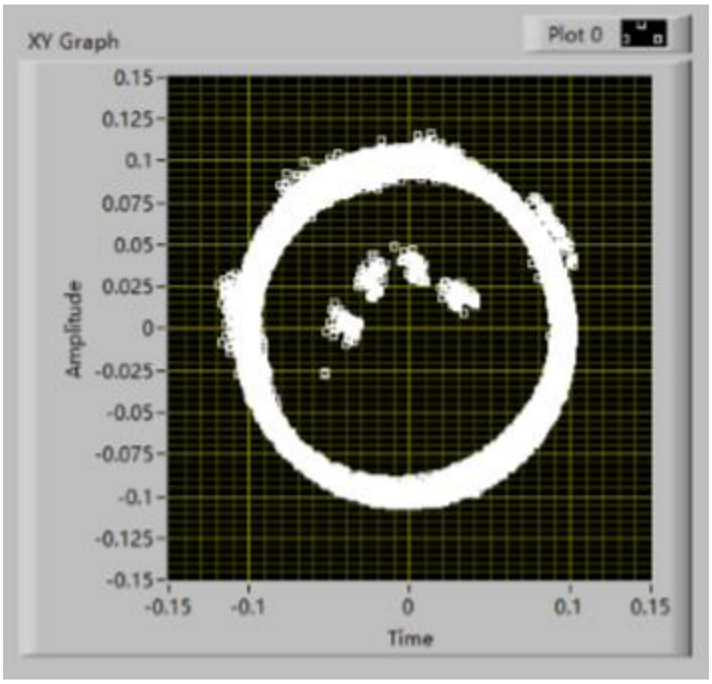
Iq=7200



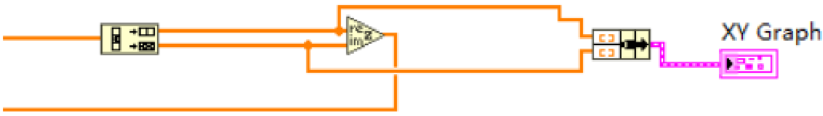
Iq=10000

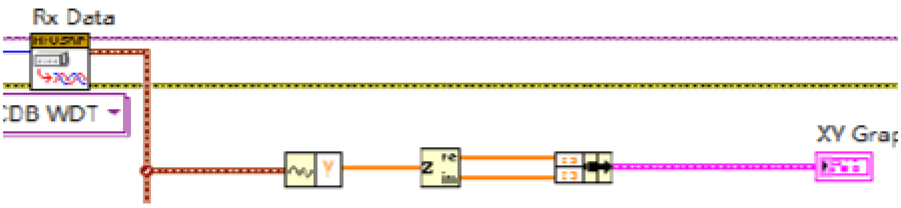


We can see that the change of IQ rate has little effect on the change of the image of the x-y phase diagram, but the change rate of the x-y phase diagram slows down when the IQ rate increases.

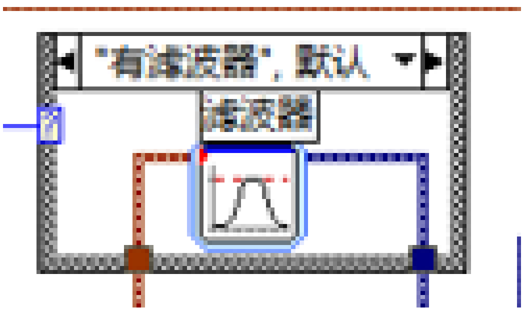


However, the image we got is different from the standard image, which we think is because the program is different from the standard program given by the teacher





5.filter

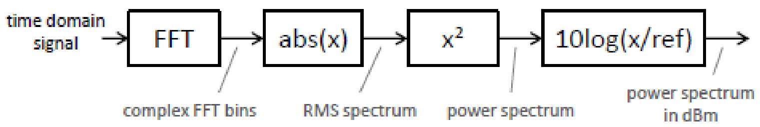


According to the music signals we produce, we can draw the following conclusions:

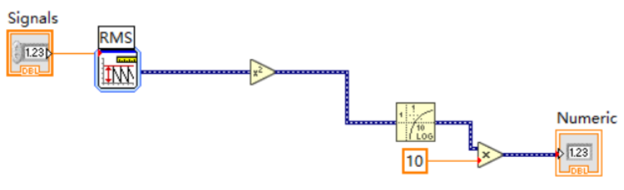
1. When we are at the optimal IQ, the filter free recovery signal is better, the music is clearer, and the sound is more similar to the radio sound, with certain electrical doping.When the filter is added, the sound signal becomes blurred, the electrical sound disappears but there is noise.
2. When IQ value is increased or decreased, the effect of the filter appears, and the greater the deviation of IQ value is, the more obvious the effect of the filter is.When Iq value deviates, the

electronic sound of the filter - free signal is more obvious and sharp.However, after the addition of the filter, the doping of the signal is significantly reduced and the sound is clearer.

Experience



According to the above schematic diagram, we designed the energy measurement module, which is shown in the figure



Form the diagram function, we get the dB power of the input signal. We use the diagram as a sub VI in transmitter. We use multiple cycles to get the average dB power of the input signal, and then we find the power of $m(t)$ is -27dB.

We can calculate the $\mathrm{SNR_O} = \frac{S_o}{N_o}, S_0 = 4\pi^2 k_f^2 p_m, N_o = \frac{8\pi^2 \eta f_m^3}{3A_c^2}$. $$\mathrm{SNR_o} = \frac{3A_c^2\beta^2}{4\eta f_m^2}$$ In this equation, $\mathrm{A_c}$ is the amplitude of carrier signal, β is modulation index $\beta = \frac{\Delta f}{f_m}$, $\frac{\eta}{2}$ is the PSD of signal, we can conclude when m(t) isn't change, if we increase Δf, we will have higher output $\mathrm{SNR_o}$. 1. The practical harvest is that through this project, we learned the advantages of two kinds of programming software in the process of DLL modulation to develop a program.We not only take advantage of the visual convenience of labview, but also take advantage of the simplicity and efficiency of matlab.This is a very creative harvest for me, and it also has some enlightenment for my future study. 2. The harvest from the theoretical level is that through this project, we basically completed a transformation process from full copy to imitation to innovation.We all know that the process of learning is copying, then imitating, and finally self-innovation.The learning of labview is the same.In this project, we completed a completely blank program. On the basis of no existing program block diagram, we gradually built the program by studying the requirements given by the teacher, explaining documents and courseware knowledge, and also completed the transformation of our learning results.

Score	

字体：英文Times new Roman；中文宋体，正文五号
文件名统一命名方式：**LabX+姓名+学号，例如：Lab1+张三+00001**
（正式报告删除此行！）