Design of virtual oscilloscope based on sound card

Bencheng Ma [a], Yaxiong Gu [b]

School of Mechanical and Electrical Engineering, Southwest Petroleum University, China

[a]mbc_8888@163.com, [b]410288153@qq.com

Abstract: This paper is based on LabVIEW to create a virtual oscilloscope with a sound card instead of a commercial data acquisition card. The waveform display, analysis, Fourier transform, sampling time and sampling rate adjustment, and amplitude of the waveform are completed. The designed oscilloscope was initially debugged, and the correctness of the block diagram was checked by LabVIEW's own detection system. Finally, the waveform was correctly displayed on the oscilloscope. Through the test, the frequency range is adjusted to between 1 and 2 to collect the signal.

Keywords: Virtual scope, Sound card, Labview.

1. Introduction

At present, the oscilloscopes on the market are mostly realistic instruments, which are cumbersome and inconvenient to carry. According to this situation, it is necessary to design a convenient method. If the oscilloscope is placed in the computer, the measurement can be more convenient. Some advanced instruments in China are still provided by other countries. Such instruments are often very expensive, and the application is also narrow. The effect of improving the effect is not great. Virtual oscilloscope is one of the commonly used electronic test instruments in the output part of the virtual instrument platform measurement and control system. It is a comprehensive instrument that can analyze and compare signals. It can use different voltage and current on the display. The virtual oscilloscope designed in this paper should design software with functions such as data reading and writing, waveform display, waveform storage and playback, parameter measurement, and spectrum analysis. As well as the front panel design, it is necessary to compare the oscilloscope map, adjustment knob, switch key and other components into the real oscilloscope form in real life. In general, it is a virtual oscilloscope with waveform display, X-axis

and Y-axis adjustment, graphics can be changed with knob adjustment, waveform amplification, and sampling frequency can be set, dual channel and other functions.

2. Overall design

2.1 Technical indicators

The virtual oscilloscope designed this time replaces the commercial data acquisition card with a sound card, so some decisive parameters will be determined by the sound card. The specific indicators are as follows:

1."5.1"channel has been widely used in various traditional theaters and home theaters,"5.1"channel is a specially designed subwoofer channel, this channel can produce a subwoofer with a frequency response range of 20 ~ 120Hz.

2. Number of samples: 24 bits; the larger the number of bits, the more data can be sampled.

3. The sampling frequency is 8 KHz, 11.025 KHz, 22.05 KHz, 16 KHz, 37.8 KHz, 44.1 KHz, and 48 KHz. According to different computers and sound card models, the frequency is different. The higher the frequency, the more times the sound sample is obtained every second. More, the better the signal quality.

4. The higher the SNR parameter, the better.

5. The lower the distortion, the better.

6, because this virtual oscilloscope is based on the parameters of the sound card to determine the performance, so the ordinary oscilloscope performance indicators cannot be pointed out, such as rise time, storage depth and so on.

7. The range of vertical division: 1mV / div - 10V / div. (1-2-5 step)

8. The range of horizontal division: 1ns / div - 1ks / div. (1-2-5-10 step)

9. Input signal frequency range: 20HZ-20KHZ.

10. Input voltage range: +40V to -40V [1].

2.2 Function of virtual oscilloscope

The virtual oscilloscope designed in LabVIEW has the following functions:

1. Collect the sound signal input by the computer and display it.

2. Waveform display and adjustment.

3. Sampling time and sampling frequency adjustment.

4. Waveform analysis Fourier transform.

5. Get the data in time.

6, single trigger, continuous trigger, pause function.

2.3 Interface design analysis

Virtual oscilloscope software in LabVIEW should have two parts: one is the virtual

panel part, and the other is the signal processing process. The virtual panel part is the user-facing part, the most surface of the software, it is the part that provides the user with the operation, and some buttons or switches that can control some functions of the oscilloscope, this part should meet some requirements of the software, and Beautiful, humanized and easy to operate. The signal processing process is to take some calculations on the signal. This includes the Fourier transform, which is the base layer of the software, by processing the data and operations, and finally sending the data to the virtual panel for display. The complete oscilloscope interface is shown in Figure 1.
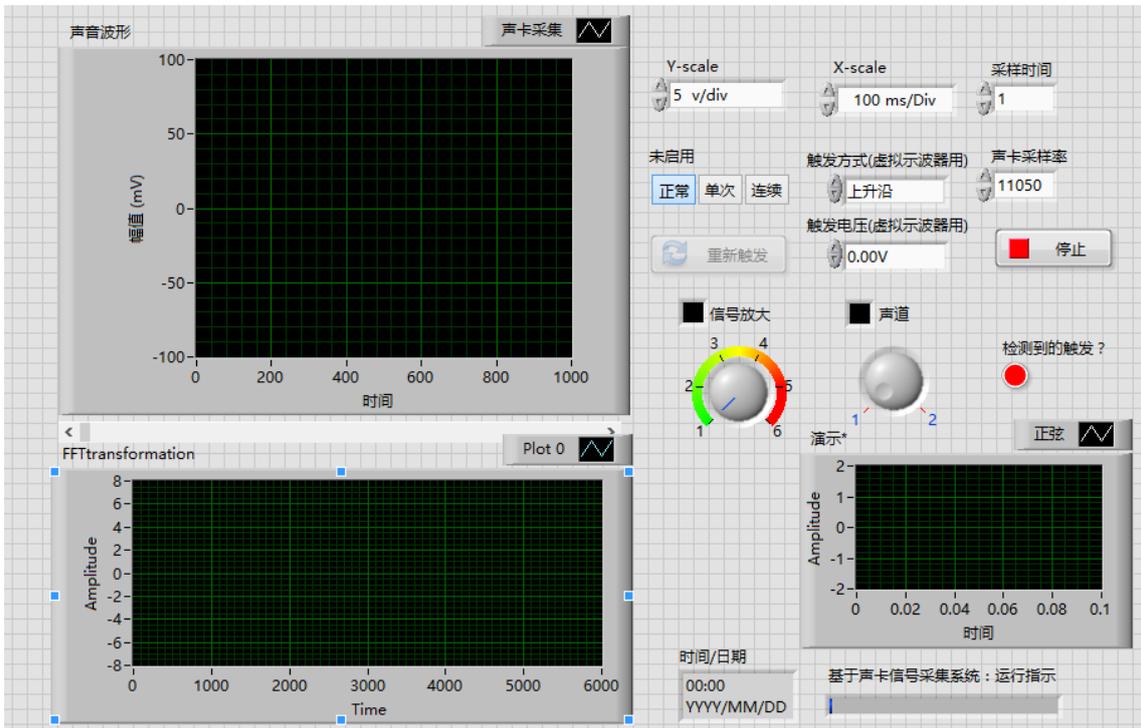

Fig. 1 Virtual oscilloscope interface design

2.4 Design analysis of Fourier transform

When the repetition period of the periodic signal is infinitely increased, the interval between adjacent spectral lines of the signal spectrum is infinitely reduced, the spectral lines are increasingly dense, and the discrete spectrum becomes a continuous spectrum. A periodic signal can be expanded into an exponential Fourier series.

$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} F(j\omega)e^{j\omega t}d\omega \tag{1}$$

This is the Fourier integral expression of the aperiodic signal *f(t)*, which is comparable to the Fourier series of periodic signals. Like the periodic signal, the aperiodic signal can be differentiated into many sinusoidal components of different frequencies. The difference is that the period of the aperiodic signal infinitely tends to be infinite, and the fundamental frequency tends to be infinitely small, so the spectrum cannot be directly used. The magnitude is made and must be made using its density function. Used on the oscilloscope is the FFT fast Fourier transform, through the Fourier transform (which can be realized by DSP or CPU faster algorithm), the signal transforms from time domain to frequency domain (also vice versa), so use Fourier transform, the waveform (time domain) image displayed on the oscilloscope can be changed into a frequency domain image, that is, a spectrogram. At this time, the Y axis is still the amplitude of the signal, but the X axis changes from time t to frequency f [2].

3. Implementation and development of virtual oscilloscope

3.1 Functional design of virtual oscilloscope

After the sound card collects the signal, use the virtual oscilloscope to obtain the data in the sound card. In the sampling module of the virtual oscilloscope, select the sound acquisition control, select whether to analyze the original signal, and firstly frequency the original signal. Domain analysis, which uses FFT, the other is to operate the signal amplification, such as attenuation, filtering, addition and subtraction, etc., and finally display on the display, this is the result of the oscilloscope running once, but if you want to keep showing the screen displays a clear waveform and runs indefinitely. Next is the horizontal indexing and vertical indexing. What needs to be considered is how to control the X and Y axes. In essence, it is the control of the Scope interface properties. In LabVIEW, there are two "X.scale" and "Y.scale". Constants extracted from the x-axis and the y-axis. By changing these two constants, you can scale the X-axis and the Y-axis. Creating a new project first. After opening LabVIEW, click Create Project, a dialog box will pop up, which is the overall file structure of the project. Then right click on the mouse and select Create a new VI. Select two waveform controls in the front panel, one to display the waveform and one to display the Fourier transform Waveform. The sound acquisition and sound acquisition controls have three parameters to set, one is the sampling time, and the other is the sound card sampling rate. This must be set, the control itself will not Settings. To set a data, you need a numeric control and a numeric knob. The numeric controls are available on the front panel and on the block diagram. In the block diagram, the sampling time is connected to the duration of the control. The sound card sampling rate is connected to the sampling rate, and the channel is connected to the number of channels. After the

sound acquisition is completed, the data is calculated and displayed. This paper uses the combination of the While structure and the event structure to wrap the connection of the previous step. The While structure determines the number of loops according to the Boolean quantity. After the end of the loop, the Boolean is used to determine whether to continue execution. The event structure is used for corresponding user interface operations, and may include multiple branches, one for each independent event handler. The event structure cannot be used separately for various events. It must be used together with the loop structure. Therefore, the loop structure is placed outside, the event structure is placed inside, and the timeout time is set to 5. The timeout period means that the execution timeout is performed when there is no trigger event after a certain period of time event. Then the sound acquisition and three controls are placed, the signal is taken out by the sound acquisition control, there are two steps, first, the signal is amplified, because the signal obtained by the sound card is very small, must pass 2-3 times Magnification can make the performance on the oscilloscope obvious. This step is to multiply the original signal and the magnification to obtain the amplified data. In the program panel, select "Function—Value—Multiply" and create a knob to make the knob range between 1-6. Connect the signal obtained by the sound to this knob to the two segments. The resulting signal is the amplified signal and connected to the oscilloscope. As shown in Figure 2.
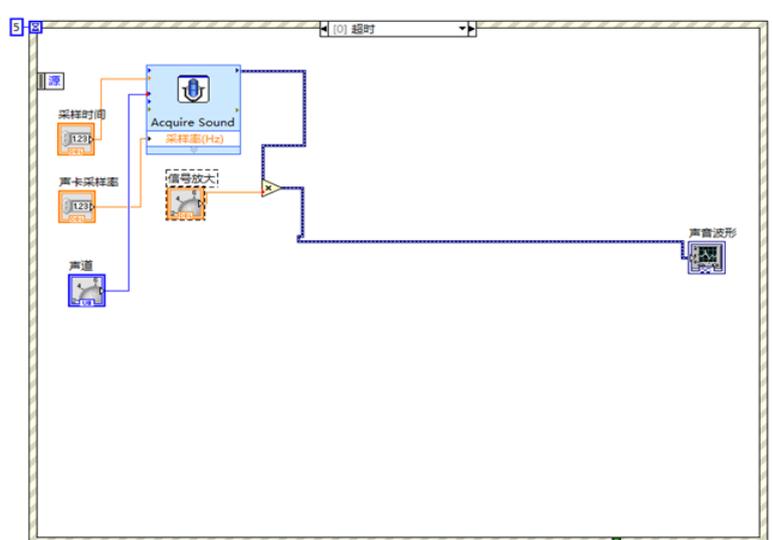

Fig. 2 Sound acquisition block diagram

In the second step, the original signal is converted into an FFT, that is, a Fourier transform, and the waveform Fourier transform analysis is displayed. It should be noted that the input of the FFT transform is static data, but the sound collection is dynamic, so the data needs to be dynamically converted to static. This requires the use of another control "Function - Express - Signal Operation - Convert from Dynamic

Data". Connect it to another waveform.

3.2 Vertical indexing and horizontal indexing

Here you need to use the sub VI in LabVIEW, create a new VI in the project, named X ruler configuration and Y ruler configuration. This requires the use of the index function, create a sub VI, and then create a handle, LabVIEW refnums also point to specific data structures. In the front panel, select the order "Control - Ref. handle - application refnum", then right click on the refnum and change it to the waveform refnum. Specifically select "Select VI Server Class - General - Graphic Image - Control - Graphic Chart - Waveform - Waveform". Then change the properties of the front panel and the waveform chart. When the property node is changed on the front panel, a reference Vi is needed. Then, the delay of the front panel is used to update the panel, and finally the waveform is updated. The refnum is connected to each attribute node of the waveform graph, and has a drawing boundary, a Y-axis or X-axis magnification factor, a Y-axis or X-axis range, a Y-axis or an X-axis label, and finally outputs to the waveform graph refnum output. As shown in Figure 3
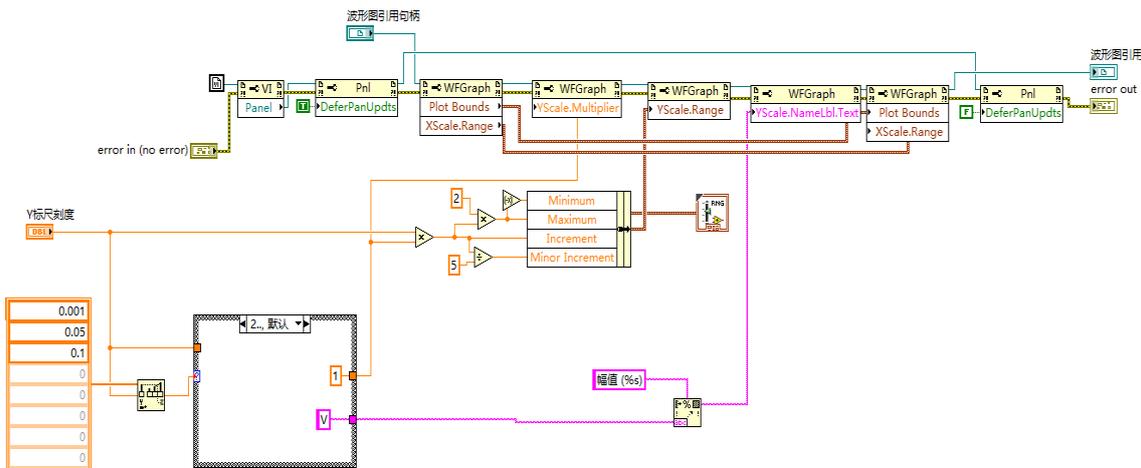

Fig. 3 Index conversion block diagram

The program below in Figure 3 mainly connects the data of the Y scale to the attribute node, so that the oscilloscope can reflect the manual operation. Edit the input and output of the sub VI in the upper right corner, just like the pins of the chip. Next, refer to this VI, select "Function - Select VI" on the block diagram, but this sub VI cannot be placed in the loop structure, you must create another structure, tiled sequential structure data flow Different from the data flow of other structures. When all data wired to the frame is available, the tiles of the tiled sequential structure are executed in order from left to right. After each frame is executed, the data is passed to the next frame which is the input of the frame may depend on the output of the other frame.

After placing two sub VI, you need to create a reference to the waveform diagram, connect to the waveform reference handle in the sub VI, connect the value constants and values of the X and Y axes to the sub VI, and connect the output of the sub VI to on the loop structure. Not only that, adding two events in the event structure in the loop structure is the change of the X axis, and the other is the change of the Y axis. In both event structures, the reference of the waveform diagram is connected with the respective sub VI. The program corresponding to the control of the respective drop-down list is connected to the sub-VI, and the programming of the changes of the X-axis and the Y-axis is completed.

3.3 Trigger

In order to synchronize the scan signal with the measured signal, some conditions can be set to continuously compare the measured signal with these conditions. The scan is initiated only when the measured signal satisfies these conditions, so that the frequency of the scan is the same as or identical to the measured signal [3].There are no specific controls in LabVIEW, so you need to do a sub VI to simulate the triggering process. The trigger is divided into four stages, finding the waveform, buffering the waveform, intercepting the waveform, and outputting the waveform. Firstly, find the location of the trigger. If it is not found, the loop continues. If it is found, the buffer waveform is saved. The waveform refresh rate is slowed down, and the waveform just found the trigger position is judged once. If the trigger position and the number of samples change. If yes, continue to buffer the waveform. If there is no change, the interception will be performed. The intercepted waveform will be output. If the starting position and the number of samples have not changed during the interception, the waveform is returned to the first step, which is the detection trigger position; if it changes, an empty array is output, so that the first step is to find the trigger position again. Trigger is also divided into single trigger, continuous trigger and normal three selection. Single trigger is to press the single trigger when outputting the waveform, then output the waveform without looping to continue outputting the waveform. If not, loop to the first step is to find the trigger location. Continuous triggering is judged in the trigger assistant. In the triggered sub VI, you need to select two basic structures, one is the conditional structure and the other is the While loop structure. In the While loop structure, one of the more important points is the shift register. The terminal consists of right-clicking on the border and selecting Add Shift Register to add it. The right terminal (the rectangle with the up arrow) stores the data after each iteration, and the shift register moves the stored data from the last iteration to the left terminal (the rectangle with the down arrow) at the beginning of the next iteration. The shift register can store any data type, including numbers, booleans, strings, and arrays, but

the data connected to the same register terminal must be of the same type. The type of shift register is the same as the type of object data that is first connected to one of its terminals. Then create a waveform constant, and create a sample number to obtain the number of samples of the waveform in the loop. As long as the number of samples and the button of a single trigger are pressed, the condition is true, and the waveform is input to in the control of waveform trigger detection, as long as the trigger is detected, the waveform is connected to the shift register. Step one to the second step to buffer the waveform.
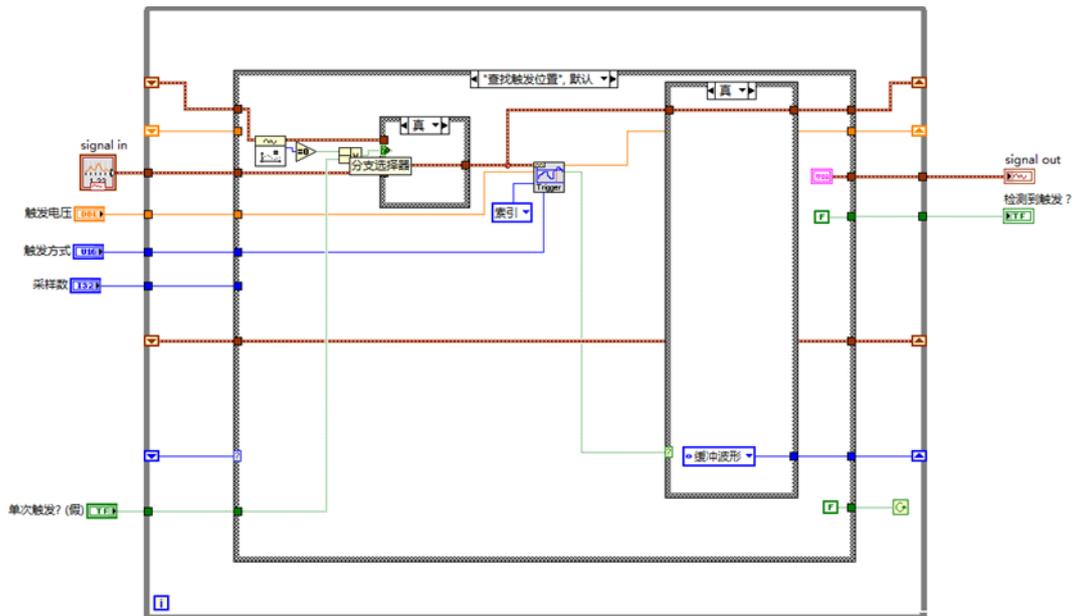
Fig. 4 Find the trigger location block diagram

In the buffer waveform, the waveform in the loop is added to the input waveform, and then the number of samples of the waveform is detected, and the number of samples is subtracted from the trigger position of the previous step. When the value is greater than or equal to the set sample value, the next step is performed.

When the waveform is intercepted, not only the waveform is acquired, but also the waveform is judged again to see if the number of samples changes. If there is no change, the loop is passed to the first step. If it changes, an empty array is given, so that the first step is to carry out, you need to use a small control, called the acquisition of the waveform subset. The final output waveform is very simple to connect the waveform just output to the output waveform, and the rest continues to loop. The final designed block diagram is shown in Figure 6
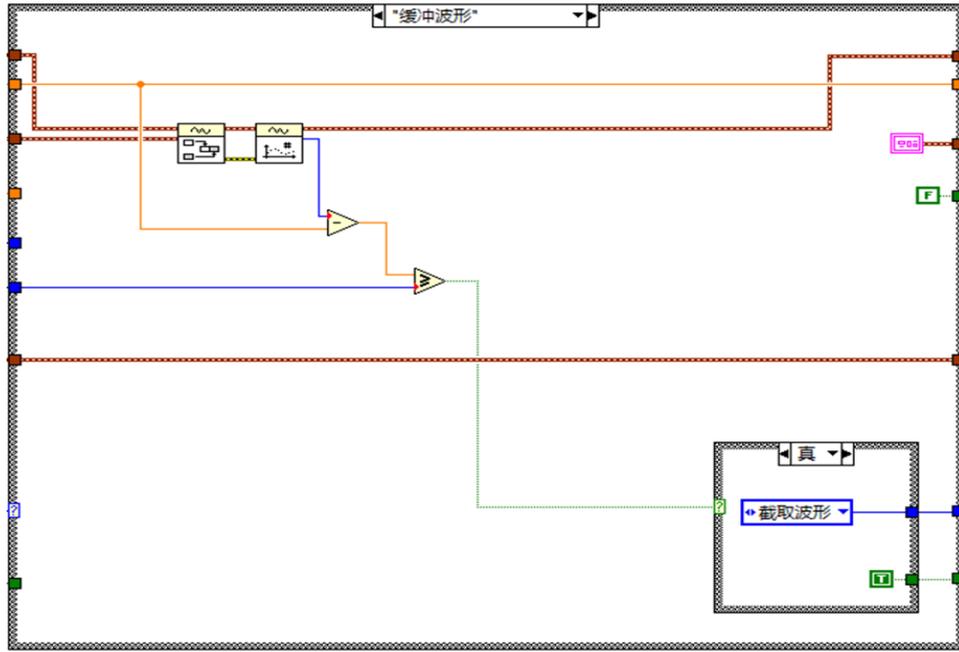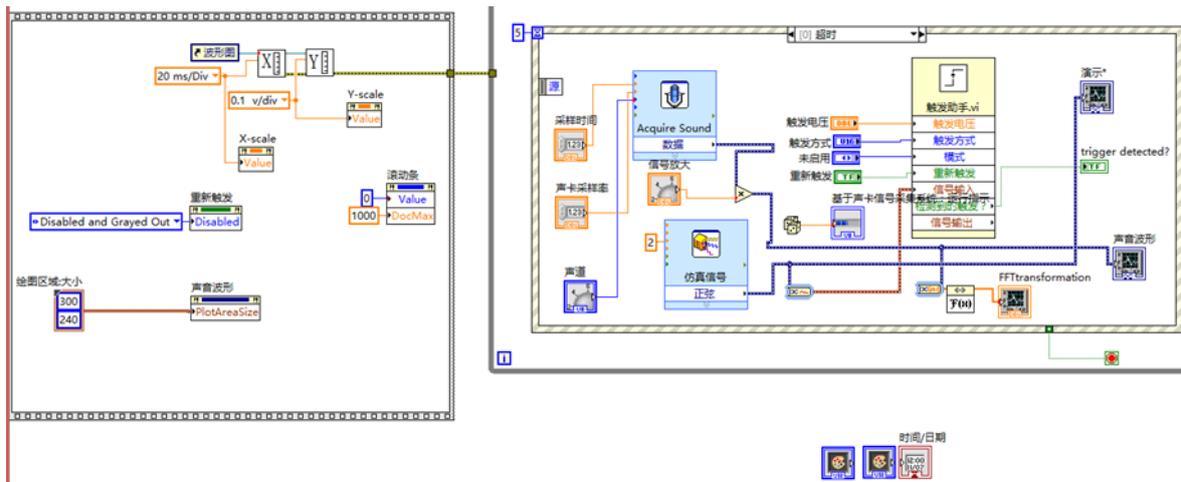
Fig. 5 Buffered waveform block diagram

Fig. 6 Complete block diagram

4. Virtual oscilloscope program debugging

The first thing to debug is the function of the two sub VI of the X axis and the Y axis. After running the program, the range of the X axis is 0-100, and the range of the Y axis is -0.2-0.2, which corresponds to the value of the Y axis on the panel. 0.1V/div, the value of the X axis is 20ms/div. Changing the X and Y axes to 200ms/div and 1V/div, the range of the X axis on the waveform is changed to 0-1000, and the range of the Y axis is changed to -2-2. Then you need to debug the knob of the signal amplification. After starting the program, it is found that there is no waveform on the

waveform, but the microphone has the signal to receive the sound. This means that the sound signal is too small, you need to zoom in, adjust the knob of the signal amplification, and adjust the knob. After 1.7768, the waveform appears. When it is adjusted to 3, the waveform is amplified and can receive a smaller sound signal. The third test is the sampling time. The smaller the sampling time is, the less data is sampled per unit time. The faster the data is displayed, the larger the sampling time is. The more data is sampled per unit time, the slower the data is displayed. The starting sampling time is 1, if the setting is 0.5, the maximum data in 500 time units is displayed. If you zoom in on the X axis, the situation in Figure 7 will appear.
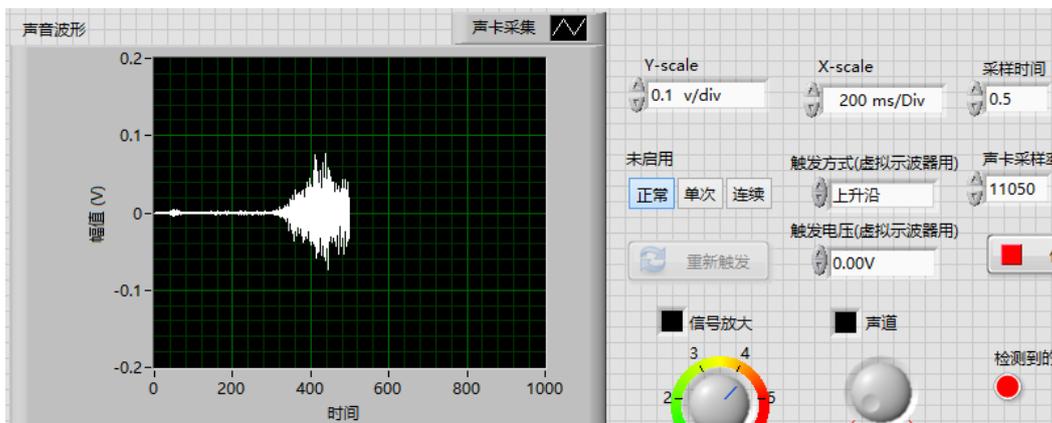

Fig. 7 Sampling time setting

Finally, we must change the channel and spectrum analysis test. The spectrum analysis test mainly depends on whether it can be displayed. After the signal appears, it can be displayed following the signal display. The test result is shown in Figure 8.

In order to distinguish the virtual oscilloscope from changing the data of the channel, the second channel is not used for spectrum analysis, only the waveform is displayed. If the waveform is not distinguished, the waveform is the same, because the channel of the sound card is divided into left channel and right channel. The final rendering is shown in Figure 9.
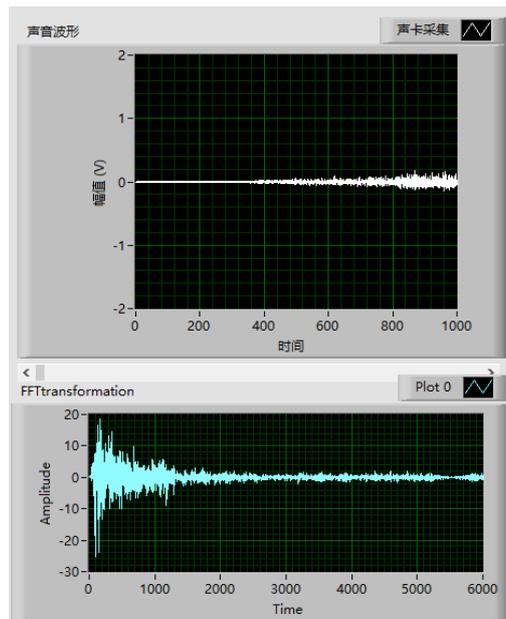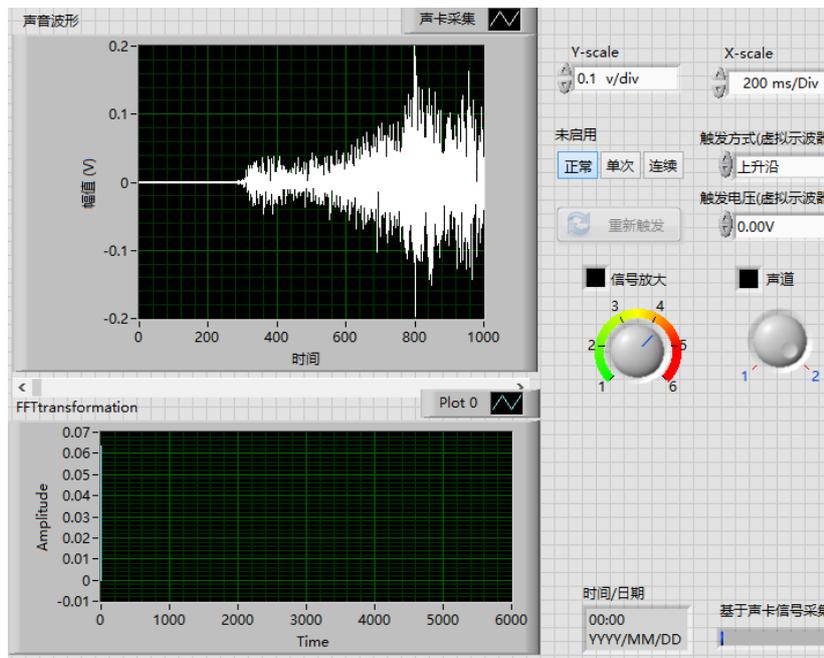
Fig. 8 Spectrum analysis test

Fig. 9 Channel change effect diagram

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