Lab 2：DSB-SC modulation

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Introduction

1. DSB-SC modulation

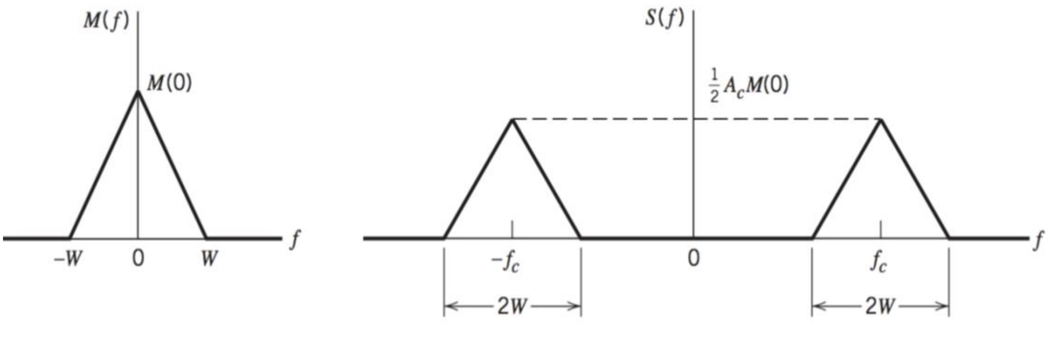
Modulation

$$
\begin{aligned}
s(t) &= c(t)\,m(t) \\
&= A_c \cos(2\pi f_c t) m(t)
\end{aligned}
$$

m(t) is the message signal, and c(t) is the carrier signal which usually use cosine signal. They both consist the mixed signal s(t).

$$S(f) = \frac{1}{2}A_c[M(f-f_c)+M(f+f_c)]$$

After Fourier Transforming, S(f) is showing above. In this equation, we must make sure f_c>2*f_m according to sampling theorem.



(a) (b)

Demodulation

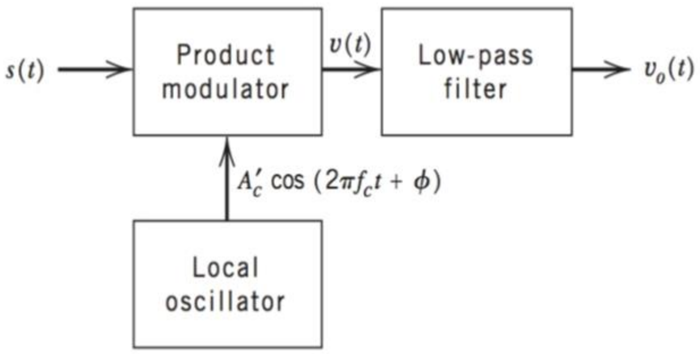


FIGURE 3.13 Coherent detection of DSB-SC modulated wave.

$$
\begin{aligned}
v(t) &= A_c' \cos(2\pi f_c t + \phi) s(t) \\
&= A_c A_c' \cos(2\pi f_c t) \cos(2\pi f_c t + \phi) m(t) \qquad\qquad (3.18) \\
&= \frac{1}{2} A_c A_c' \cos(4\pi f_c t + \phi) m(t) + \frac{1}{2} A_c A_c' \cos\phi \, m(t)
\end{aligned}
$$

Multiple a cosine signal with same f_c (frequency), and go through a low pass filter.

$$
v_o(t) = \frac{1}{2} A_c A_c' \cos\phi \, m(t) \qquad\qquad (3.19)
$$

Then we get m(t) multiple a coefficient.

$$
P_{\mathrm{DSB}} = \overline{s_{\mathrm{DSB}}^2(t)} = \overline{m^2(t)\cos^2 \omega_c t} = \frac{1}{2}\overline{m^2(t)} = P_s
$$

Modulation efficiency is 100%, and DSB modulation save the power of carrier signal. But it can’t demodulate through envelope. It needs coherent demodulation.

2. Coherent Demodulation

Coherent demodulation, also known as synchronous detection, applies to the demodulation of all linearly-modulated signals. The key to coherent demodulation is that the receiver recovers a coherent carrier that is strictly synchronized with the modulation carrier.

Coherent demodulation refers to the use of a multiplier to input a reference signal that is coherent with the carrier frequency (same frequency in phase) and multiplied by the carrier frequency.

The diagram is showing in DSB modulation

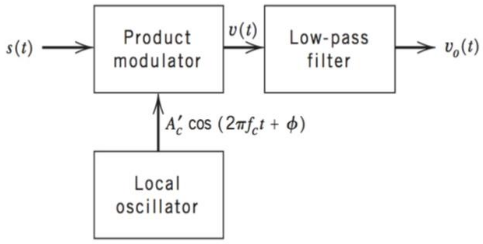


FIGURE 3.13 Coherent detection of DSB-SC modulated wave.

3. Carrier Recovery and Law of squares

Coherent demodulation requires the carrier to be restored at the receiving end, and this coherent carrier must be in phase with the modulated carrier in the received signal in order to perform coherent demodulation in the demodulation. The recovery of the coherent carrier is a prerequisite for coherent demodulation. Recovering a carrier at the receiving end that is at the same frequency and in phase with the modulated carrier is the so-called carrier synchronization problem.

Nonlinear transformation – filtering

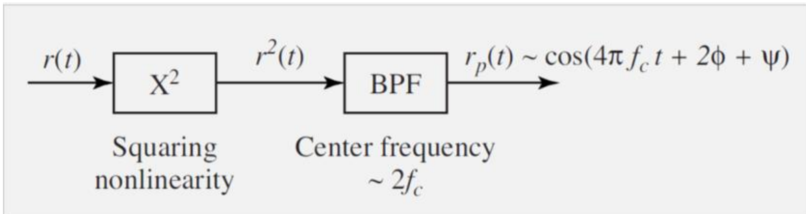
The received two-sided band modulated signal that suppresses the carrier can generally be expressed as

$$S(t) = A(t)\, cos\,[2\pi f_c t + \varphi(t)] + n(t)$$

For the M phase shift keying signal MPSK, a ring to the power of M can be used, and the box diagram of the ring to the power of M is shown in Figure 4.1. For example, the 2PSK signal has only 0 and π two modulation phases, and the square ring can be used to multiply the phase shift keying in the signal to 2π, thereby eliminating or greatly suppressing the modulation phase change, and the output of the square law device used in the square ring (or frequency doubling ring) is

$$e_1(t) = s^2(t) = \frac{1}{2}A^2(t) + \frac{1}{2}A^2(t)\, cos\,[4\pi f_c t + 2\varphi(t)]$$

After low pass filter, we get [A(t)]^2. Use sqrt(), we can get A(t),out message signal.And we can get a carrier signal with 2f_c frequency multiple 1/2*A(t)^2, so go through a band pass filter(center freq 2f_c), we can find the index of max power. This is the frequency of double f_c+f_offset. According to this frequency, we can recover the carrier signal..

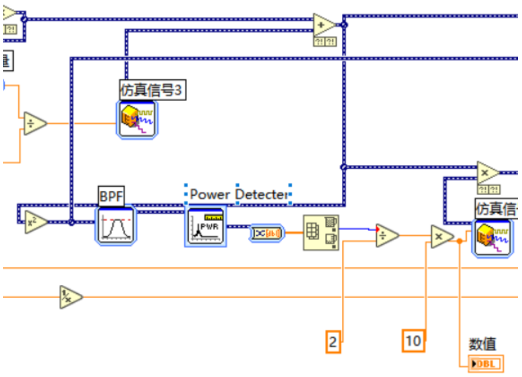


4. Phase Locked Loop

The working principle of the phase-locked loop is to detect the phase difference between the input signal and the output signal, and convert the detected phase difference signal into a voltage signal output through the phase detector, and form the control voltage of the voltage-controlled oscillator after filtering by the low-pass filter, control the frequency of the oscillator output signal, and then feedback the frequency and phase of the oscillator output signal to the phase detector through the feedback path.

During the operation of the phase-locked loop, when the frequency of the output signal proportionally reflects the frequency of the input signal, the output voltage maintains a fixed phase difference with the input voltage, so that the phase of the output voltage and the input voltage is locked.

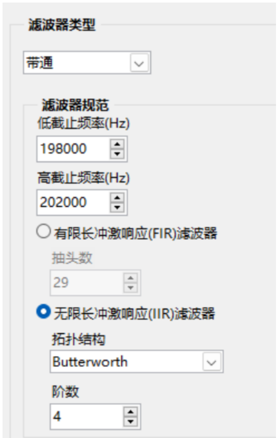
LabVIEW Express Module



1. Frequency multiplier module

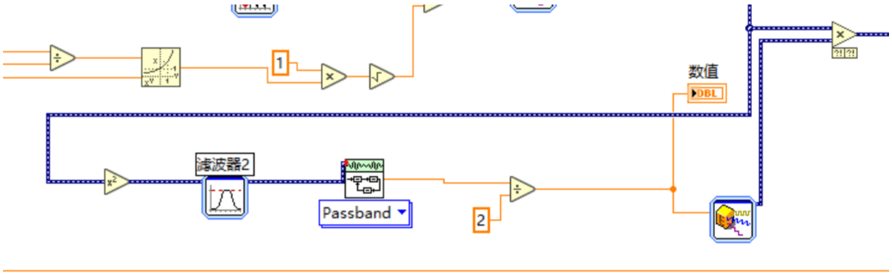
We use function x^2 to make the mixed signal (with noise) square. And then go through the band pass filter, Power Detector to find the index of the max power. The index is the frequency of double f_c + f_offset. Of course, the frequency multiplier needs to multiply by ten at the end, depending on the value of the sample rate and the number of samples (the filter filters a certain number of samples). So that we get the carrier signal from the modulation signal.

2. Band-pass filter module



Because we set f_offset 200Hz, so we need to make the band wind to 2*400Hz at least. And we set the butterworth filter's order to 4 to make the time delay lower.

3. Phase Locked Loop module



After squaring our signal, and go through band pass filter. We use Phase Locked Loop module to input the signal after BPF, and get the double frequency of f_c+f_{offset}. So, we recover the carrier signal.

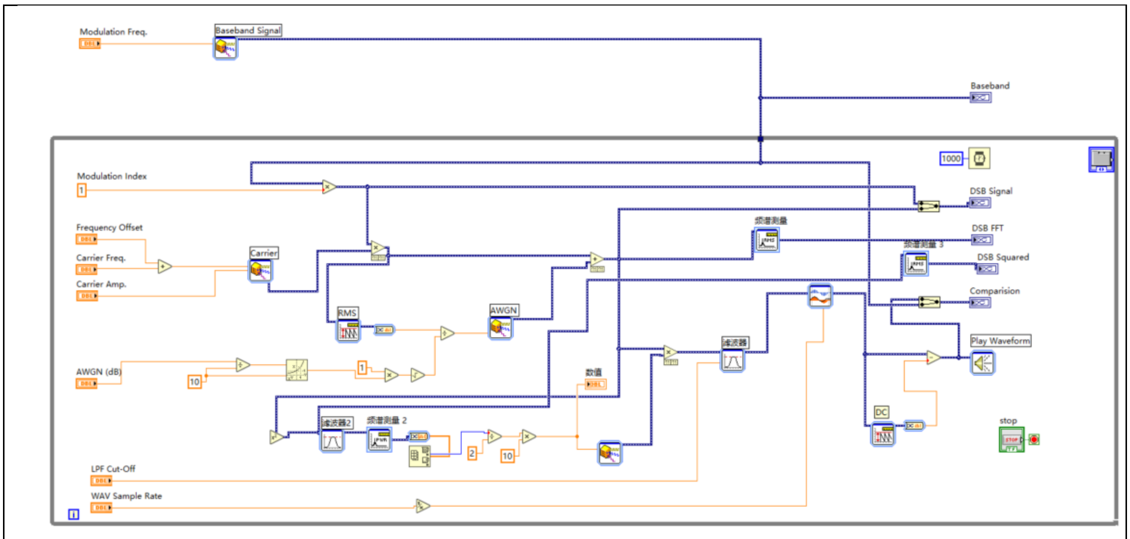
Lab results & Analysis:

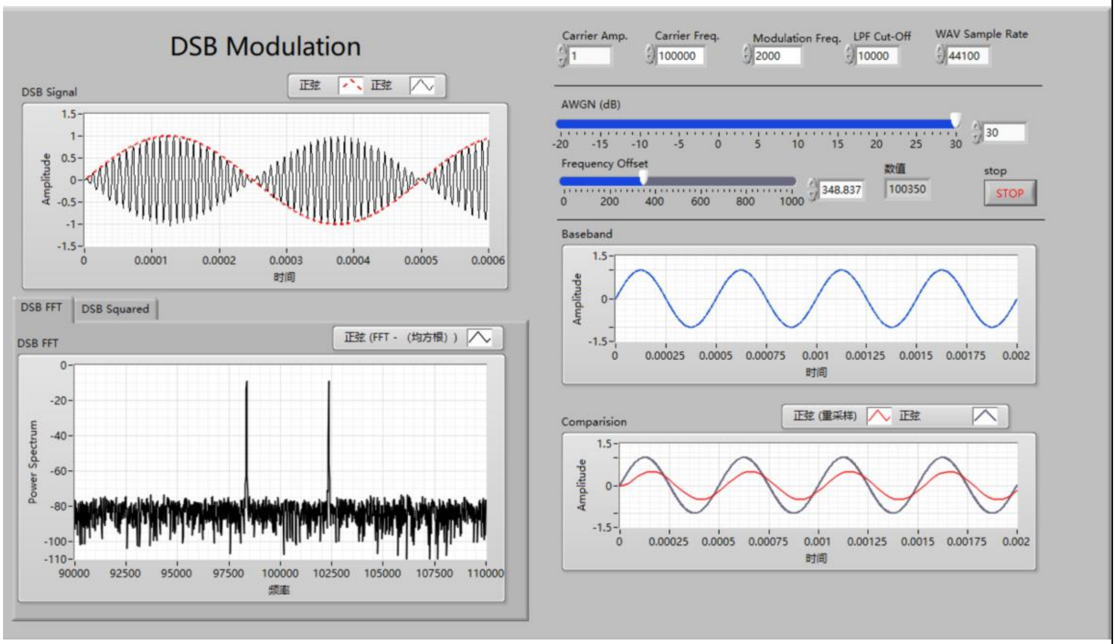
1.Single-tone signal and music signal test

First of all, we use the method of DSB modulation and demodulation to carry out a single-tone test, using the single-tone test to check and reflect the DSB modulation and demodulation procedures have no problems. In practical applications, the receiver often does not know the exact number of the carrier frequency, so we can solve this problem in two ways. First, we can adopt the square law method, that means that the frequency multiplier is used to square the modulated signal. After this operation, the spectrum of twice the message signal will be moved to $2f_c$. Therefore, after passing a band-pass filter, the spectrum component can be obtained, and then the spectrum information can be obtained by using the spectrum measurement module, and then converted into an array. The index corresponding to the maximum value of the array, namely $2f_c$, is extracted and then be divided by 2. At the same time, since the sampling number is one tenth of the sampling rate, it needs to be multiplied by 10. The exact carrier frequency is obtained. The second method is to obtain the estimated carrier frequency by using phase-locked loop method to obtain the exact frequency after frequency multiplier.

In the experimental simulation, we use the setting of frequency offset to offset the original carrier frequency, and estimate the carrier frequency by using frequency doubler and phase-locked loop respectively, and present the estimated carrier frequency to compare the estimated effect.

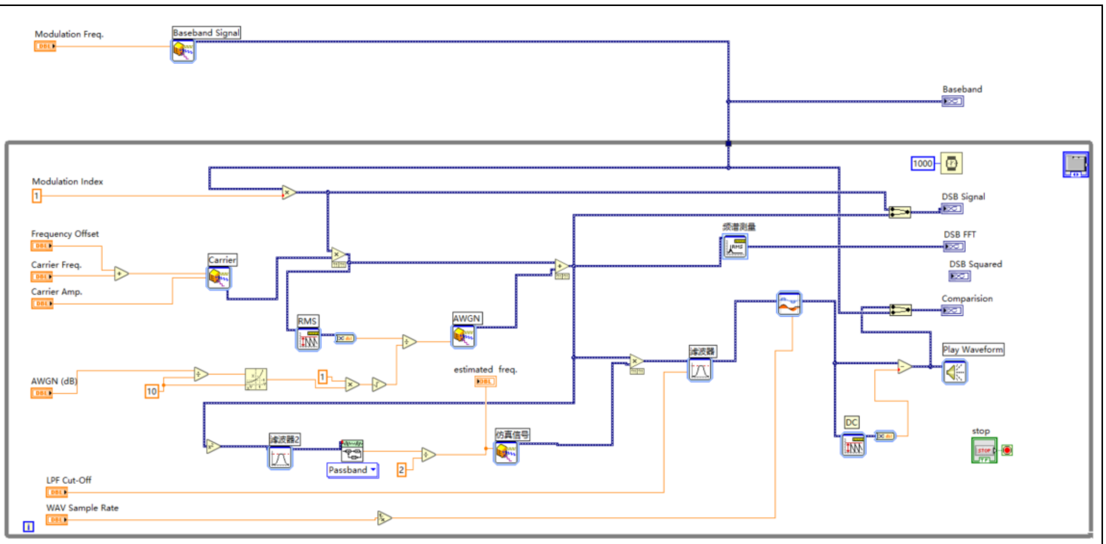
The program block diagram and simulation results of single-tone test with frequency doubler and spectrum component acquisition are as follows:

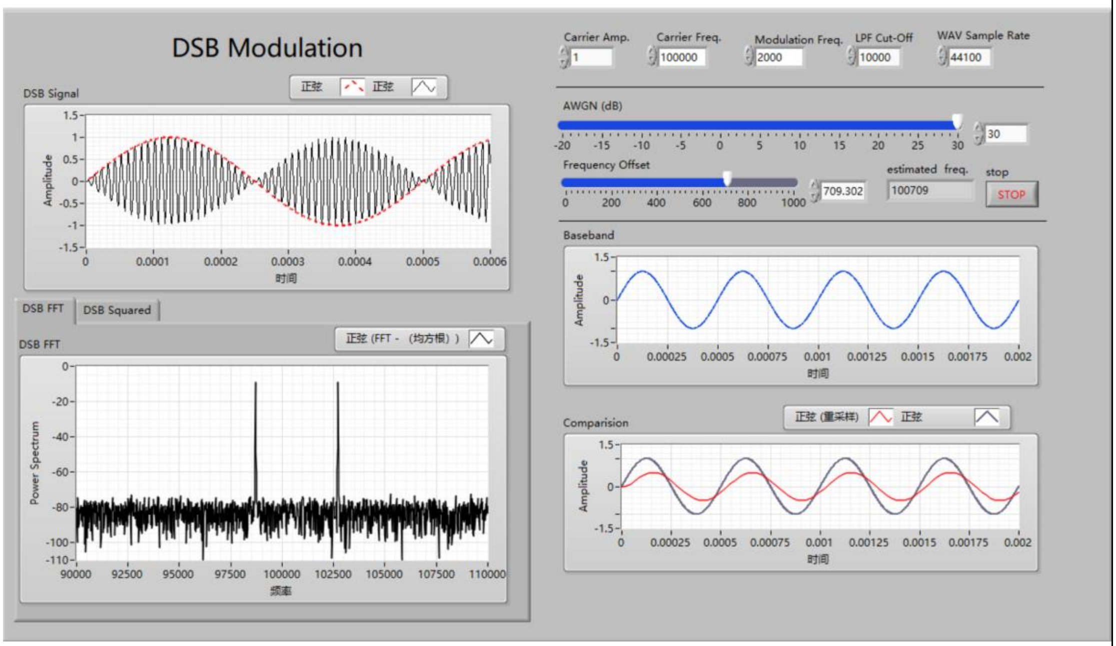




By observing the simulation results, it can be found that the carrier frequency after the estimated frequency offset obtained by the method of obtain the frequency spectrum using the index of the max value of the array after frequency doubling with frequency doubler is more accurate, and the difference with the correct result is about 2HZ.

The program block diagram and simulation results of single-tone test by using the frequency doubler and using the phase-locked loop operation method are as follows:

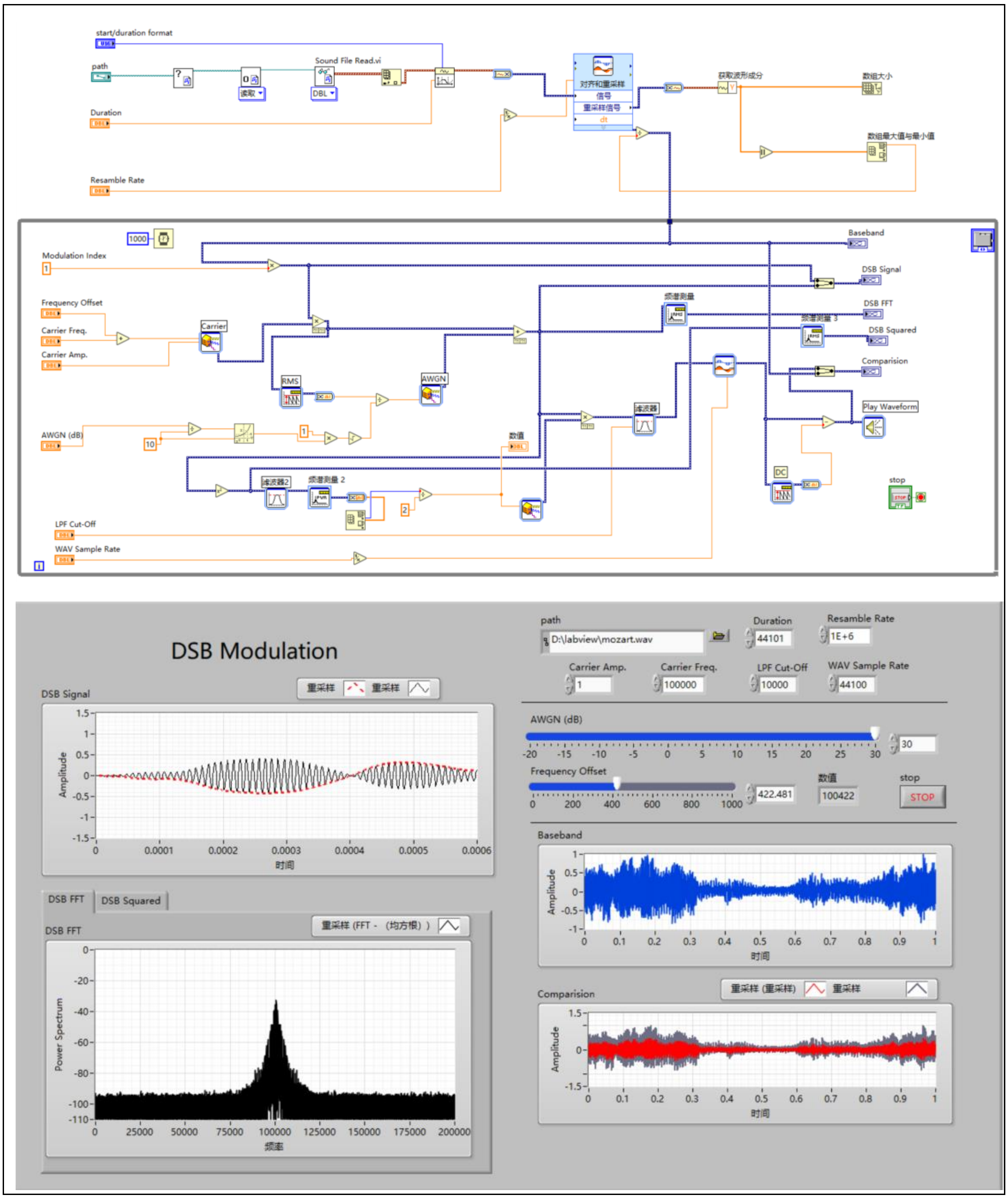


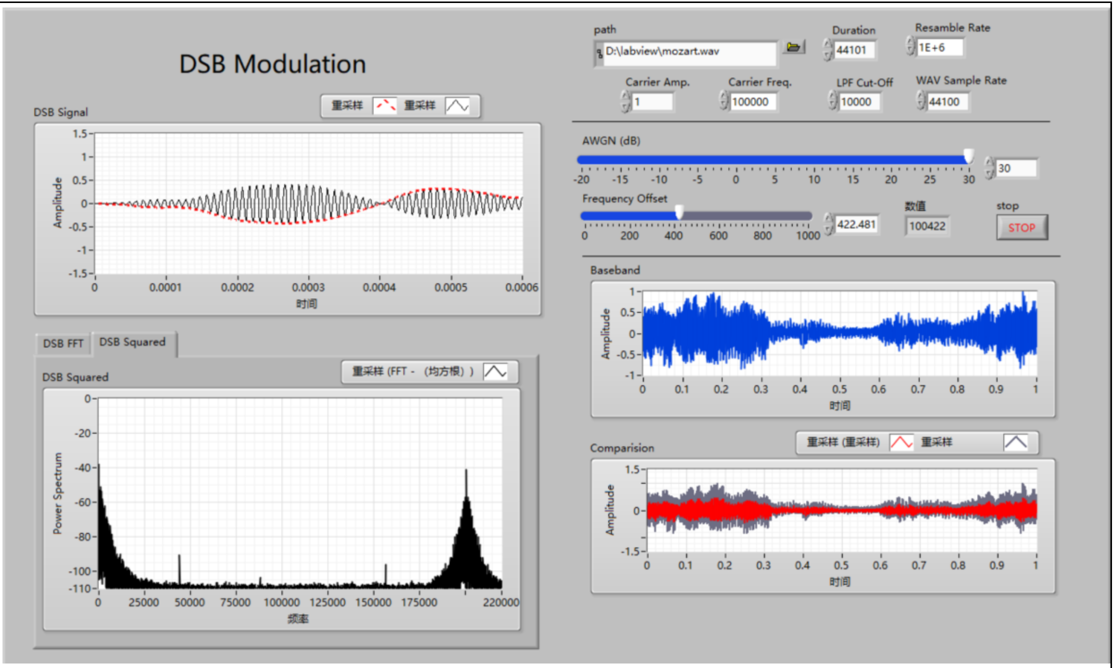


By comparing the estimation results of spectral component acquisition and phase-locked loop, we can find that the estimation results of phase-locked loop are better.

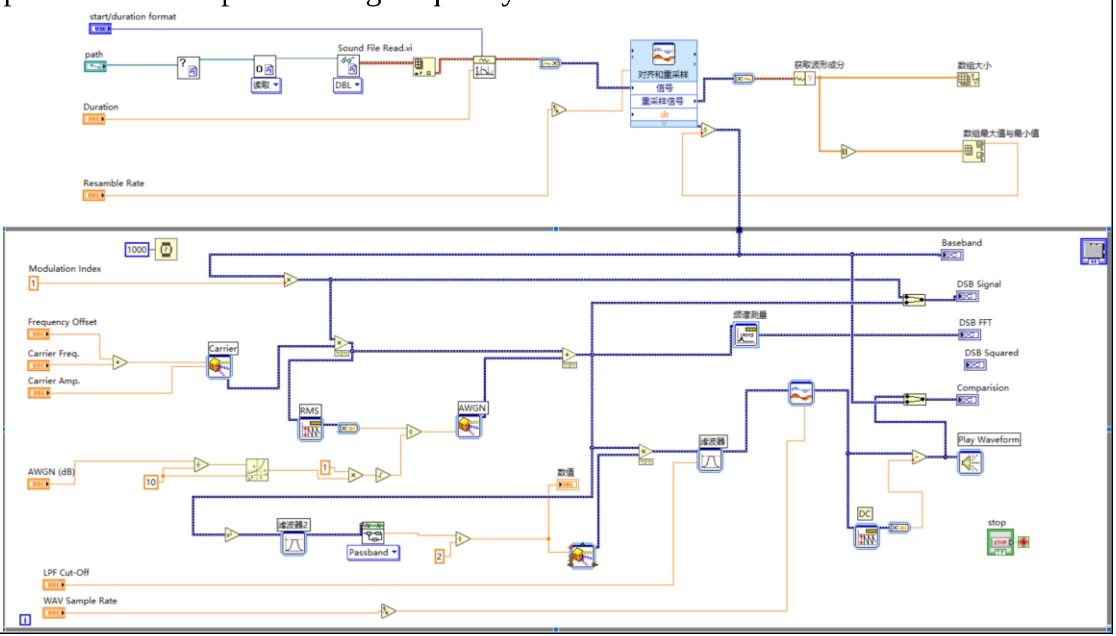
Then, after the single-tone test to check the program block diagram is no problem, we can conduct music signal test to verify the effect of our program in practical application. Frequency doubler and phase-locked loop are also used to test.

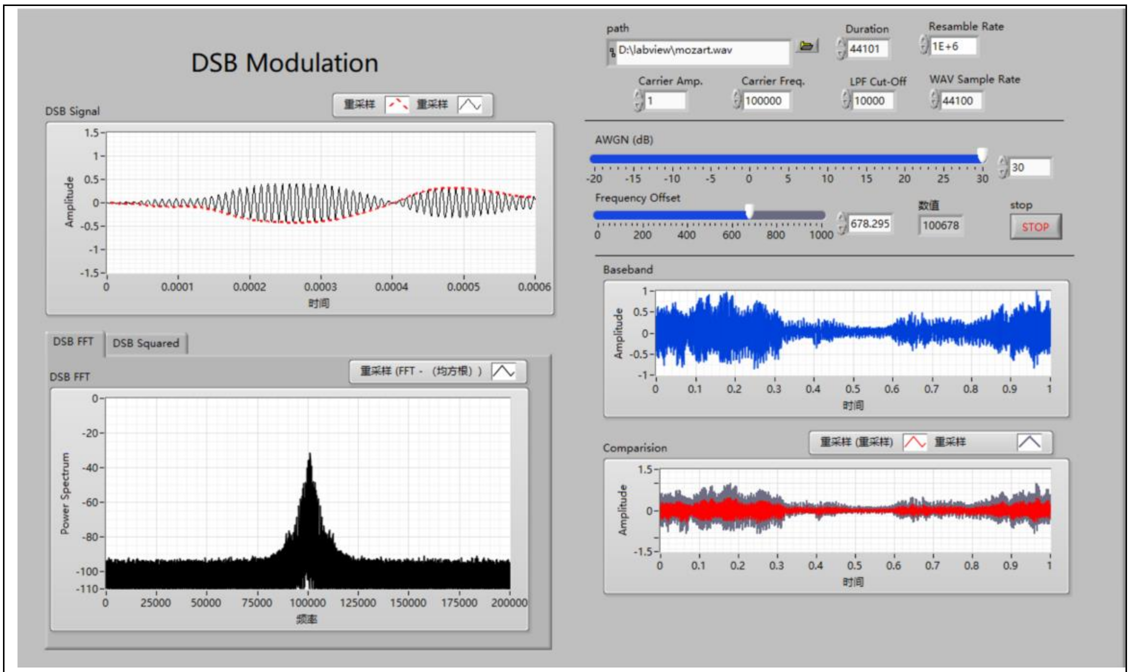
The program block diagram and simulation results of using frequency doubler for music signal test are as follows:


start/duration format
path
Sound File Read.vi
Duration
Resamble Rate
对齐和重采样
获取波形成分
数组大小
数组最大值与最小值
Modulation Index
Frequency Offset
Carrier Freq.
Carrier Amp.
Carrier
RMS
AWGN
AWGN (dB)
LPF Cut-Off
WAV Sample Rate
Baseband
DSB Signal
DSB FFT
DSB Squared
Comparision
Play Waveform
stop
DSB Modulation
path
D:\labview\mozart.wav
Duration
44101
Resamble Rate
1E+6
Carrier Amp.
1
Carrier Freq.
100000
LPF Cut-Off
10000
WAV Sample Rate
44100
AWGN (dB)
30
Frequency Offset
422.481
数值
100422
STOP
Amplitude
时间
DSB FFT
DSB Squared
Power Spectrum
频率
重采样 (FFT - (均方根))
重采样 (重采样)
重采样



The program block diagram and test results of the music signal test with phase-locked loop after using frequency doubler are as follows:



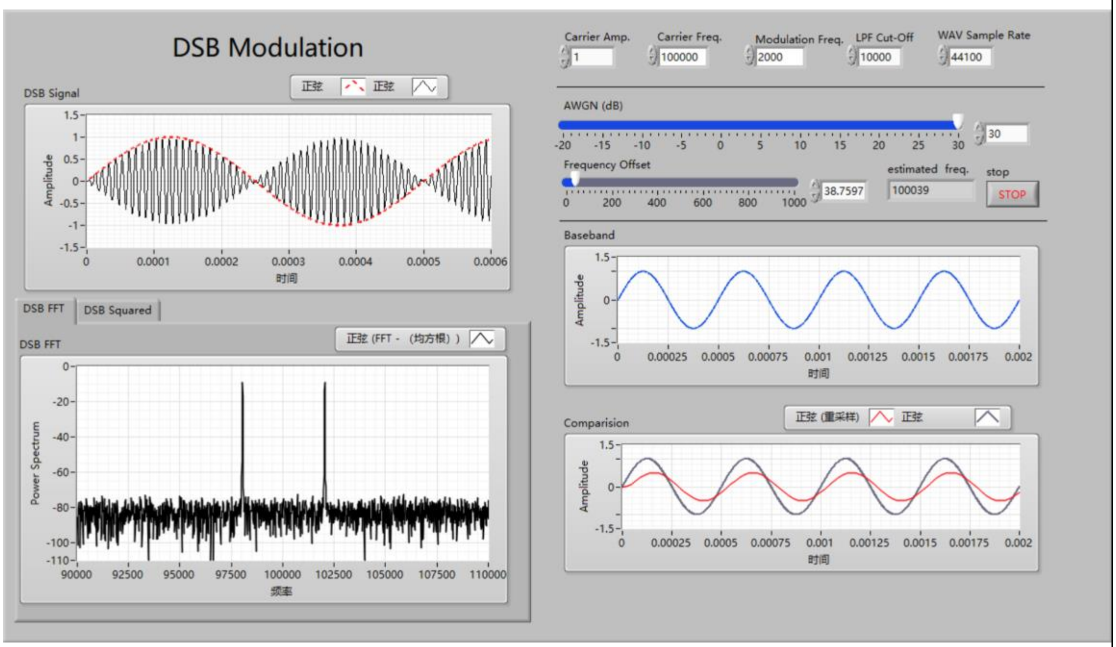


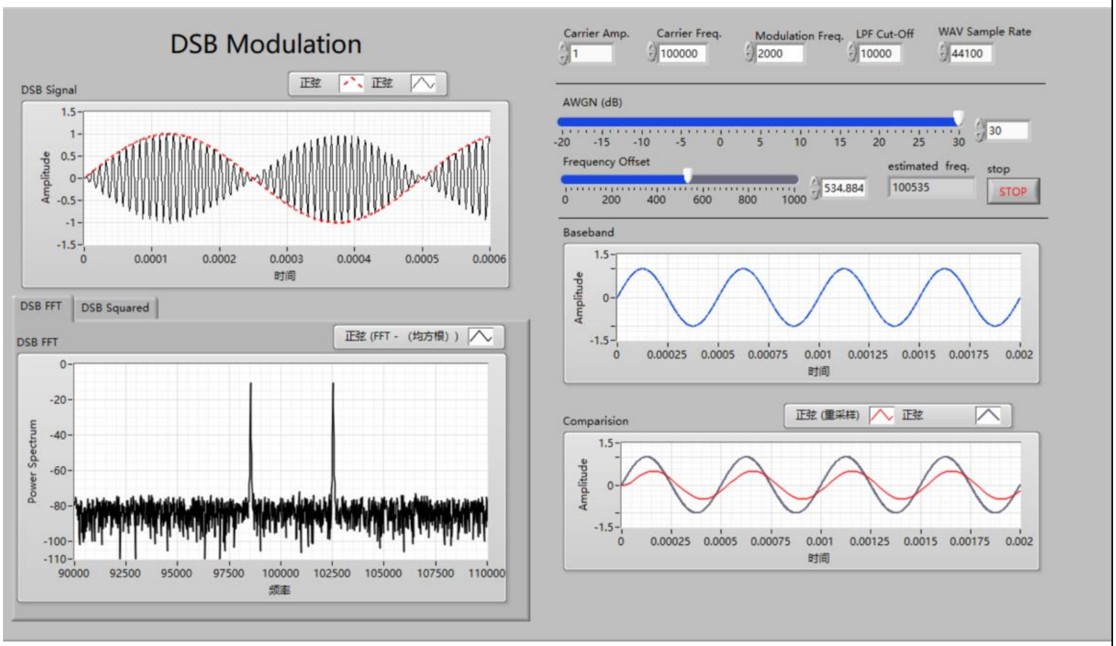
Similarly, by comparing the estimation effect of two methods of spectrum component acquisition and phase-locked loop, it can be found that the estimation effect of phase-locked loop is better. Therefore, we will later test the influence of factors such as frequency offset, cut-off frequency and order of low-pass filter on DSB modulation and demodulation system on the block diagram of the program using phase-locked loop.

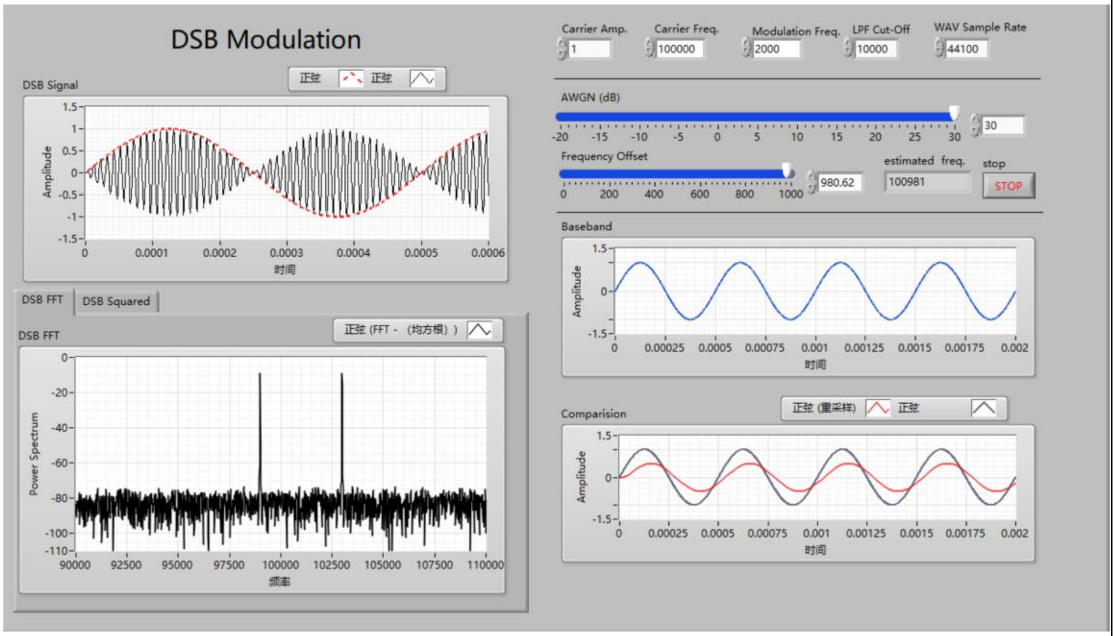
2. The factors influencing the system of DSB modulation and demodulation

2.1 The influence of frequency offset on DSB system

In the single-tone test, the frequency offset is constantly changed, and the waveform and sound effect generated by simulation are compared at the same time. In the single-tone test, the simulation results generated by randomly selecting different frequency offset are as follows:

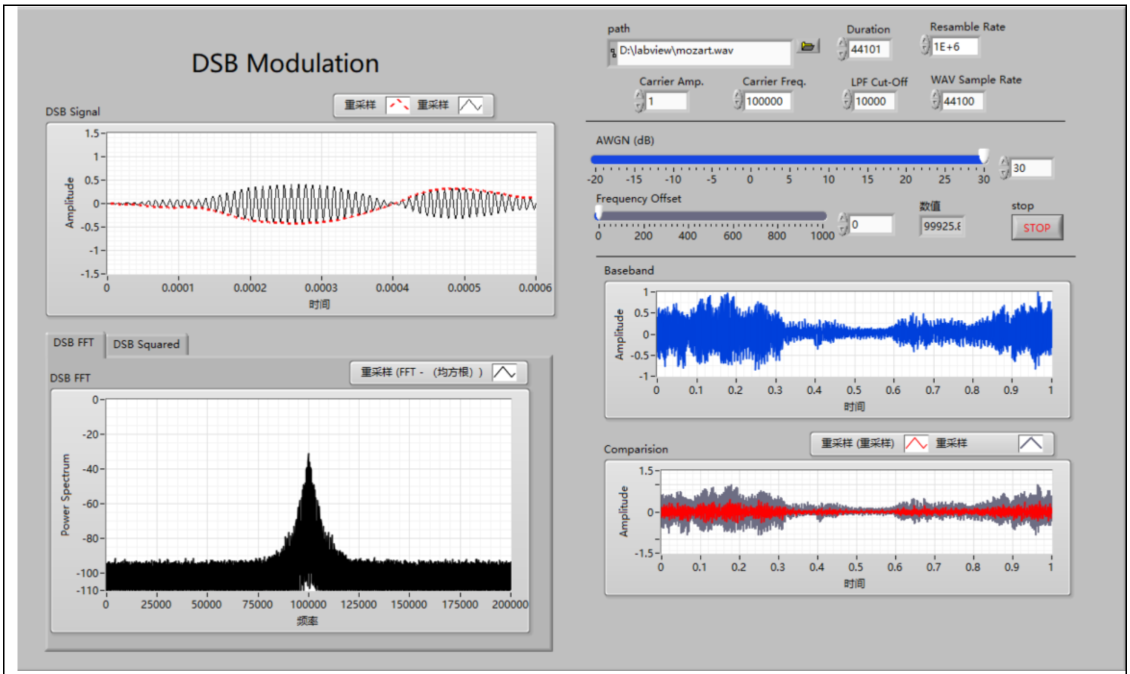


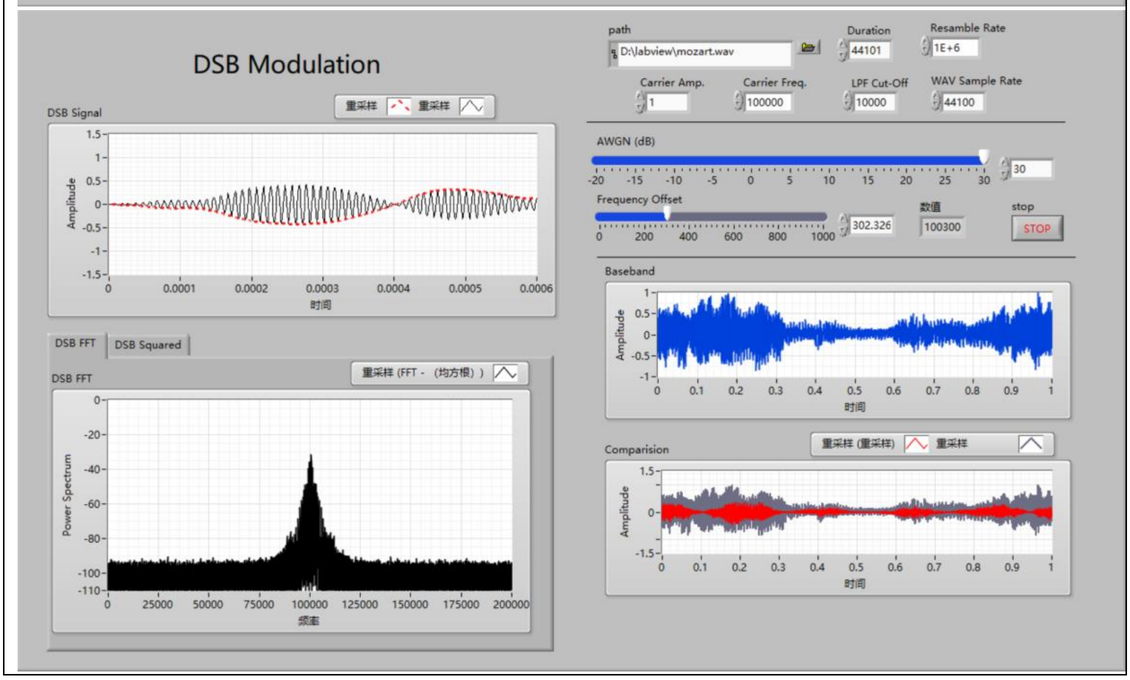


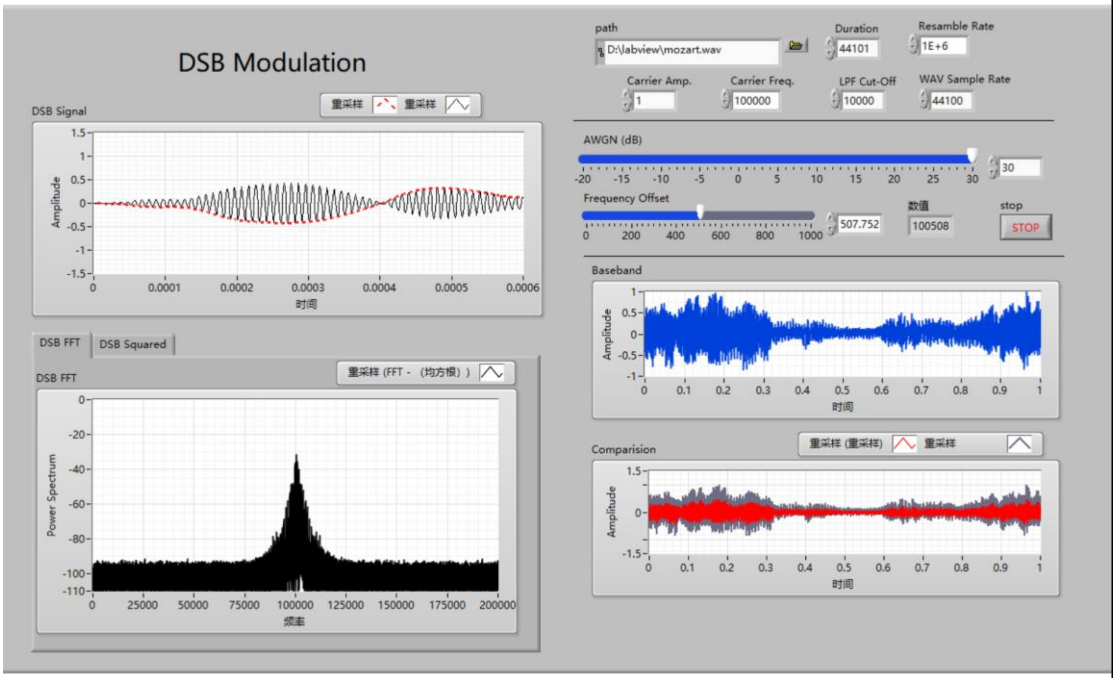


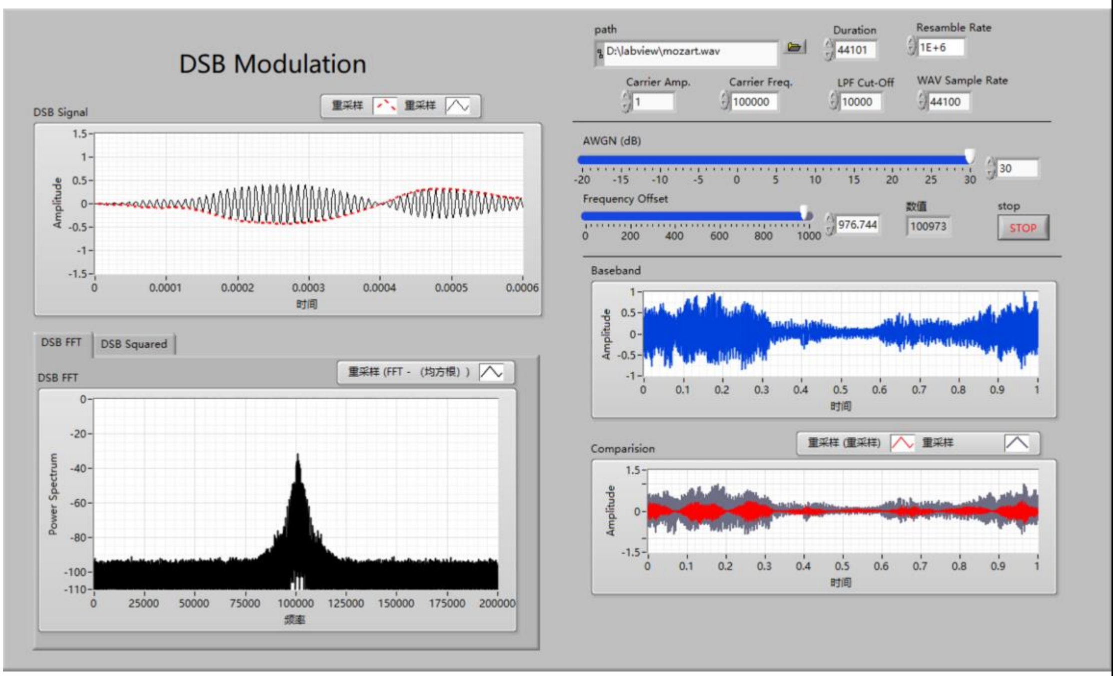
By observing the above set of figures, it can be found that the changes of the frequency offset do not affect the modulation and demodulation of DSB. If the original frequency plus frequency offset is between the passbands of the bandpass filter, the number of the carrier frequency can be perfectly estimated and better demodulation can be achieved.

In the music signal test process, the influence of frequency offset on the simulation results was tested, the number of the frequency offset was changed, and the waveform and sound effect of the recovered signal were compared. Different simulation results were obtained as follows:


DSB Modulation
DSB Signal
重采样
重采样
Amplitude
时间
DSB FFT
DSB Squared
DSB FFT
重采样 (FFT - (均方根))
Power Spectrum
频率
path
D:\labview\mozart.wav
Duration
44101
Resamble Rate
1E+6
Carrier Amp.
1
Carrier Freq.
100000
LPF Cut-Off
10000
WAV Sample Rate
44100
AWGN (dB)
30
Frequency Offset
0
数值
99925.8
stop
STOP
Baseband
Amplitude
时间
Comparision
重采样 (重采样)
重采样
Amplitude
时间


DSB Modulation
DSB Signal
重采样
重采样
Amplitude
时间
DSB FFT
DSB Squared
DSB FFT
重采样 (FFT - (均方根))
Power Spectrum
频率
path
D:\labview\mozart.wav
Duration
44101
Resamble Rate
1E+6
Carrier Amp.
1
Carrier Freq.
100000
LPF Cut-Off
10000
WAV Sample Rate
44100
AWGN (dB)
30
Frequency Offset
302.326
数值
100300
stop
STOP
Baseband
Amplitude
时间
Comparision
重采样 (重采样)
重采样
Amplitude
时间

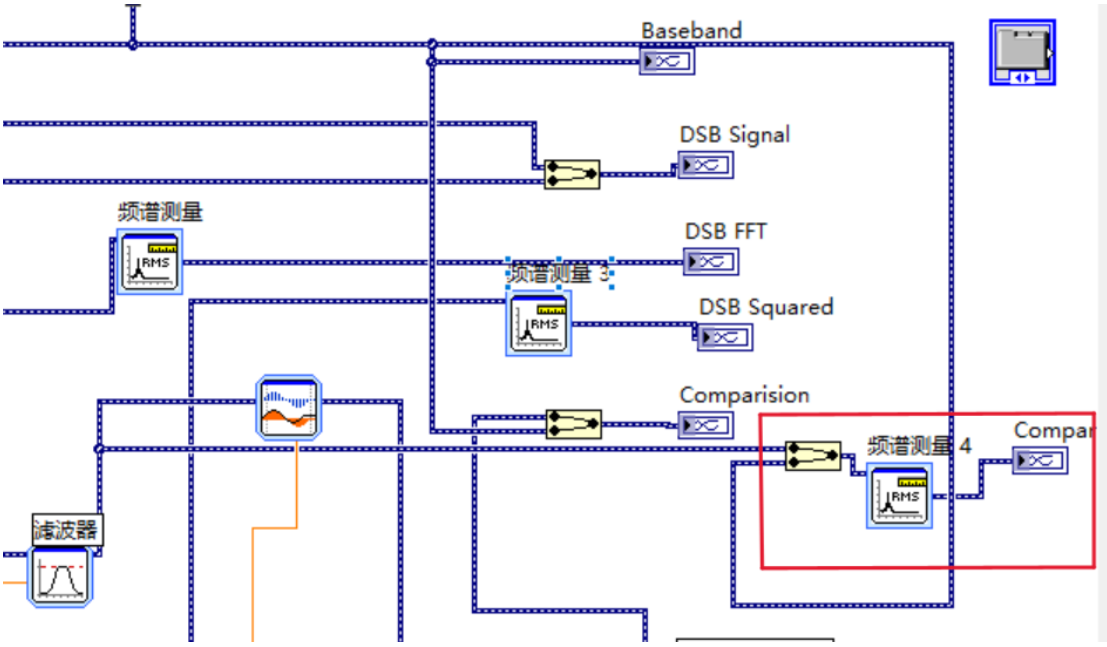




It can be found that the changes of the frequency offset have a great impact on the simulation results. The main reason is that when the frequency shift is too small (close to 0HZ) or too large (close to 1000HZ), the estimation of the carrier frequency will produce a large error, while when the frequency shift is in the middle region, the influence of the frequency shift is not significant. We infer that this is because the spectrum component of message signal in speech test is very rich and the principle of phase-locked loop itself leads to the result. By observing the DSB FFT diagram, it can be found that the bandwidth of message signal is very wide, up to 50000HZ, so there are many spectrum components contained after passing the bandpass filter, which may have a great impact on the estimation of the phase-locked loop. Therefore, different frequency offset will lead to different results of the phase-locked loop. In the above analysis of the single-tone test, we conclude that different frequency offset will not have a great impact on the simulation results. This result also confirms our conjecture. At the same time, the method of obtaining the maximum array index after frequency doubler is used to test the speech again, and it can be found that frequency offsets will not have a great influence on the result at this time, because this method always obtains the index at the largest number of the array, so the abundance of spectrum components will not have a great influence, but the phase-locked loop does not. The above verifies our conjecture that the rich spectral components of speech signals and the inherent properties of the phase-locked loop cause the result that frequency shift have a large influence on the results.

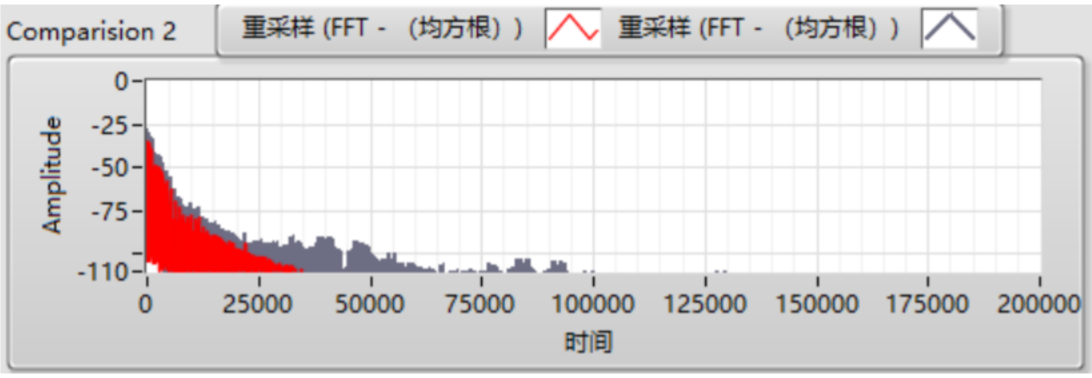
2.2 The influence of LPF cut-off frequency on DSB system

In order to better observe the effect of the filter and compare the influence of different cutoff frequencies and orders on the effect of the filter, a new display module is added in this experiment. The merge signal module is used to display the spectrum information of signal and baseband signal after passing the filter in a graph so that it can be better observed that the effect of changing the cutoff frequency and order of the filter. The diagram is as follows:

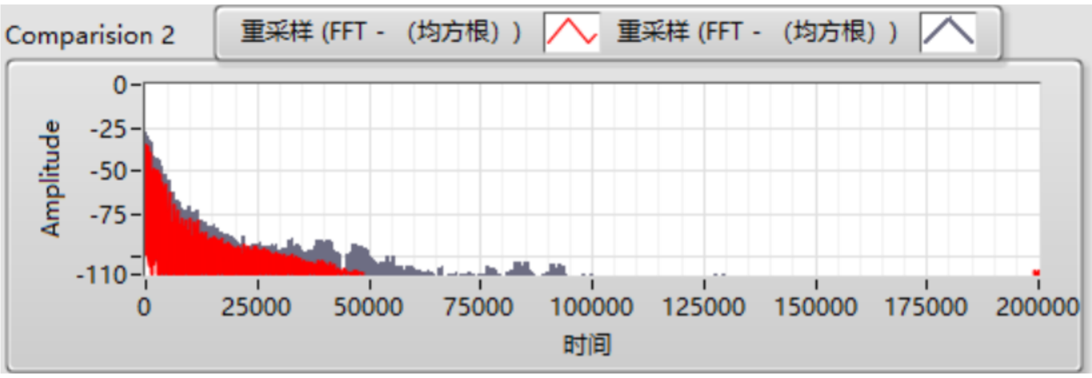


Firstly, we set the cut-off frequency of LPF to 10000HZ, and the result is:

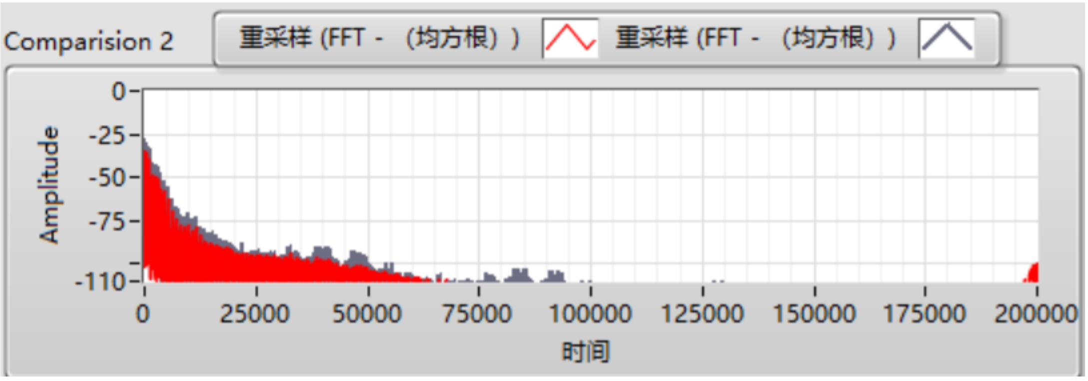
When the cut-off frequency is 20000HZ:



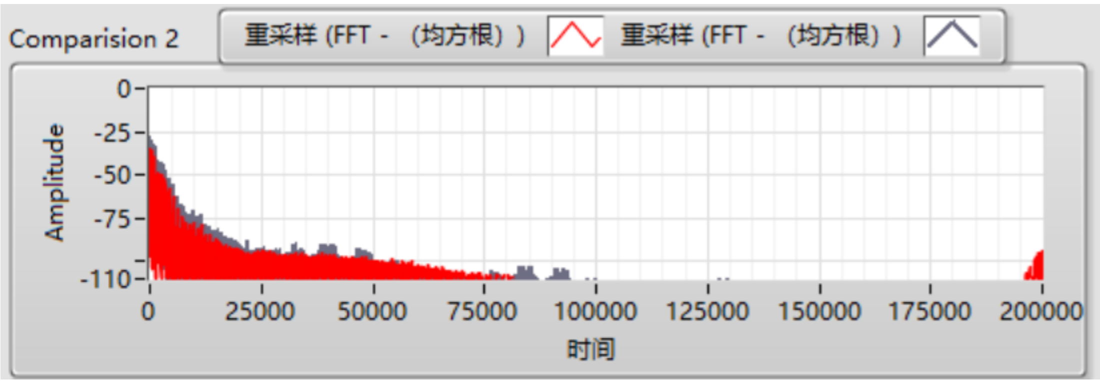
When the cut-off frequency is 30000HZ:



When the frequency is 40000HZ:



When the cut-off frequency is 50000HZ:

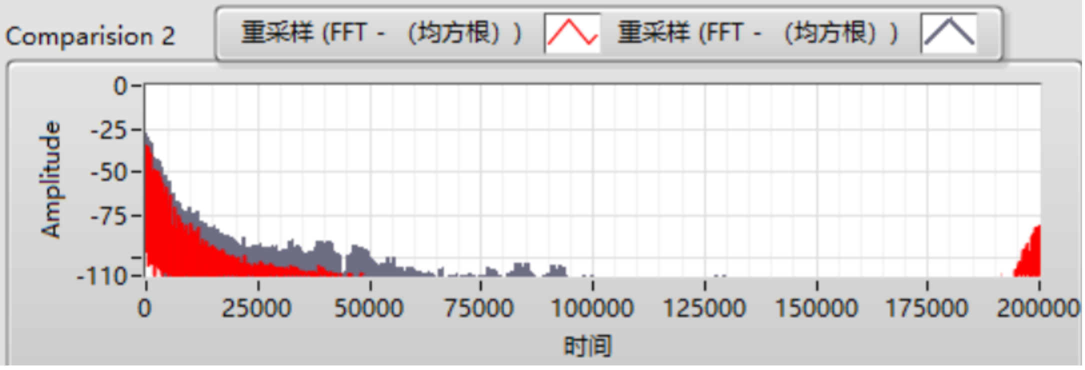


As can be seen from the above set of figures, with the increase of cutoff frequency, the spectrum component of the obtained signal becomes richer and closer to the original spectrum of the baseband signal. However, when the cutoff frequency is greater than 50000HZ, it can be seen that more noise signals will be added in. It can also be found that noise will start to have a greater impact by listening to the sound.

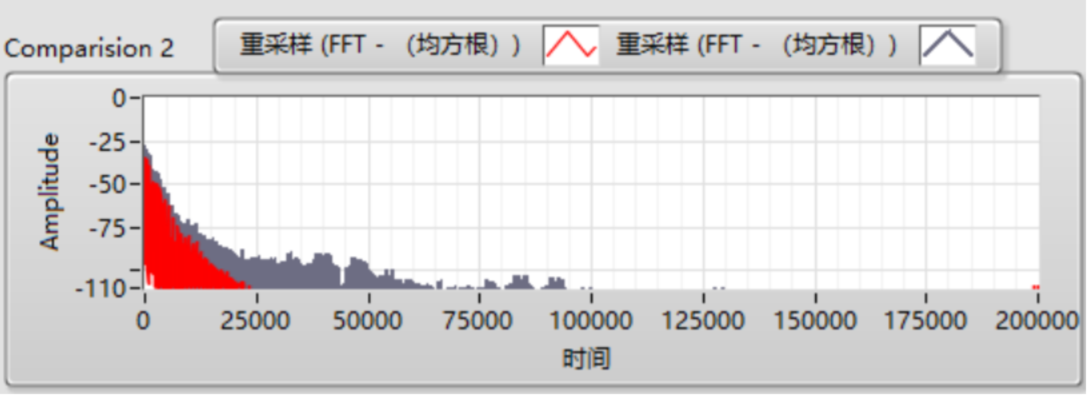
2.3 The influence of LPF order on DSB system

Similarly, we also take the above methods to compare.

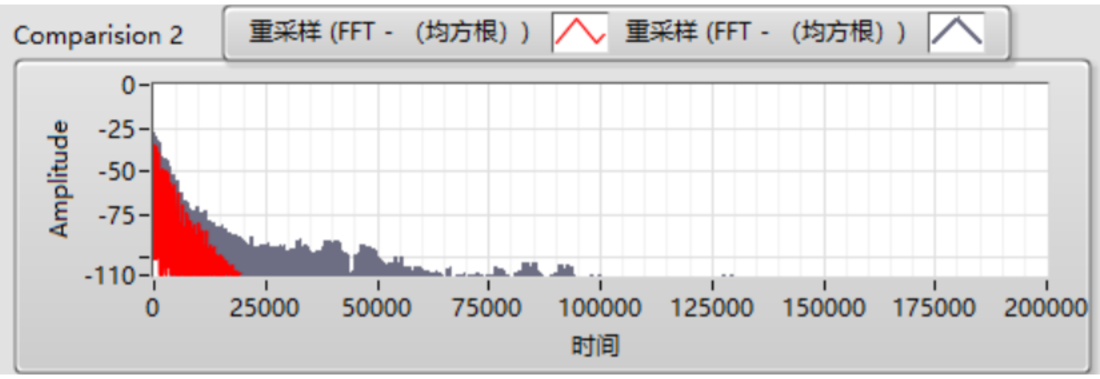
When the order is 1:



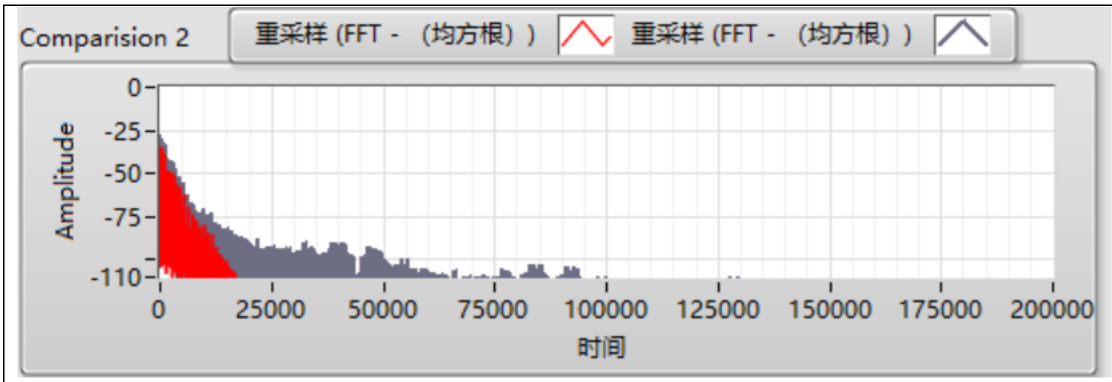
When the order is 2:



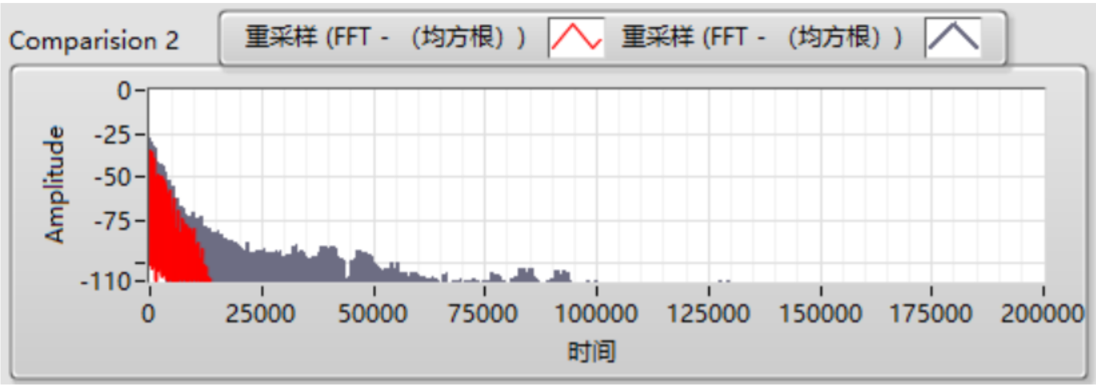
When the order is 3:



When the order is 4:



When the order is 9:



As can be seen from the figure above, as the order increases, the obtained spectrum becomes steeper near the cutoff frequency. Therefore, the higher the order, the less signal noise is obtained, but the effective information is also reduced. Considering the increase of the cost brought by the increase of the order, it is a better choice to use about the fourth order.

3.Advantages and disadvantages of DSB modulation system

DSB modulation is the abbreviation of suppressing carrier bilateral band modulation, its advantages are high power utilization, but the bandwidth is the same as AM, the frequency band utilization is not high, the reception requires synchronous demodulation, and the equipment is more complex1. The application value of DSB modulation system lies in its high-power utilization, which can improve the efficiency of signal transmission to a certain extent, and reduce the noise interference in the signal transmission process. However, the disadvantages of the DSB modulation system are that the frequency band utilization is not high, the reception requires synchronous demodulation, and the equipment is more complex.

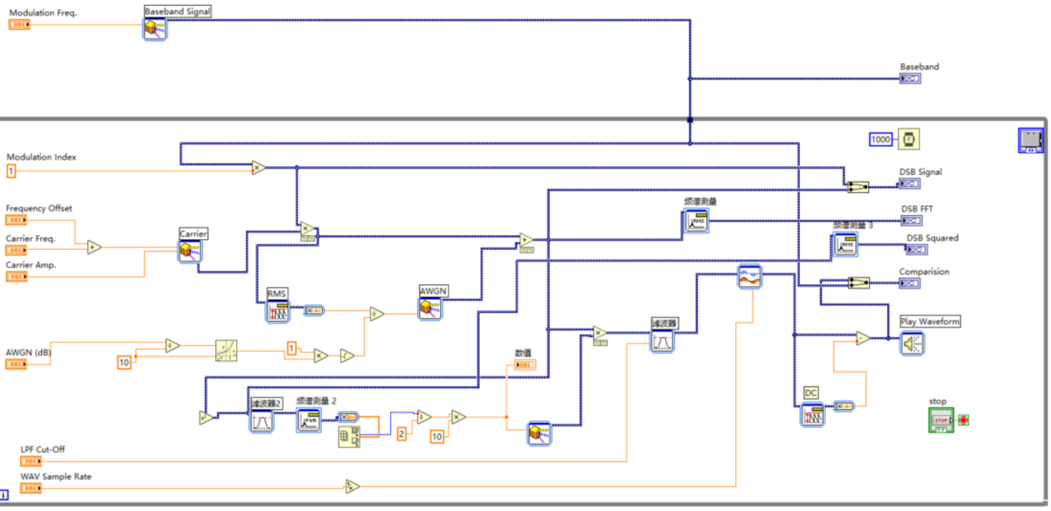
In short, DSB modulation systems have important application value in some specific occasions, such as radio and television, radio communication and other fields.

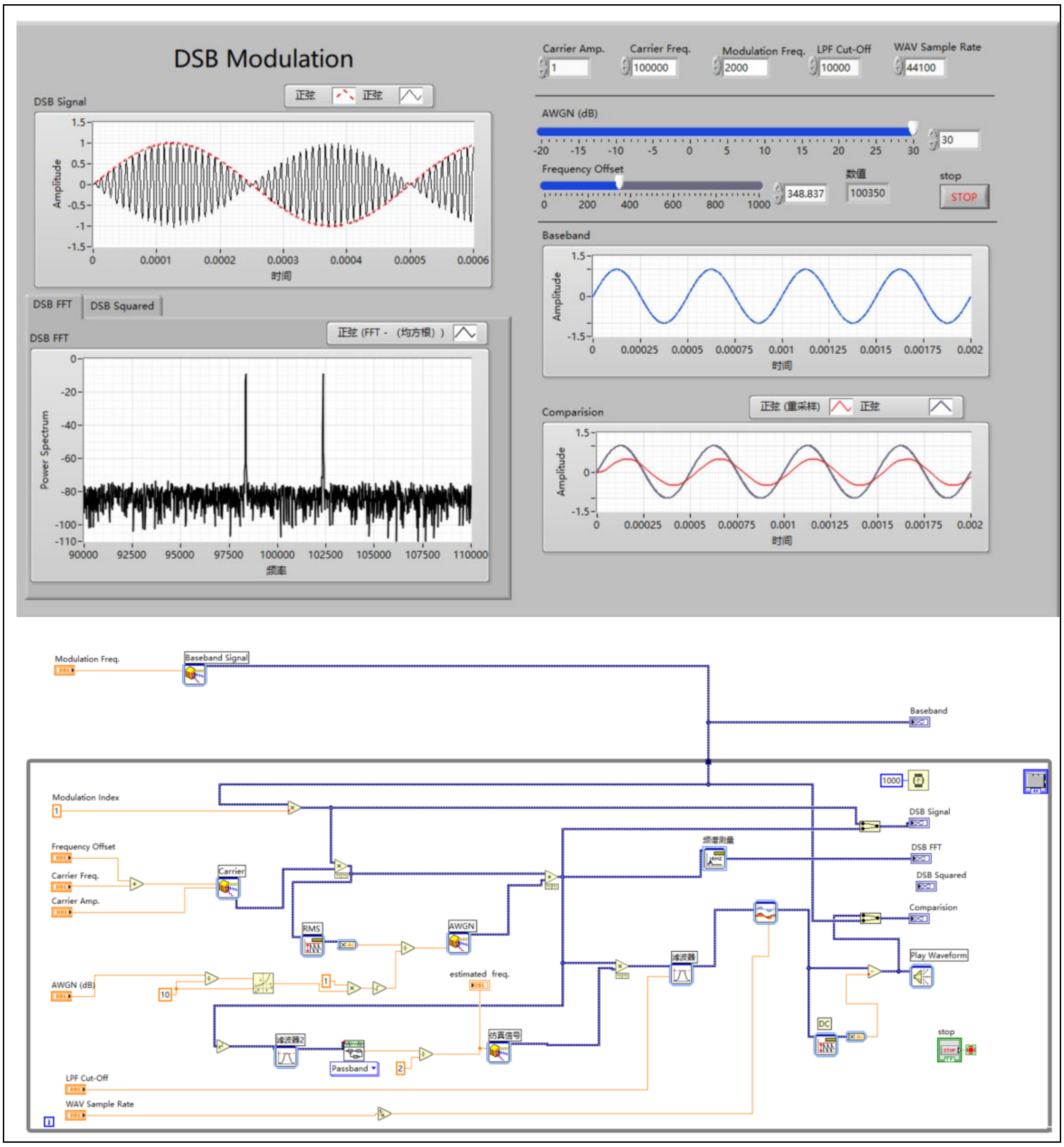
In short-wave communication, DSB modulation technology can be used for voice and data transmission, and its high-power utilization rate can improve the efficiency of signal transmission to a certain extent, and can also reduce noise interference in the signal transmission process2. In addition, DSB modulation technology can also be used in radio communication fields, such as radio broadcasting, radio telephony etc.

Experience

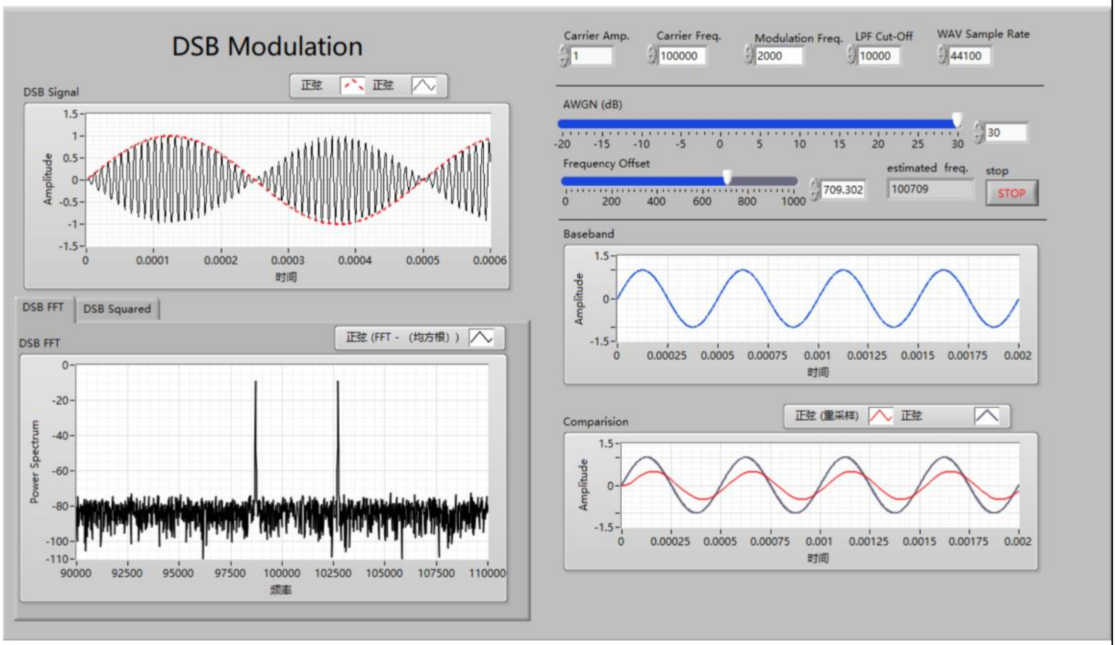
1. Class screenshot

Single-tone signal test

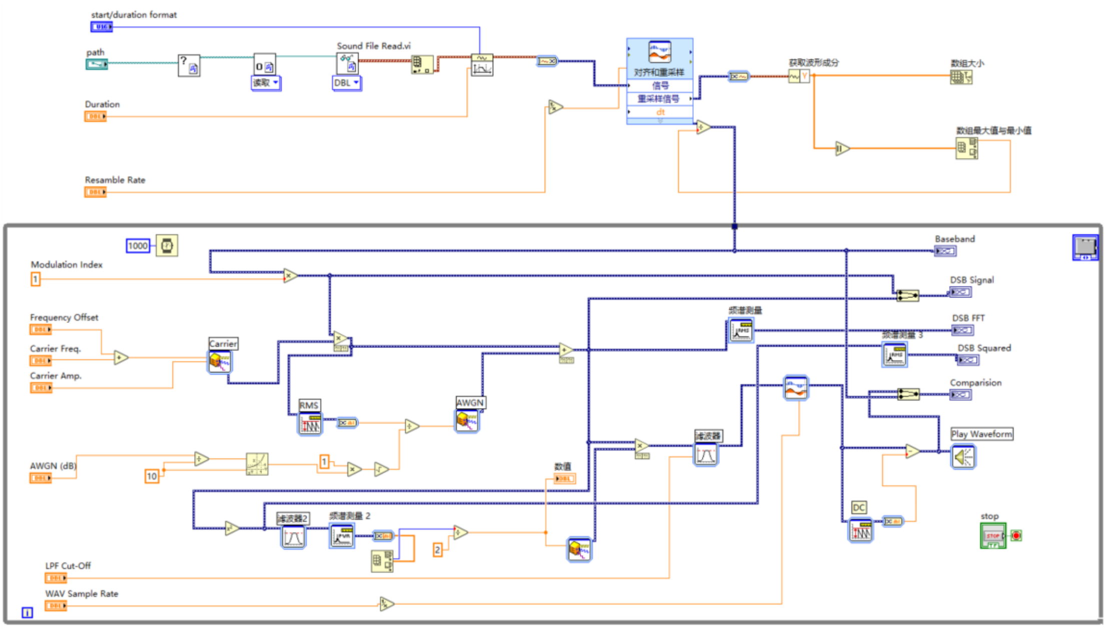


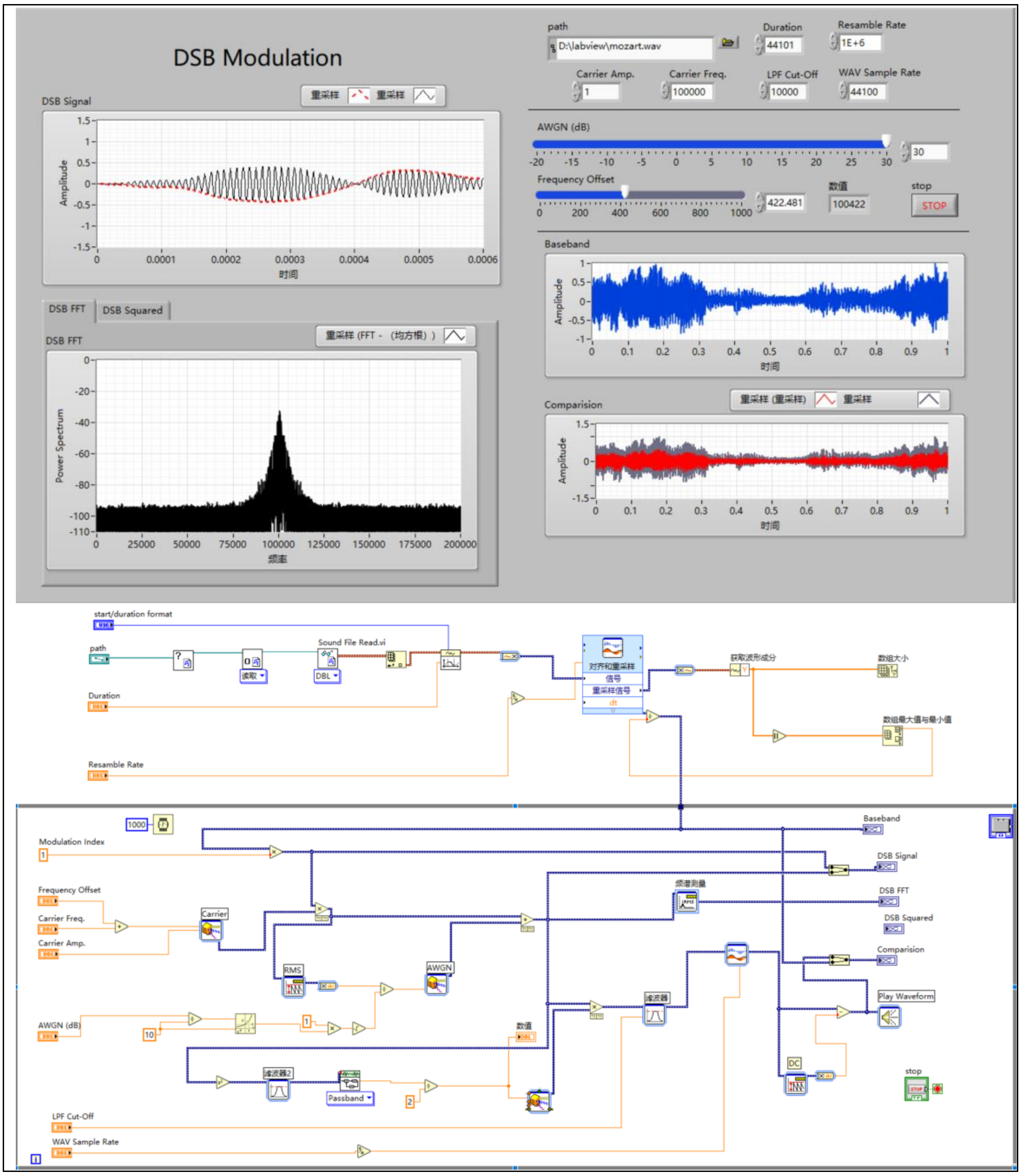


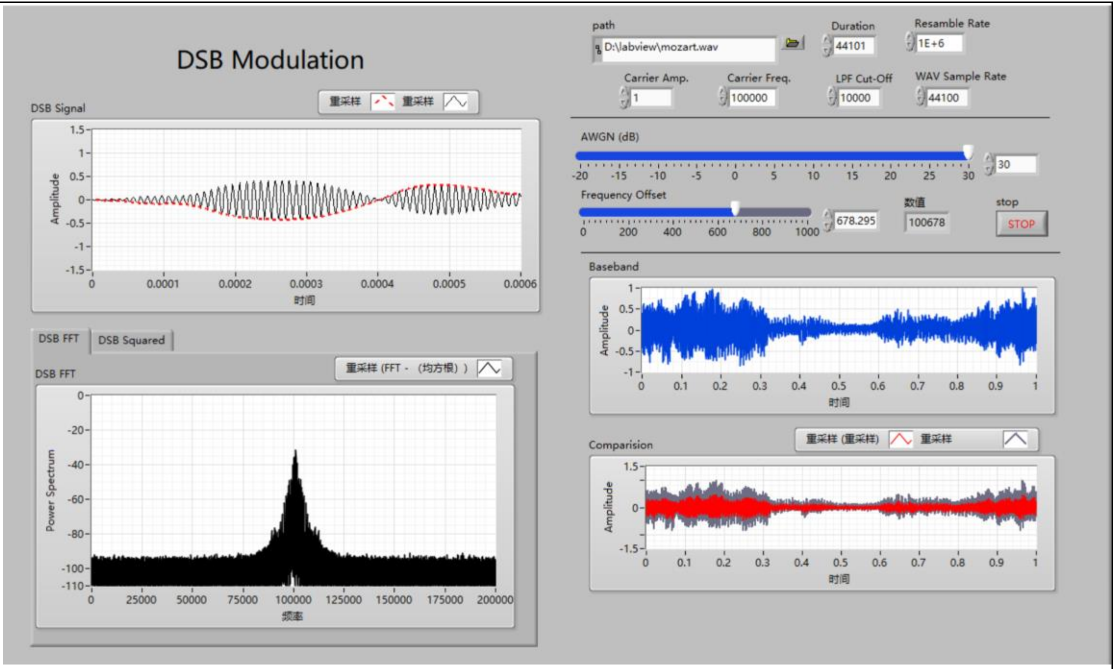
DSB Modulation
DSB Signal
正弦
正弦
Amplitude
时间
Carrier Amp.
1
Carrier Freq.
100000
Modulation Freq.
2000
LPF Cut-Off
10000
WAV Sample Rate
44100
AWGN (dB)
30
Frequency Offset
348.837
数值
100350
stop
STOP
DSB FFT
DSB Squared
正弦 (FFT - (均方根))
Power Spectrum
频率
Baseband
Comparision
正弦 (重采样)
Modulation Freq.
Baseband Signal
Modulation Index
Frequency Offset
Carrier Freq.
Carrier Amp.
Carrier
RMS
AWGN
AWGN (dB)
estimated freq.
滤波器2
Passband
仿真信号
LPF Cut-Off
WAV Sample Rate
频率测量
滤波器
DC
DSB Signal
DSB FFT
DSB Squared
Comparision
Play Waveform
stop
1000



Music signal test




DSB Modulation
DSB Signal
Amplitude
时间
DSB FFT
DSB Squared
Power Spectrum
频率
path
D:\labview\mozart.wav
Duration
44101
Resamble Rate
1E+6
Carrier Amp.
1
Carrier Freq.
100000
LPF Cut-Off
10000
WAV Sample Rate
44100
AWGN (dB)
30
Frequency Offset
422.481
数值
100422
stop
STOP
Baseband
Comparision
start/duration format
Sound File Read.vi
对齐和重采样
获取波形成分
数组大小
数组最大值与最小值
Modulation Index
Carrier
RMS
AWGN
频谱测量
滤波器
滤波器2
Passband
Play Waveform
DC

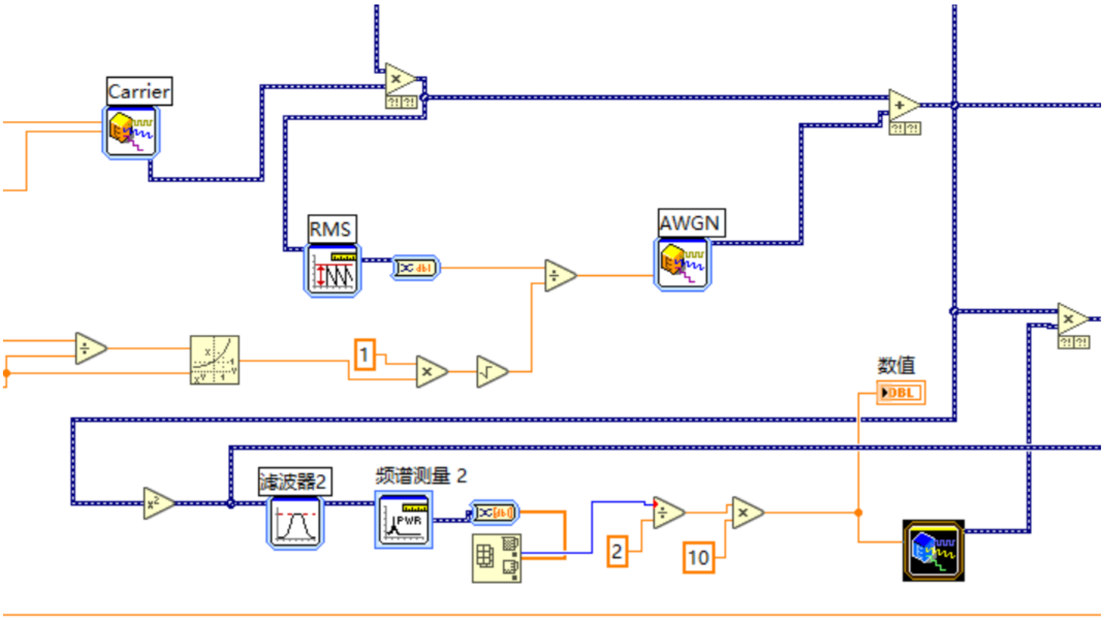


2. The problem we met in this experiment

1. Instead of setting the reset phase, the continuous output waveform is selected, which will cause the final output waveform to change continuously. Because the total waveform sampling stop point may not be the end of a full cycle, continuous output causes the next waveform generation to continue with the last generated phase, while resetting the phase generates a new waveform from the start of the period with the phase starting from zero.

2. When using the square law to use the index value at the maximum extracted power, you need to multiply the final output by 10. This is related to the sampling rate of 1M we set and the sampling number of 100k and the difference of ten times the carrier frequency, which we only have a certain understanding of this problem in the end, and do not clearly understand the essential reason

3. Then, in the test of music signal, after modifying the framework of monophonic test, errors were found in the running, as shown in the figure below. After analysis, we found that the sampling number was changed to be the same as the sampling rate in order to display the signal for 1s. Currently, due to the modification of sampling number, we do not need to multiply the carrier frequency by 10 in the square law estimation process, so the problem is solved by removing the multiplied 10 module.



4. As we mentioned above, when the frequency offset is small or large, the use of phase locked loop will produce certain errors. We speculate that this may be caused by the principle of phase locked loop. Due to the time problem, we do not fully understand why his principle leads to such a result.

3.Contribution

We two finish the whole LabView program together. Zhang Haodong complete the analysis of the procedure and the results and fixed the bugs in the procedure and finish the analysis of factors affecting DSB modulation and demodulation system. The introduction and analysis of the advantages and disadvantages of DSB modulation and demodulation system were completed, and its social application value was elaborated by Song Yihang.

Contribution ratio :50%,50%

Score	97