Lab 1：AM

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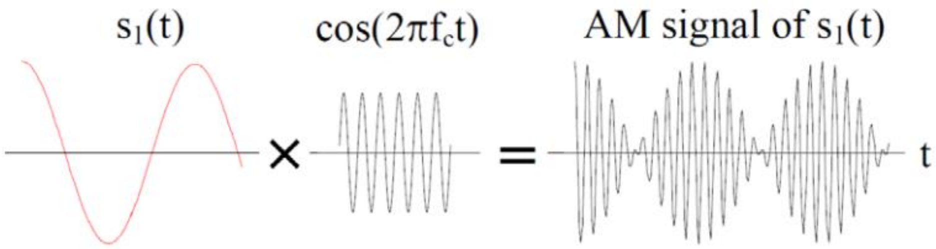
Introduction

1.The principle of AM modulation and demodulation

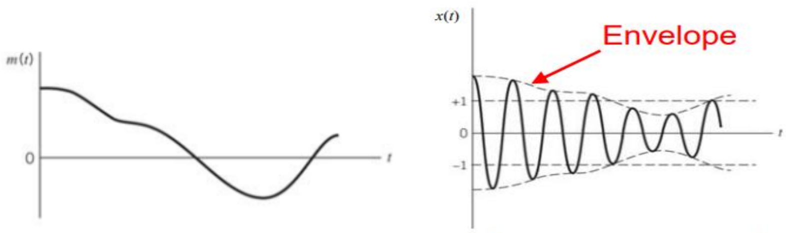
The AM modulation and demodulation are useful technology that are used widely in communication area. AM means amplitude modulation. In AM, the amplitude of a sinusoidal carrier wave is varied in accordance with the message signal. Its purpose is to let a carrier signal with higher frequency to carry the message signal that we want to transmit. The most general form is as follows:

$$s(t) = A\big(1 + k_a m(t)\big)\cos(2\pi f_c t)$$

Where k_a is modulation index, f_c is the frequency of carrier. 1 is the DC offset. Therefore s(t) is modulated signal and it will be transmitted but in this process there are many noise in the signal channel.



Receiver must do the demodulation to demodulate the modulated signal to recover the message signal. The method is to obtain the envelope of s(t) because the envelope signal is just the message signal if the condition of $|k_a m(t)| < 1$ and $f_c >> W$, where W is the message bandwidth will be met. This demodulator includes rectifier and LPF (low pass filter). The rectifier will be realized by the operation of taking absolute value in LabVIEW that is used in this experiment.



Message signal　　　AM wave with $|k_a m(t)|<1$

2.The modules of LabVIEW Express

LabVIEW Express module is a simplified programming tool, it can be set by the user with some parameters, automatically configured by LabVIEW to generate a template of the program. There are many modules of LabVIEW Express used in this experiment including Spectral measurement, Simulated signal generation, Dynamic data conversion, LPF, Signal measurement and Resample.

Spectral Measurement: this module is used to obtain the frequency spectrum of the signal with the method FFT so that the graph of the spectrum can be expressed. In this experiment, there are two modules to obtain the frequency spectrum of AM signal and the signal after rectifier in order to analyze the consequence in frequency spectrum by AM and rectifier.

Simulated Signal Generation: this module is used to generate many signals with different types such as sinusoidal waves, rectangle waves, triangular waves, and sawtooth waves. In this experiment we need a cosine signal as carrier. So, the phase is set as 90, the sample rate is 1000000HZ.

Dynamic Data Conversion: this module can convert the dynamic data type to other data type such as Value, Boolean, arrays etc. which can be used by other VI or function. For example, in this experiment, it changes the information of waves to double type to participate in subsequent calculation afterward. It also can convert the arrays data to waves data in getting the speech signal information from the file to participate resampling and other process.

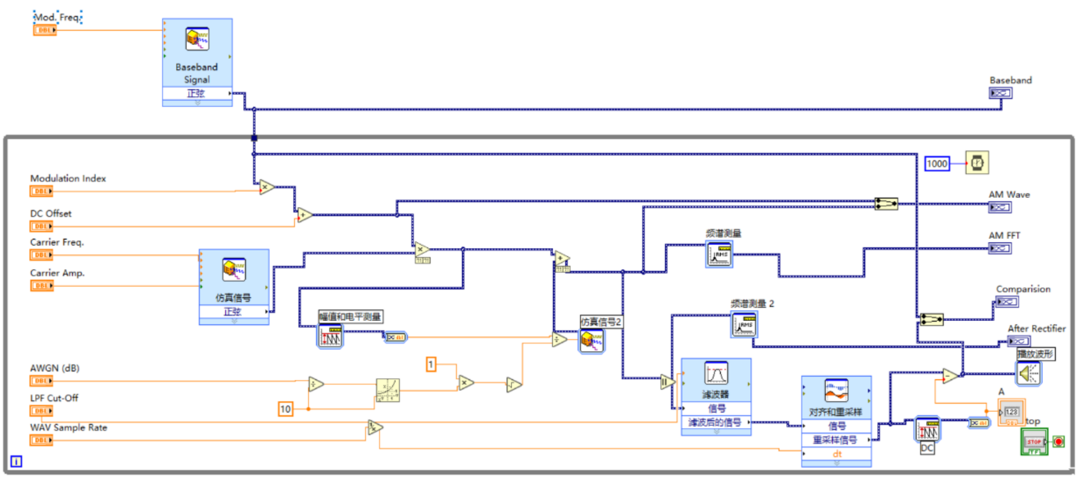
Lowpass Filter: this module is a filter to obtain the part of signal whose frequency is less than the cut-off frequency. In this experiment, the LPF is to obtain the signal with low frequency after the rectifier. Together with the rectifier, the LPF realize the envelope detection to get the recovered signal. In this experiment, the cut-off frequency is set as 10000HZ.

Align and Resample: there are two modules, one is after the filter and the other is used to obtain the baseband signal in the Mozart music. In fact, the signal after the filter cannot be played in the sound player because the sound player in this experiment can only play the signal whose sampling rate is between 100HZ and 200000HZ. But the sampling rate of the signal after the LPF is 1000000HZ that is beyond this range. So, this module is demanded to resample the signal to change the sampling rate. This resampling rate is set as 44100HZ. The second module is used to resample the speech signal after getting from the file to match the sample rate of others because only one sample rate is allowed in a system.

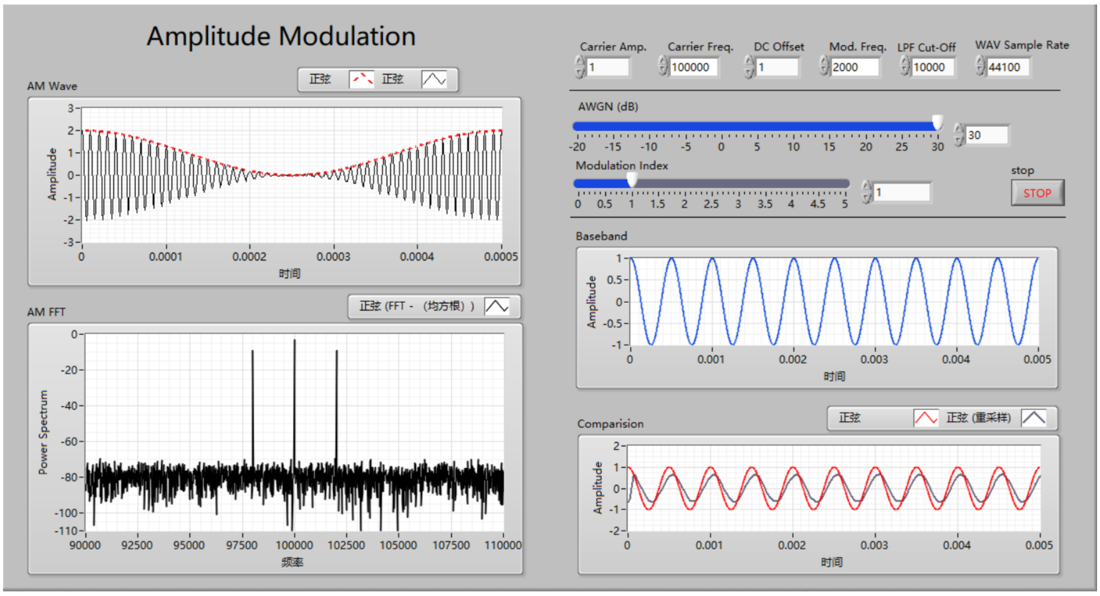
Signal Measurement: this module is to obtain the DC component with the method of taking the average value in order to minus the DC offset in the recovered signal after LPF and Resample. Another module is to calculate the root of the mean square (RMS) to generate the Gauss noise according this value.

Lab results & Analysis：

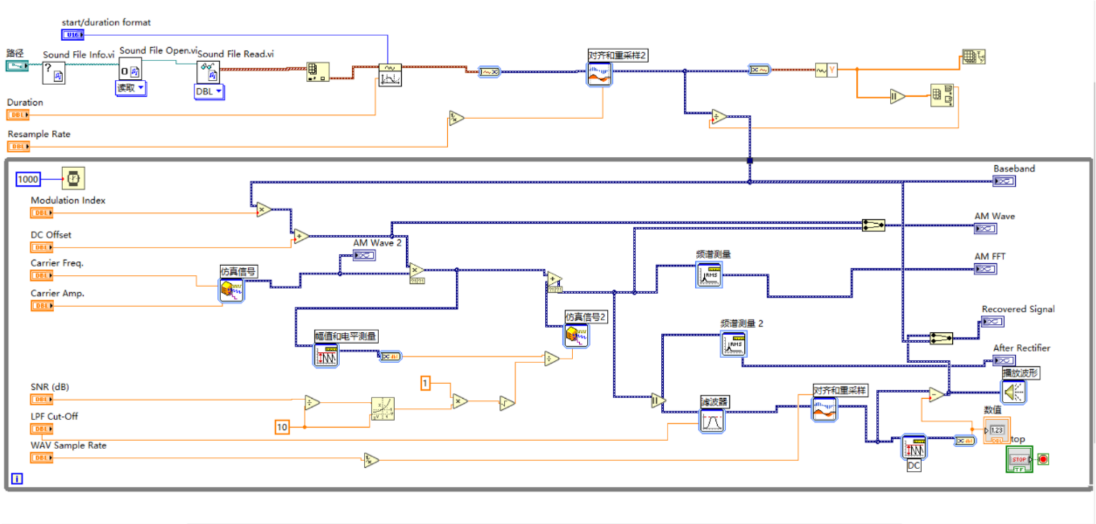
1.Single-tone test AM modulation program block diagram：



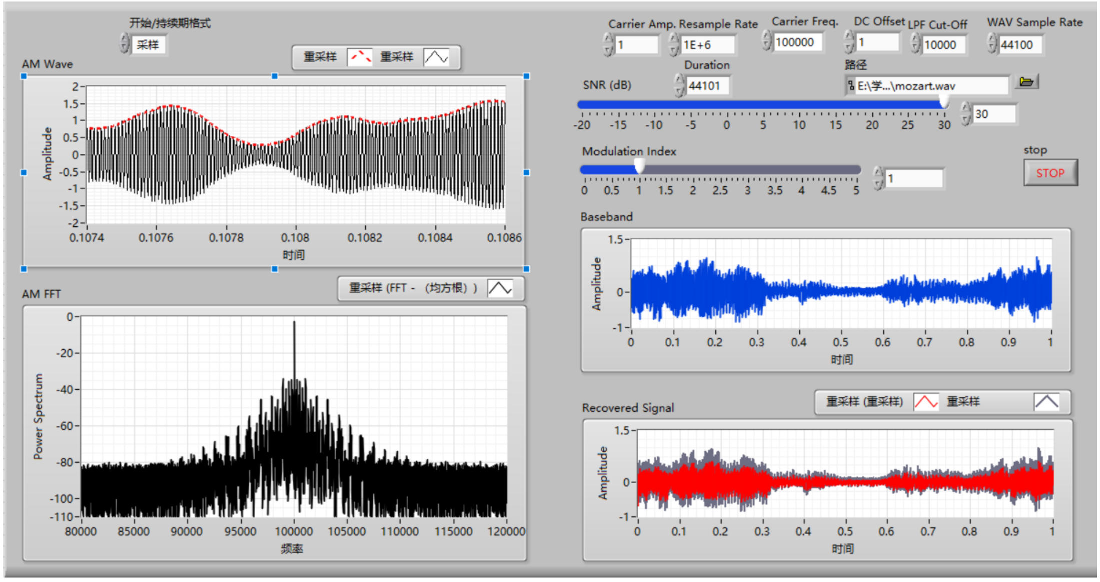
Simulation results：



Audio test program block diagram：

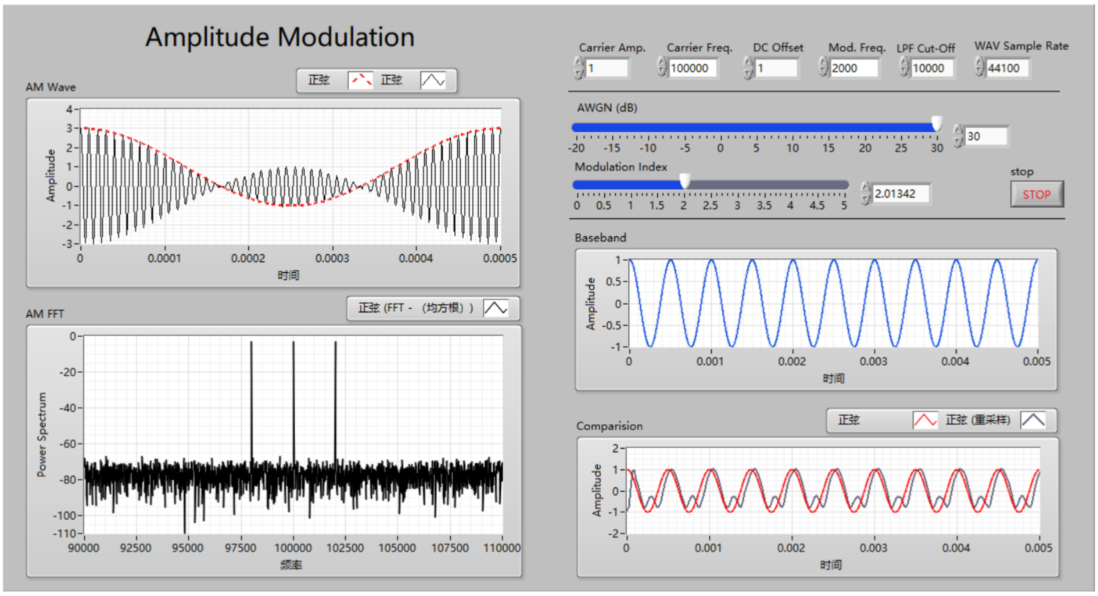


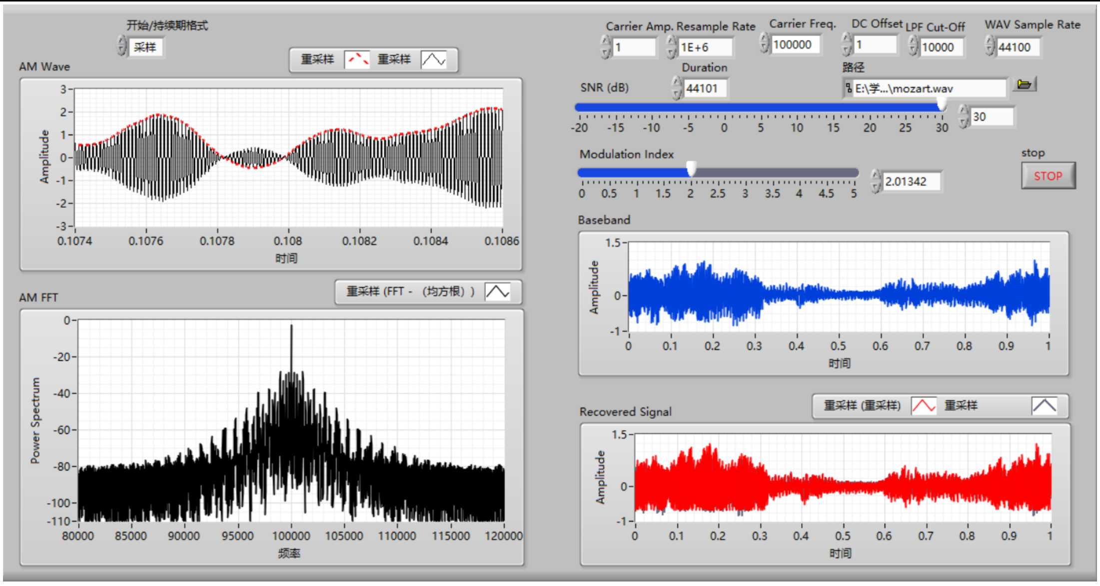
Simulation results：



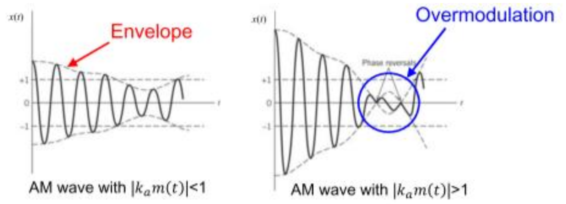
Factors affecting AM modulation:

1. overmodulation:

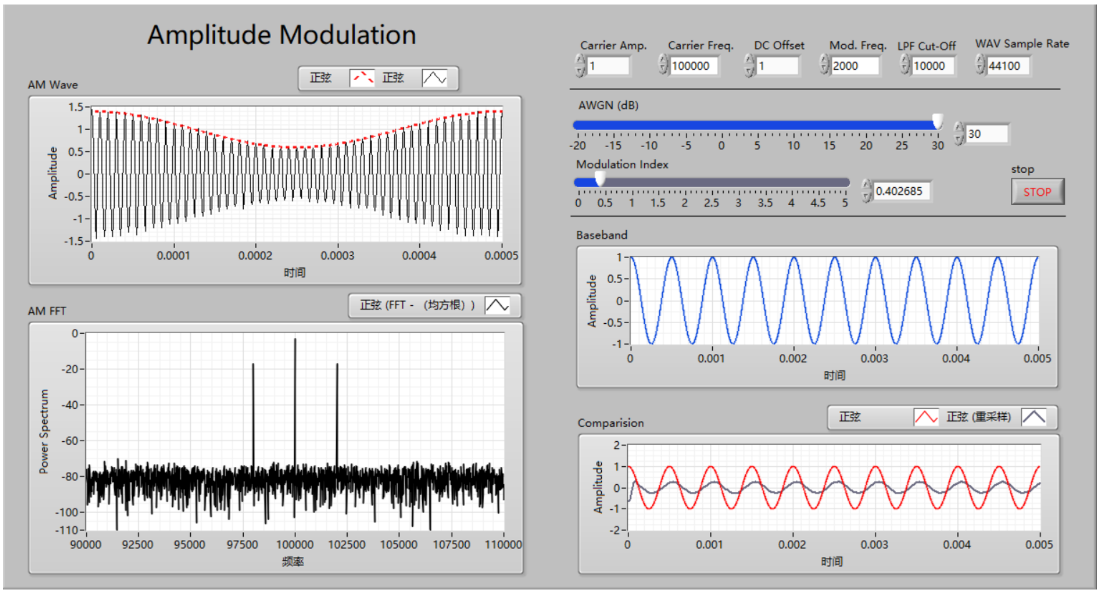


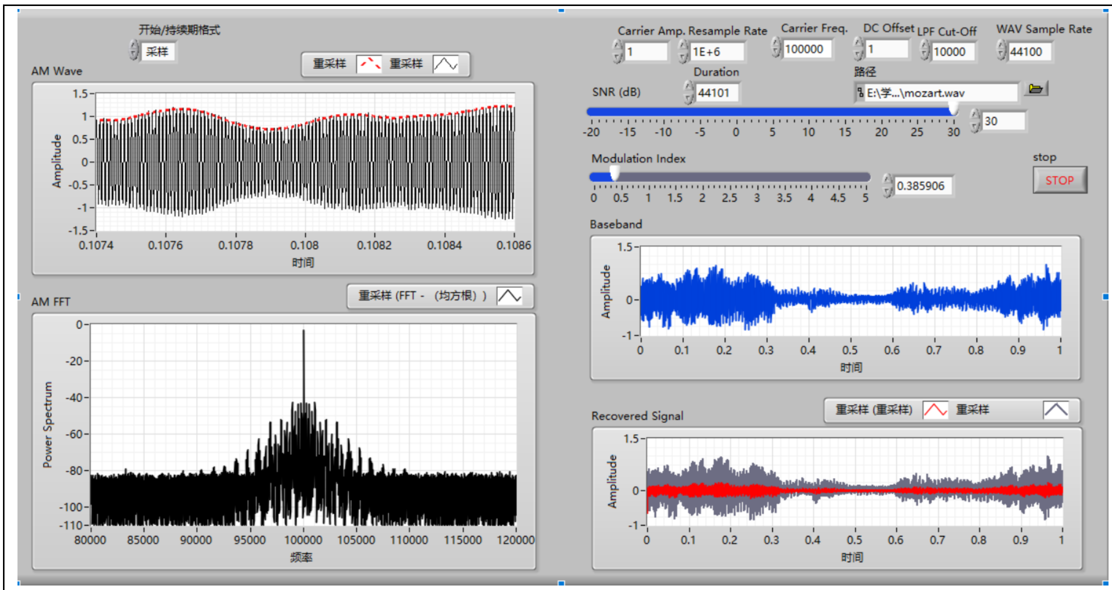


It can be seen from the envelope diagram that if we overmodulate (that is, the Modulation Index is greater than 1), the envelope will take the negative part when the envelope is detected and demodulated, which leads to distortion of the restored original signal..

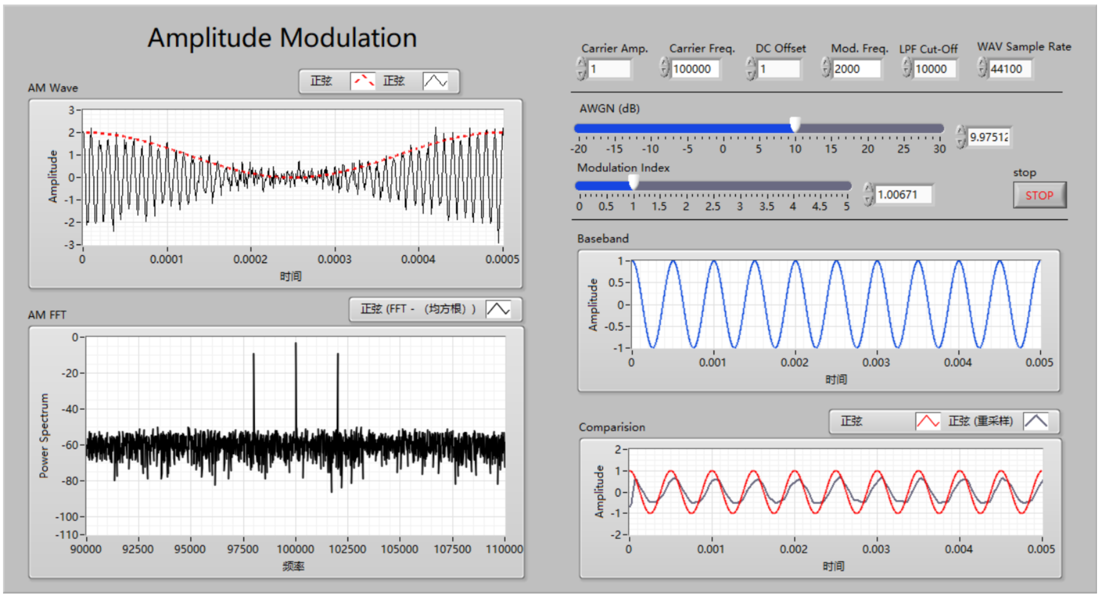


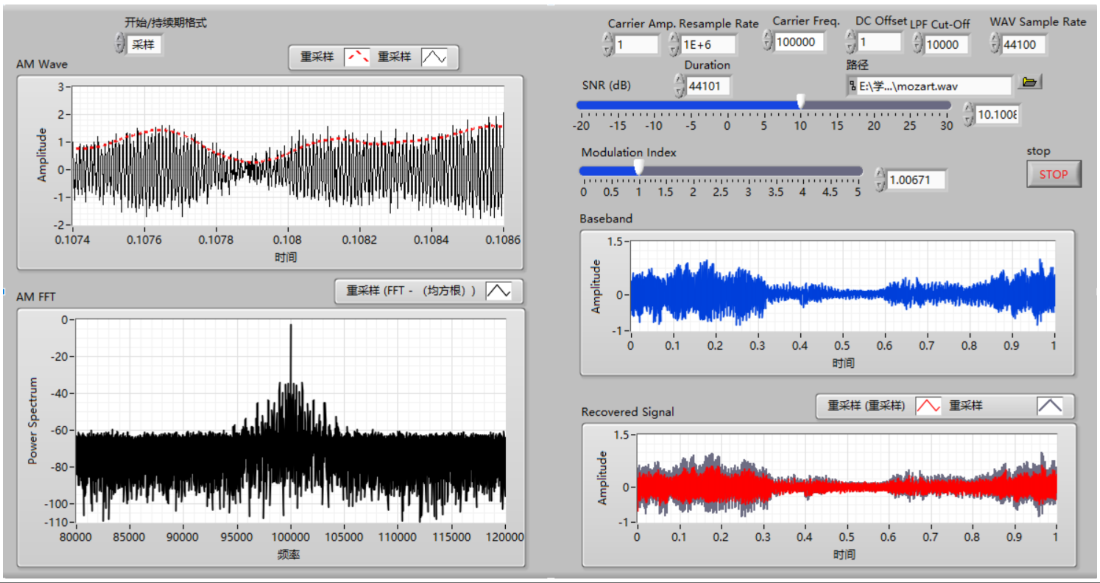
If it is equal to 1, the original signal will be better recovered. If it is less than 1, a constant greater than 1 needs to be multiplied in the final demodulation, so that the demodulated signal can be restored to the original energy, and the energy is too small to avoid listening not sure.

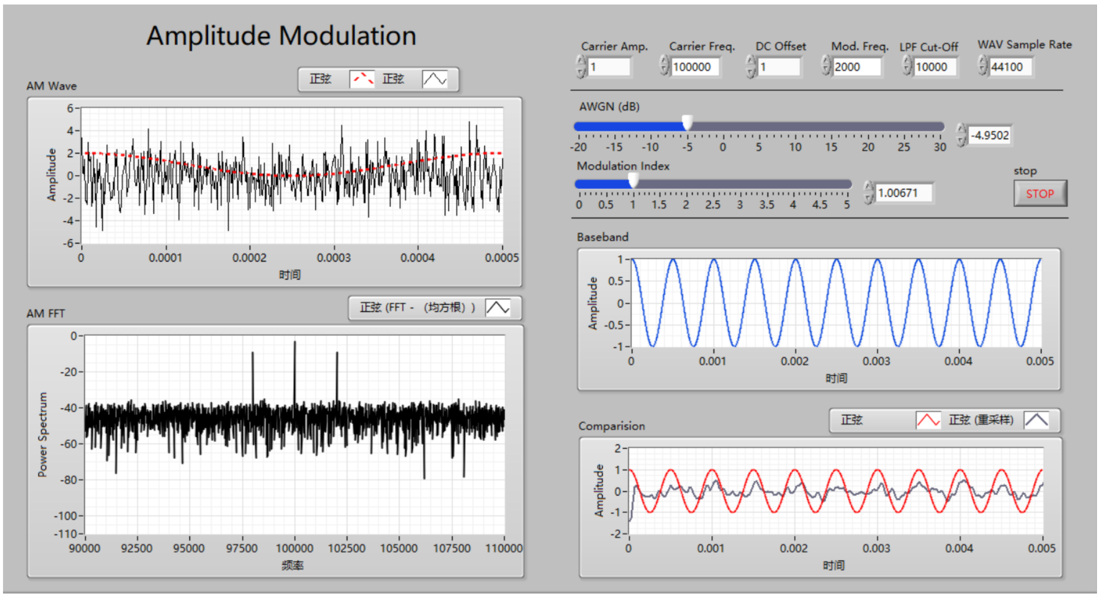


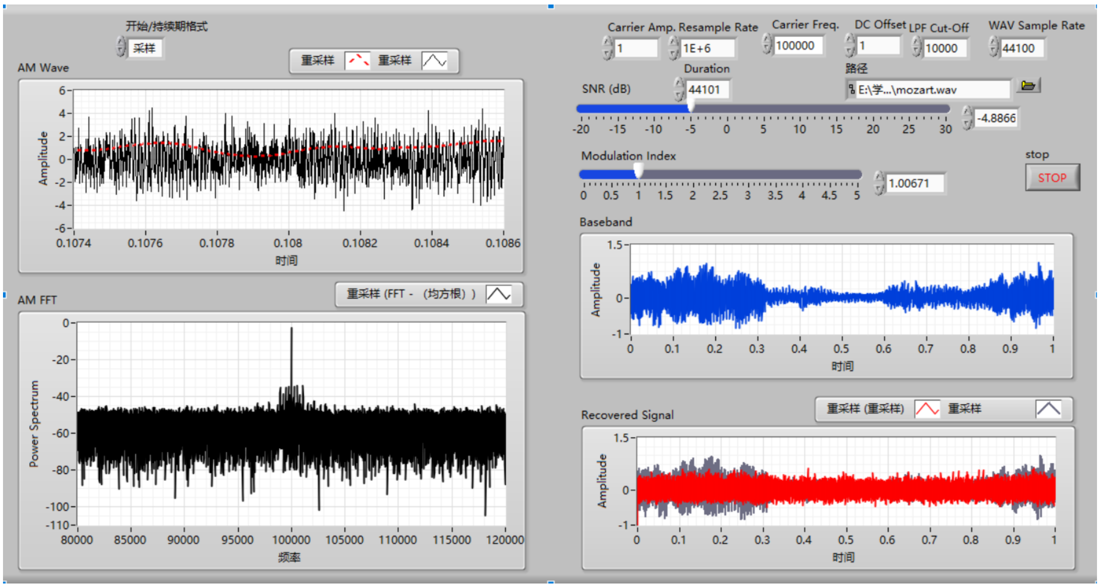


2. White noise（SNR）





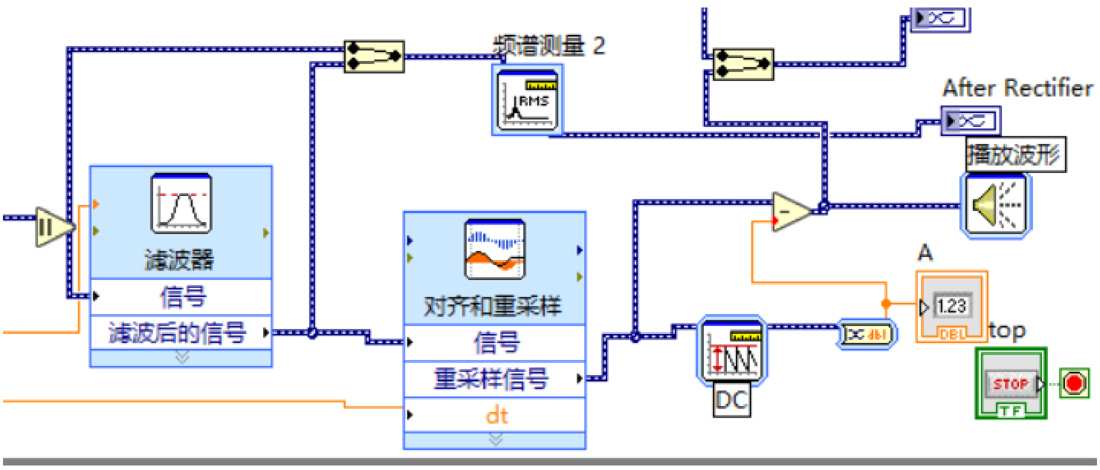




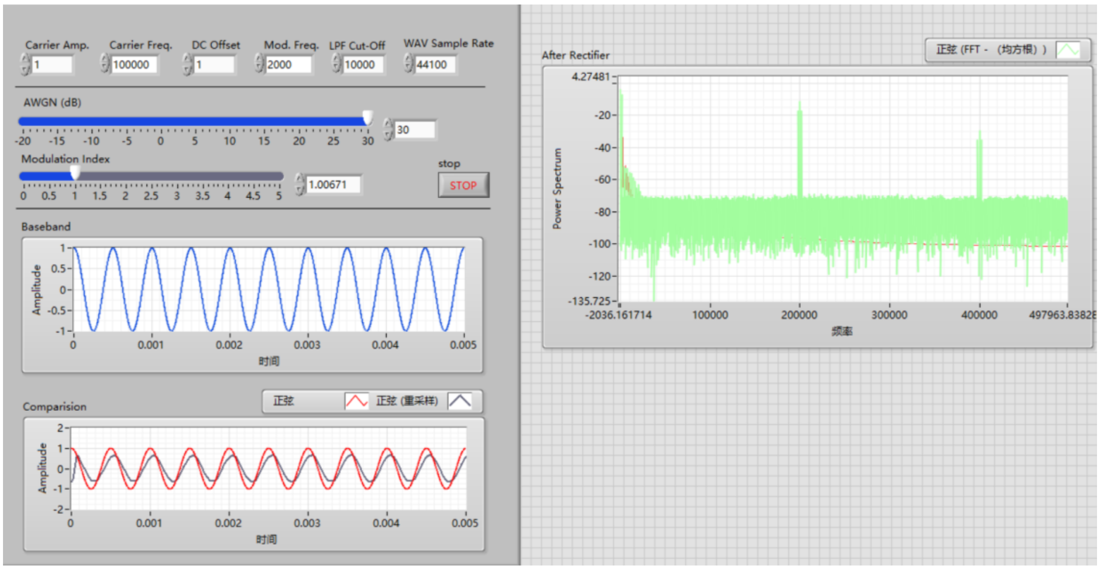
From the image, we can see that when the signal-to-noise ratio SNR=10dB, -5dB, there will be huge noise in the signal, and the audition effect of the audio restoration is very poor, and there will be rustling interference sound. It can be seen that when the noise is stronger than the signal Sometimes, the demodulation almost fails. The time-domain image has almost nothing to do with the original signal. The frequency-domain image after Fourier transform shows that although there is still a peak at the frequency corresponding to the original signal, its energy is not enough to compare with Gaussian noise. Almost equivalent to Gaussian noise.

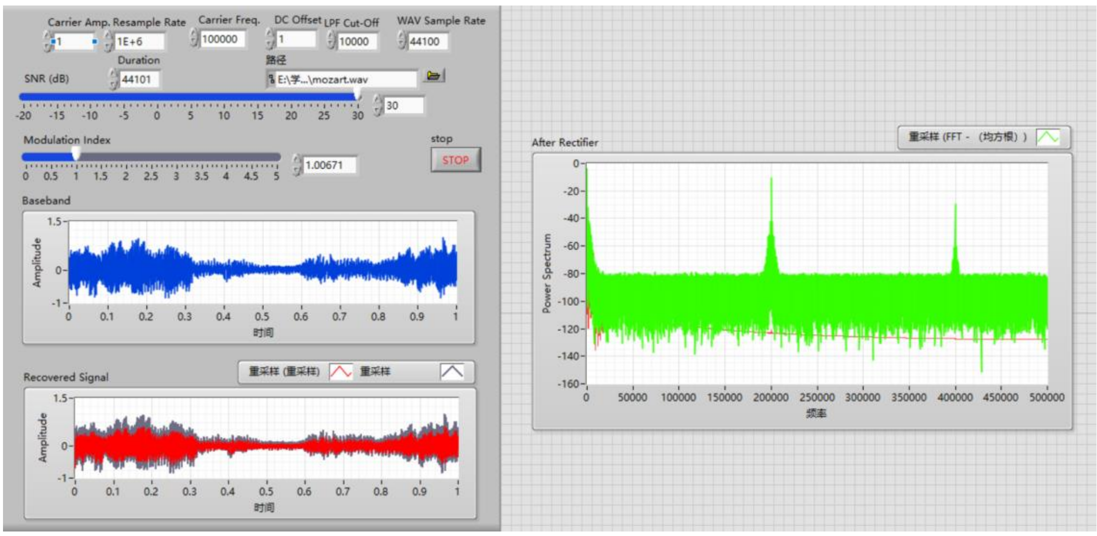
When the signal-to-noise ratio is greater than 1, the sound can be clearly heard as it gradually increases. Observing the envelope image, it can be seen that it gradually approaches the original signal, and the time and image gradually match the original signal. The components of the noise in the frequency domain gradually decrease.

3.Low pass filter cutoff frequency

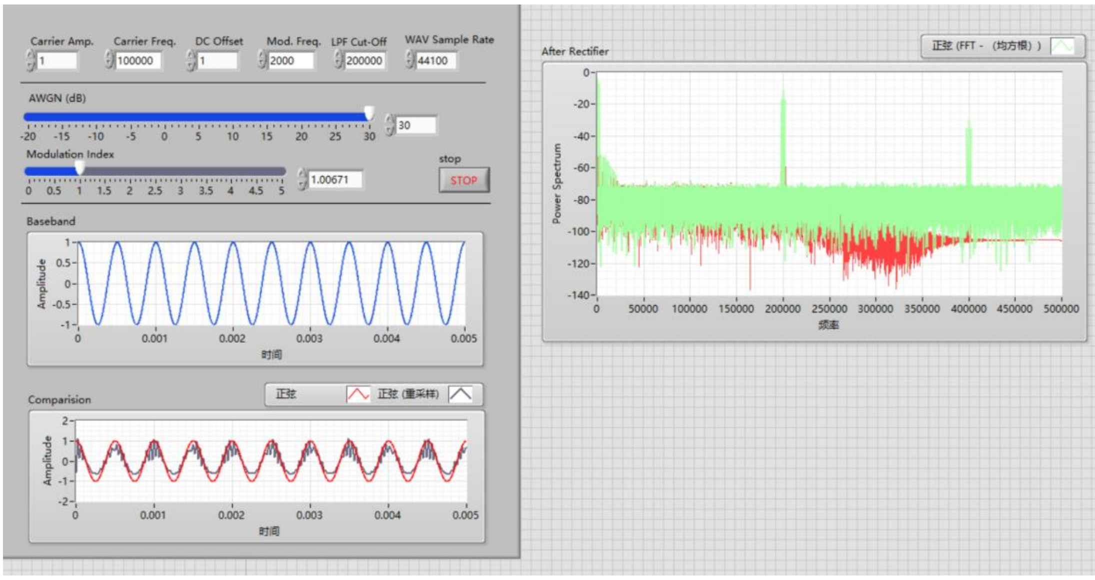


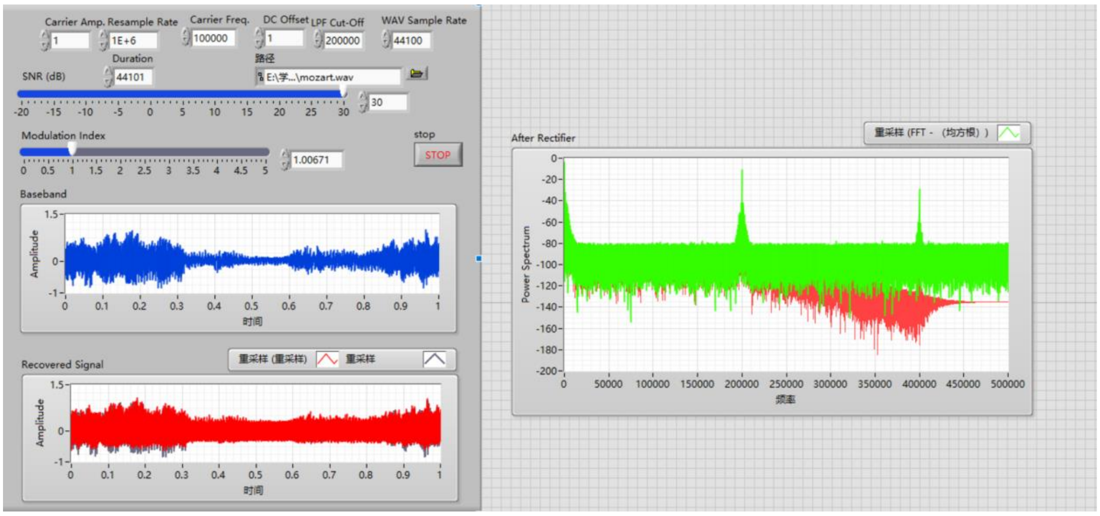
Here, the merge signal is used to compare the two output signals that have been filtered and those that have not been filtered on the same picture.





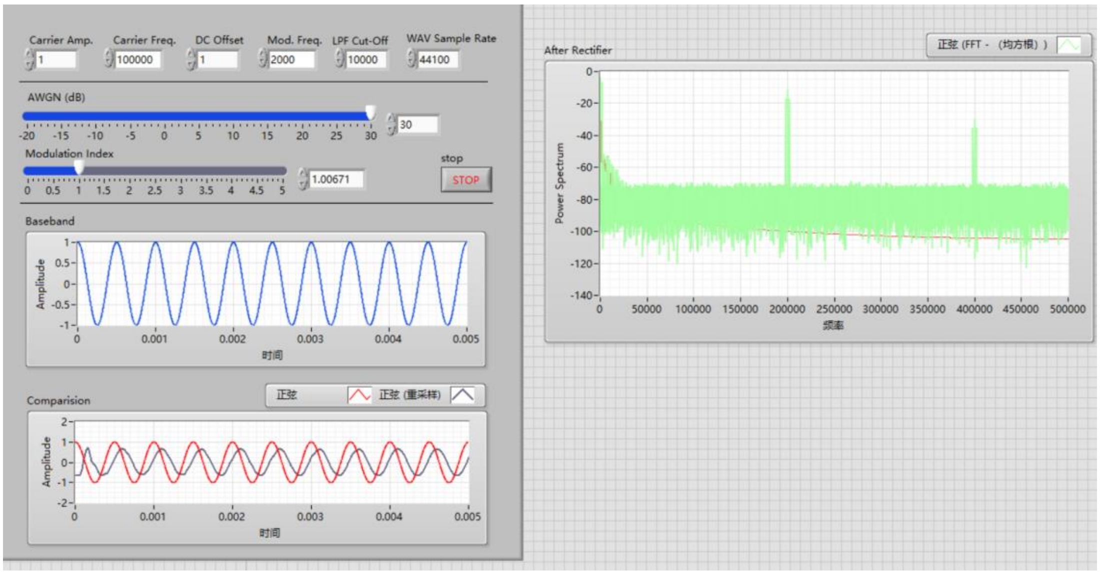
When the cutoff frequency is set to 10000Hz, the signal effect is better, and the spectrum after fft only contains the original signal part.

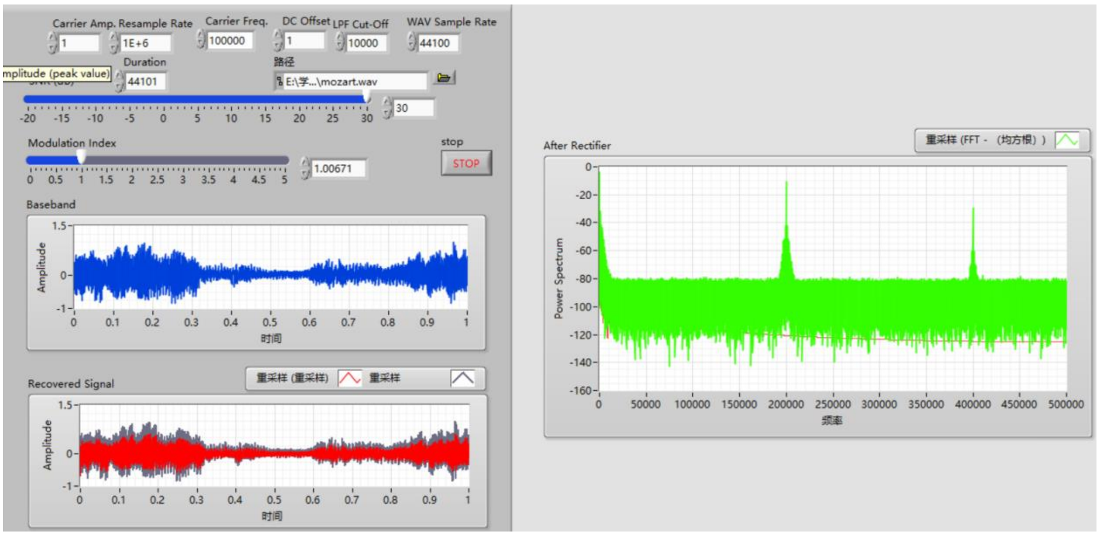




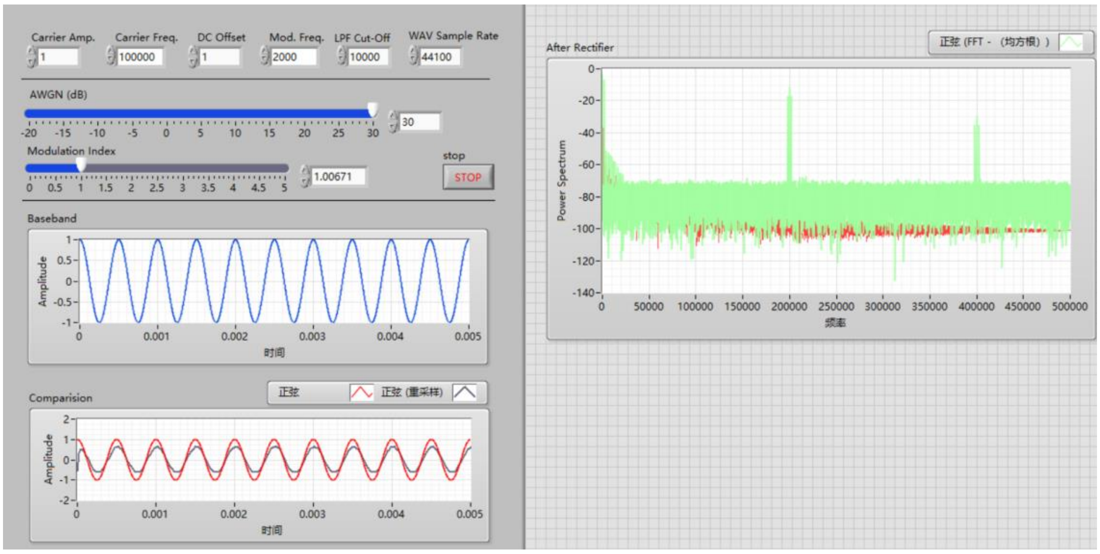
It can be seen that when the cut-off frequency is 200000Hz, the interference signal caused by full-wave rectification will not be filtered out, and the sound will not sound clear.

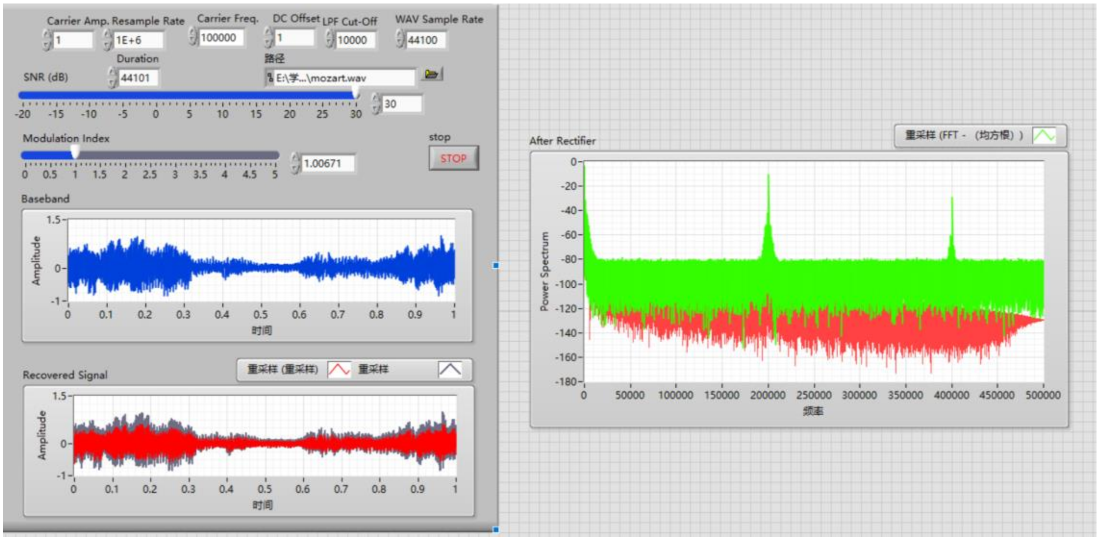
4filter order:





It can be seen that when the order of the Butterworth filter is adjusted to 10, there will be interfering high-frequency signals that will hinder the restoration of the original signal, and this part belongs to the filter's own noise and cannot be processed





In the same way, it can be seen that the inherent noise of the filter when the order is 1 still exists and cannot be processed.

Therefore, we use a low-pass filter with an order of 4 for the best effect.

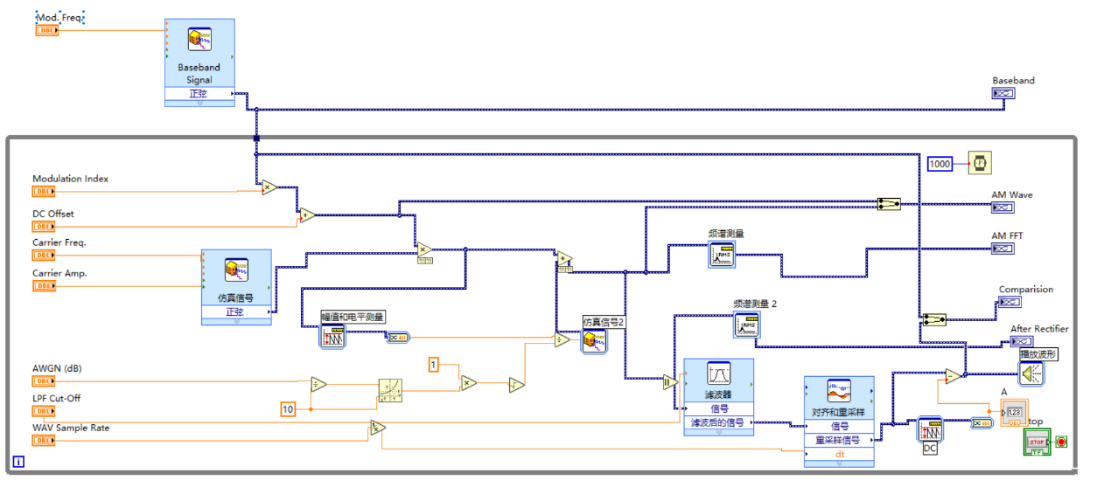
2.The advantages, disadvantages and values of AM

AM as a conventional and useful communication technology has many advantages. For example, the receiving device is simple, requiring only an envelope detector to restore the signal. And it has larger transmission power, wider coverage and it is suitable for long-distance transmission. However, there are also some disadvantages in AM. Firstly, the power utilization is low because both the carrier and baseband consume power, but only the baseband carries information. Secondly, the frequency spectrum is wasted because the signal bandwidth is wide and the upper and lower sideband are symmetric, but only one is needed to restore the signal. Thirdly, the anti-interference ability is poor. If the carrier encounters the selective fading of the channel in transmission, it will appear overmodulation distortion in envelope detection

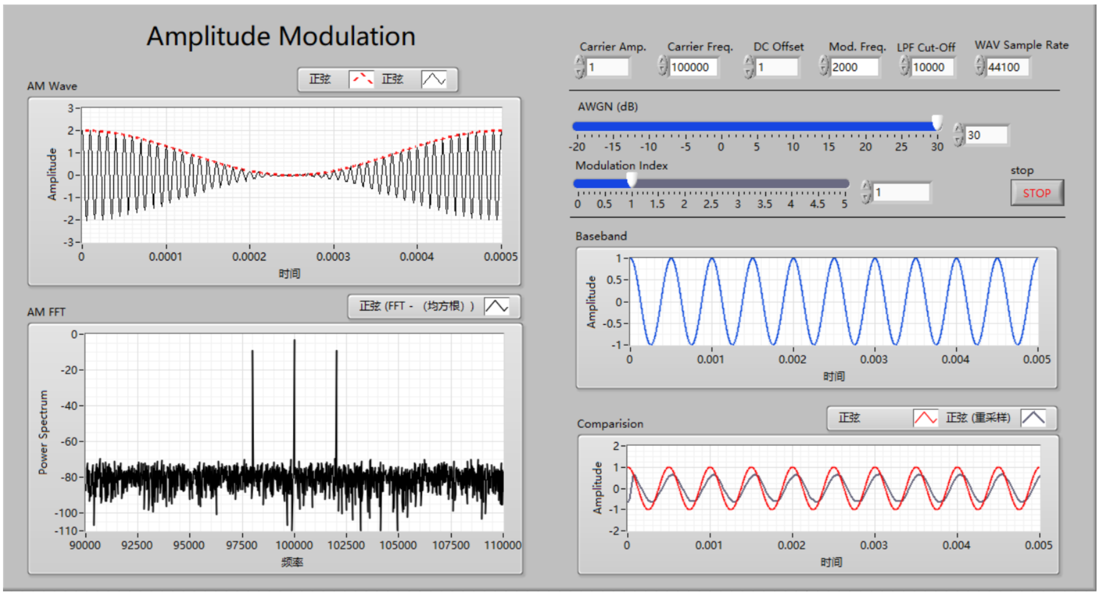
The advantages of AM make it have a wide range of uses, but its shortcomings also limited its application, in general, it produced a great social value. There three aspects. For broadcasting: AM modulation can spread over long distances in the medium and short-wave bands and is suitable for broadcasting services covering wide areas or across national boundaries. For communication: AM modulation allows voice transmission during radio calls using single sideband (SSB) or double sideband (DSB) mode, saving power and band resources. For radar: AM modulation can be used in radar systems for target detection and ranging using pulse amplitude modulation (PAM) mode.

Experience

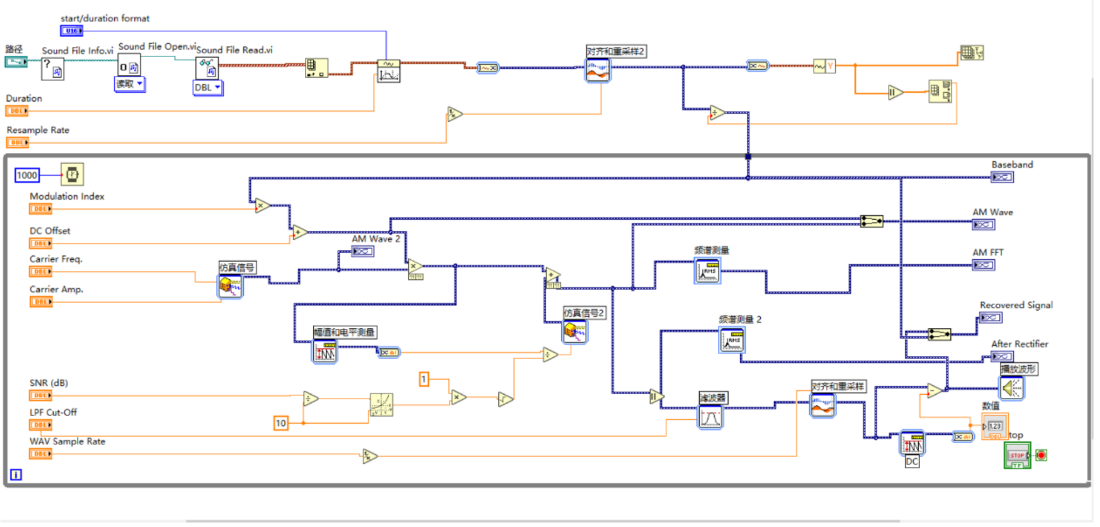
1.Single-tone test AM modulation program block diagram



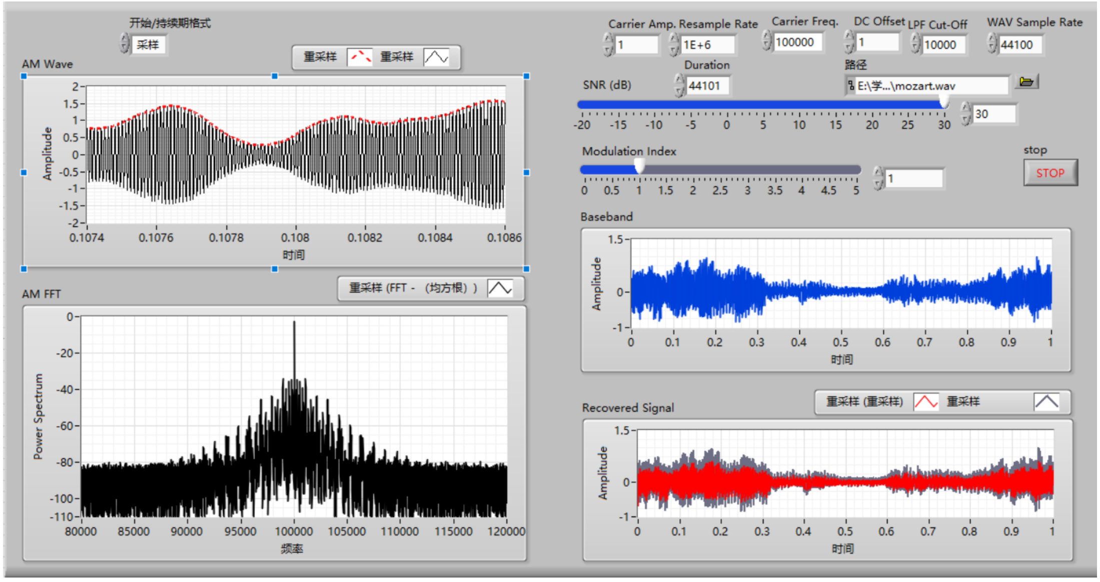
Simulation results：



Audio test program block diagram：



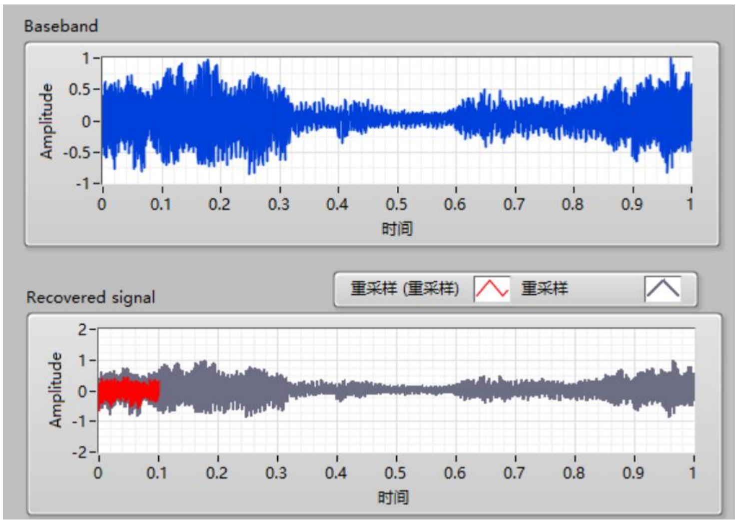
Simulation results：



2. The problem we met in this experiment

In this experiment, we encountered a little difficult problem. In the AM test experiment of speech signal, we first implemented the initial experiment according to the course knowledge and courseware content, but we found that the sound played was very short and the original voice signal could not be recovered completely after running. We extended the time range of the waveform and found that the recovered voice signal was only 0.1s, while the original voice signal was 1s. The graph of this result is at bottom. We found that normal 1s signal could be displayed in the Baseband graph, while there was only 0.1s waveform in the AM FFT image, so we speculated that there was something wrong when multiplying the carrier wave. Open the module of carrier generation, we find that the carrier signal is only 0.1s, which is the reason that the recovery signal is only 0.1s, so we continue to analyze the reason and find that the sampling rate is 1000000HZ, and the sampling number is one tenth of it, which leads to its only 0.1s. So, we're going to change the sample number by 1,000,000. And at the same time, we also change the sample number in noise generation. Therefore, this problem is solved.

In the process of solving this problem, our ability to analyze and solve the problem has been greatly improved, and deepened the understanding of AM communication system procedures. We're also a little bit clearer about the concepts of sampling rate and sampling number.



In addition to the above problems, we have also encountered the problem of unable to play sound for many times, which is caused by the different sampling rate Settings of different modules in the system. Therefore, we deepened our memory and understanding of this knowledge point.

3. Contribution

We two finish the whole labview program together. Song Yihang complete the bug patch about the 0.1s and 1s problem, and finish the Analysis of Factors Affecting AM Modulation System. The introduction and analysis of the advantages and disadvantages of AM modulation system were completed, and its social application value was elaborated by Zhang Haodong.

Contribution ratio :50%,50%

Score	自评分：98