Lab 3：FM

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Introduction

1. FM Modulation and Demodulation:

Modulation

The general expression for a carrier waveform can be shown as:

$$f(t) = A_c \cos[2\pi f_c t + \phi(t)] = A_c \cos\theta(t)$$

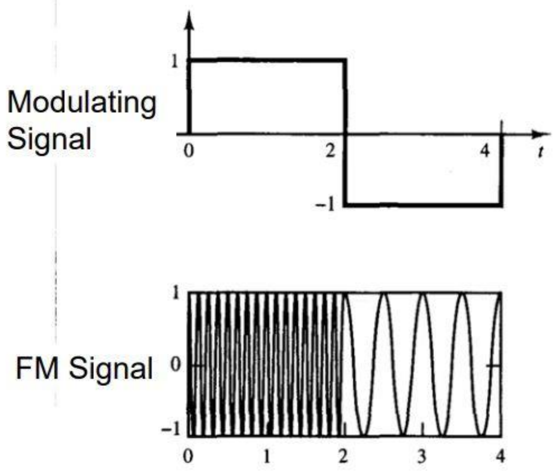
Where the $f_c(t)$ is the original carrier frequency. If the instantaneous carrier frequency $f_i(t)$ is varied linearly with the message signal $m(t)$, i.e., $f_i(t) = f_c(t) + k_f m(t)$ where $k_f m(t)$ is the instantaneous carrier frequency deviation, $\Delta f_i(t)$ and k_f is called as the frequency sensitivity that is a constant, this process is called the frequency modulation(FM). Using the relationship of the angle and frequency, the angle of the modulated signal can be obtained:

$$\theta_i(t) = 2\pi \int_{-\infty}^{t} f_i(\tau)d\tau = 2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} m(\tau)d\tau$$

Therefore, the frequency modulated (FM) signal is expressed as:

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

An example of FM can be given as follows:



As can be seen from this picture, the frequency of carrier signal is varied because of the message signal.

If the message signal is a single-tone signal, for example $m(t) = \cos(2\pi f_m t)$ then the FM signal can be written to:

$$f_{FM}(t) = A_c \cos[2\pi f_c t + \beta \sin 2\pi f_m t]$$

where $\beta \triangleq \frac{\Delta f}{f_m}$ is referred to as modulation index of FM.

According to the amplitude of β, the FM can be divided into two categories. When β≪1 (In general, β≤0.2), the FM is called NBFM (Narrow-Band FM), on the contrary FM is called as WBFM (Wide-Band FM). In NBFM, there are a lot of approximations as follows:

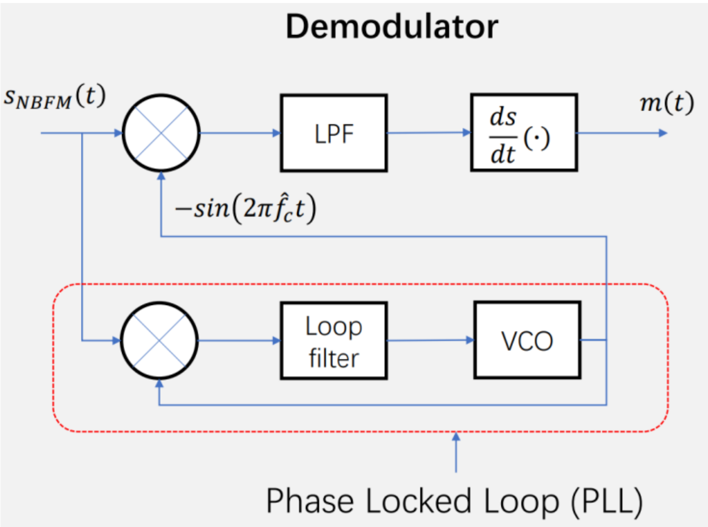
$$\cos(\beta \sin \omega_m t) \approx 1, \quad \sin(\beta \sin \omega_m t) \approx \beta \sin \omega_m t$$

The NBFM signal can be deduced to:

$$f_{NBFM}(t) \approx A_c[\cos \omega_c t - \beta \boxed{\sin \omega_m t \sin \omega_c t}]$$

Demodulation

For NBFM the demodulation is easy to implement, it only needs to multiply it by $-\sin(2\pi f_c t)$, then m(t) can be obtained after a LPF. The flow chart is shown as follows where the Phase Locked Loop (PLL) is used to obtain the frequency of the carrier signal which will be introduced on latter.

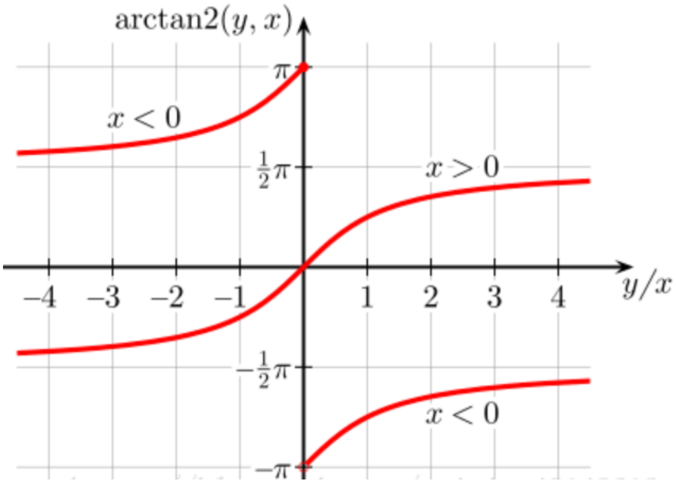


However, when β is larger enough, the approximations can not be used, so the demodulation is more complex in WBFM. One method is using the arctangent method whose principle is to firstly multiply sine and cosine terms in FM signal by sine and cosine respectively to demodulate the sine and cosine terms of message signal respectively. At this time, the tan term can be obtained by dividing these two terms. Then, the arctangent module is used to obtain the part in tan term, namely

$2\pi k_f \int_{-\infty}^{t} m(\tau)d\tau$, and then the derivative operation is carried out on it and remove the coefficient to get m (t).

2. Arctangent Method:

arctan(x) is used to calculate the angle of x=tan(θ), but the range domain of common function arctan(x) is from $-\pi/2$ to $\pi/2$, so it only deals with quadrants one and four. So in this experiment, we will use atan2(y, x) whose range domain is from $-\pi$ to π, so that it can handle any angle in the four quadrants. The image of this function is:

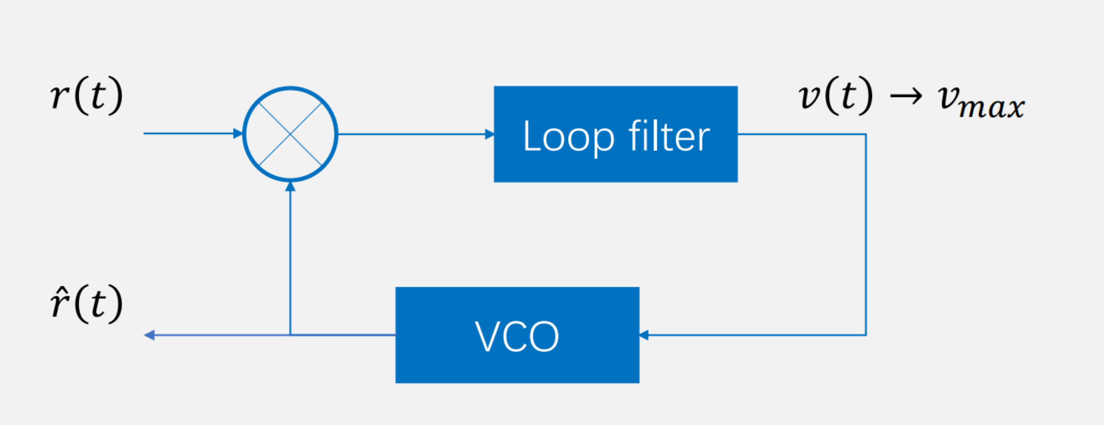


Its method of calculating is: when x>0, atan2(y/x) is namely equal to arctan(y/x), but when x<0 and y≥0, atan2(y/x) is equal to arctan(y/x) $+\pi$, when x<0 and y<0, atan2(y/x) is equal to arctan(y/x) $-\pi$.

3. Phase Locked Loop (PLL):

The PLL uses a feedback control circuit to track the frequency of the input signal so that the frequency of the output signal is the same as that of the input signal. When the frequency of the output signal is equal to the frequency of the input signal, the output voltage and the input voltage maintain a fixed phase difference, that is, the phase of the output voltage and the input voltage is locked. Therefore, in this experiment, we use phase-locked loop to obtain the carrier frequency and demodulate the signal of this frequency.

Phase-locked loop is usually composed of frequency discriminator loop filter and voltage-controlled oscillator. Frequency discriminator is also known as the phase comparator, its role is to detect the phase difference between the input signal and the output signal, its role is to detect the phase difference between the input signal and the output signal, the signal after the low-pass filter to form the control voltage uc (t) of the voltage controlled oscillator, the oscillator output signal frequency control, so that the frequency and input frequency close to, Until the final feedback signal and the frequency of the input signal is the same, that is, after the realization of locking, the phase difference between the two signals is a fixed stable value.

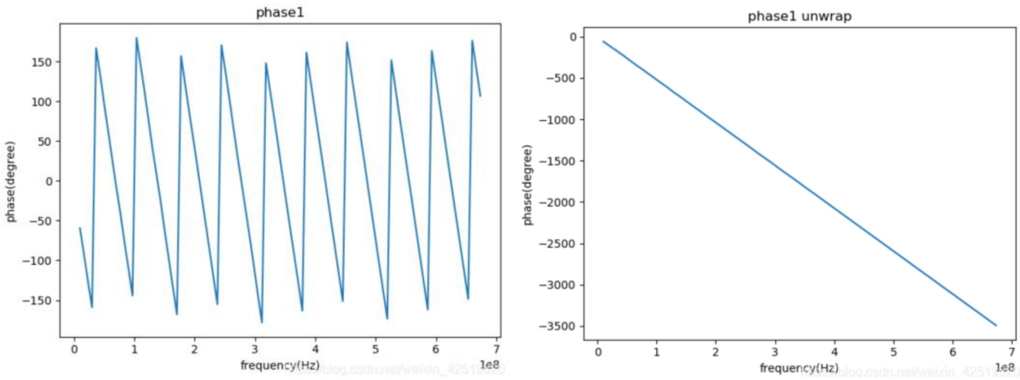


4. Voltage-controlled oscillator (VCO):

VCO is used to change the output frequency according to the input voltage. Many electronic applications require changing or controlling the frequency or phase of a signal depending on the amplitude of another signal. Typical applications include communication systems, frequency chirps in radar, and phase tracking in PLL. There are two types of VCO, one is harmonic oscillator which can generate the sinusoidal signals, another type is relaxation oscillator which can generate either a square, triangular, or sawtooth signal. In harmonic oscillator, there are three types: RC Oscillator, LC Oscillator, Crystal Oscillator. In RC Oscillator the frequency is $f = \frac{1}{2\pi RC}$, and the capacitors are variable capacitors, that is, they can vary according to voltage so the frequency can be changed by voltage. In LC Oscillator, $f = \frac{1}{2\pi\sqrt{LC}}$, so that is same as RC.

5. Unwrap Phase:

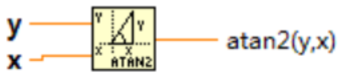
When calculating the Angle or phase, the system will set the phase between -π and π or -180 degrees to 180 degrees, so when its value crosses plus or minus π or 180 degrees, it jumps to the boundary and recalculates according to its trend. This state is called as “the wrap of phase”. Therefore, we should do unwrap on phase that means to eliminate this jump. That is, when the phase goes beyond this boundary, it is compensated by adding or subtracting 2π or 360 degrees to eliminate the jump. The effect before and after the unwrap operation can be seen below:



LabVIEW Express Module

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 that will be introduced as follows:

Atan2 Module:



This module uses the arc-tangent method to get the Angle in tan(θ). The specific principle has been described above. atan2(y, x) is the arctangent of y and x, in radians. The value range of atan2(y, x) is [-π, π]. y can be a scalar number, an array of numbers, or a cluster of numbers, an array of numbers, etc. x can be a scalar value, a numeric array or cluster, an array of numeric clusters, and many other data types.

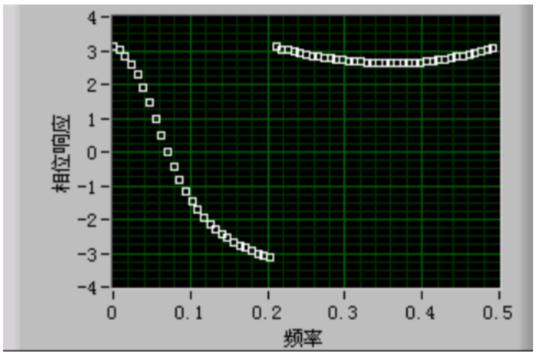
Unwrap Phase Module

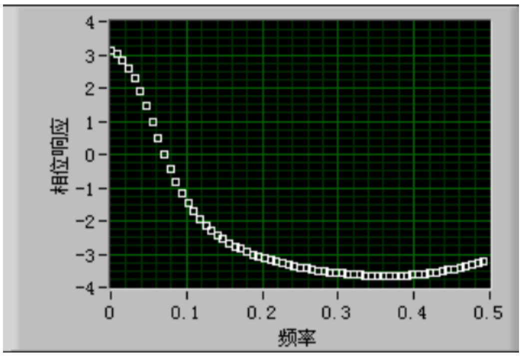
The module is to delete the discontinuous part whose absolute value exceeds π or 180, and expand the phase array according to the phase unit. For example, if the difference between adjacent values in the phase is greater than pi, the unit of phase is radian input and radian output, and the module is to calculate and expand the phase through the following equation:

$$P\_Out[\mathrm{i}] = \begin{cases} P[i] - \left\lfloor \dfrac{P[i] - P\_Out[i-1]}{2\pi} + 0.5 \right\rfloor * 2\pi & i = 1, \dots, N-1 \\ P[i] & i = 0 \end{cases}$$

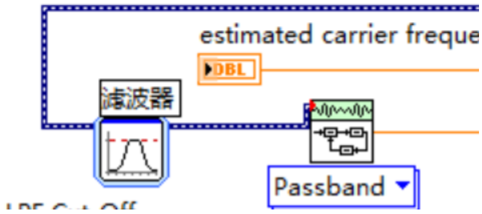
P_out is the unwrapping phase, P is the phase, N is the length of the phase, $\lfloor\,\rfloor$ is the floor operation.

The images below show the effects before and after the phase expansion. The first diagram is the unexpanded phase, and the second diagram is the expanded phase.



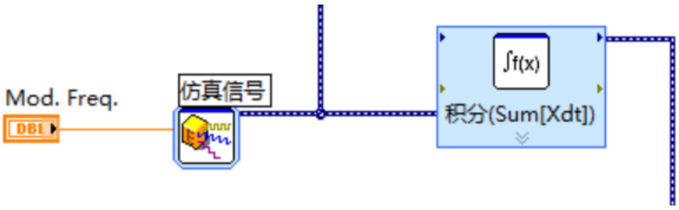


Phase Locked Loop (PLL) Module：



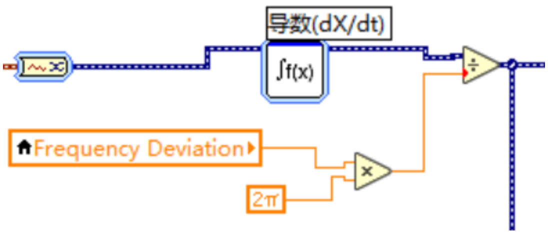
In this experiment, the carrier frequency is obtained by phase-locked loop and output the frequency to obtain sine and cosine signals of the frequency, to use the signal for partial demodulation. Its basic principle has been introduced above.

Integrator Module:



This module is used to integrate the message signal. Since m(t) changes the frequency, the change in phase is its integral. The integrated signal is transmitted to the following program to obtain the complete expression of FM signal.

Differentiator Module:



This module is used to conduct derivative operation on the signal, because the integral term m(t) is obtained after the arc tangent module and phase expansion module, so to obtain m(t), it is necessary to conduct derivative of it, and divide the signal after derivative by the coefficient to obtain m(t).

Spectral Measurement Module: 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.

Simulated Signal Generation Module: 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 cosine and sine signal as carrier, and we also need the cosine and sine signal with same frequency to demodulate. The cosine signal's phase is set as 90, and all sample rate are set to 1000000HZ.

Lowpass Filter Module: 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 demodulation of multiplying cosine or sine signal.

Bandpass Filter Module: this module is a filter to obtain the modulated signal around the carrier frequency f_c which purpose is to obtain the carrier frequency f_c in PLL module after that.

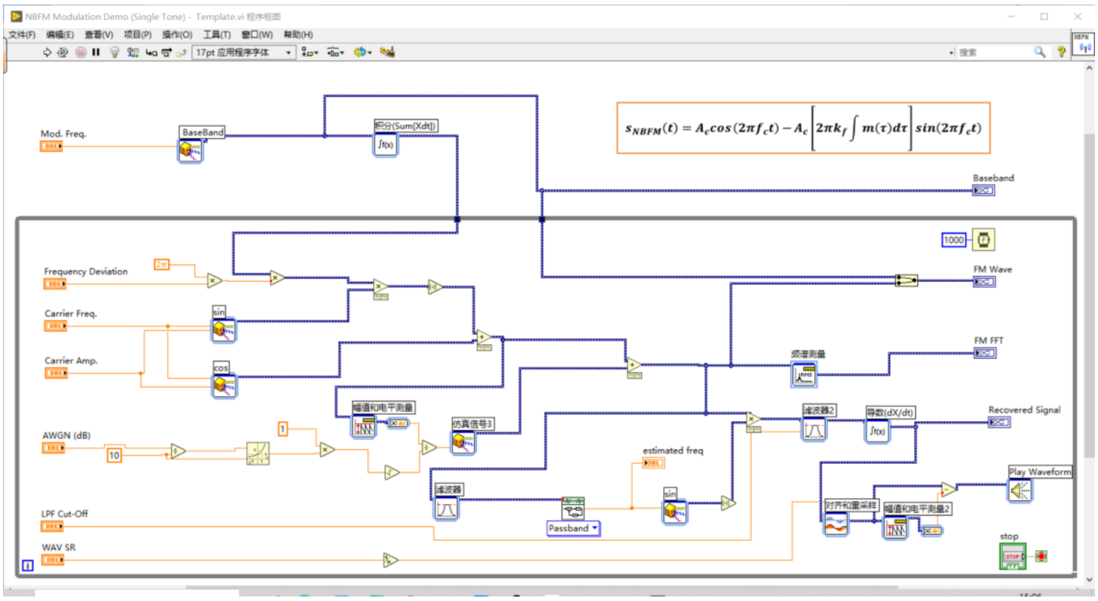
Align and Resample Module: there are two modules, one is after the filter and the other is used to obtain the baseband signal in the Mozart music. In fact, the signal after the filter cannot be played in the sound player because the sound player in this experiment can only play the signal whose sampling 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.

Lab results & Analysis:

1. NBFM

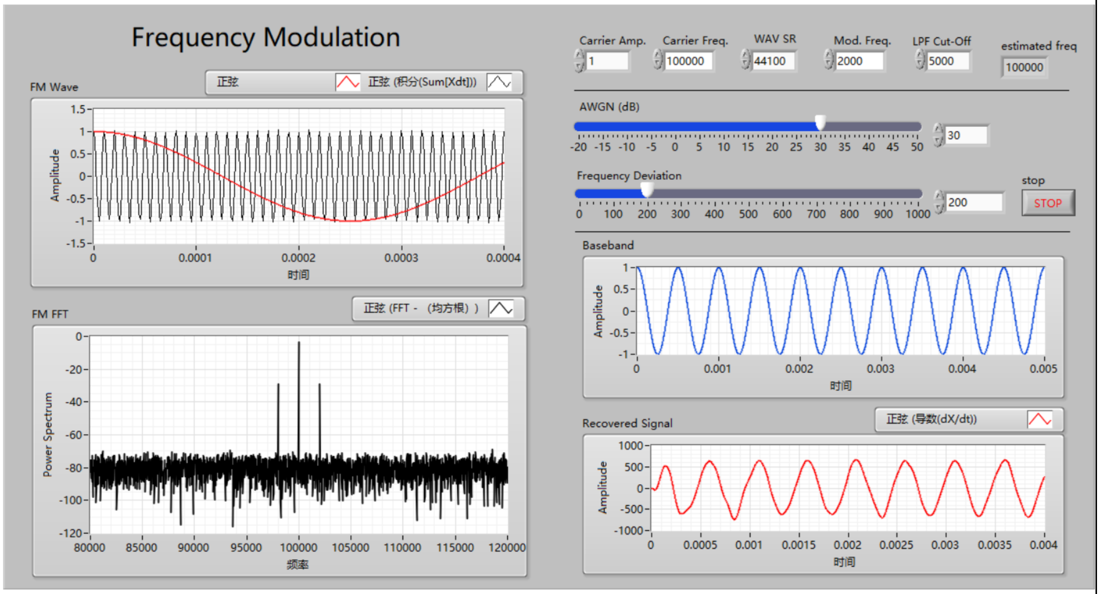
i. **single tone**

(1) Block diagram



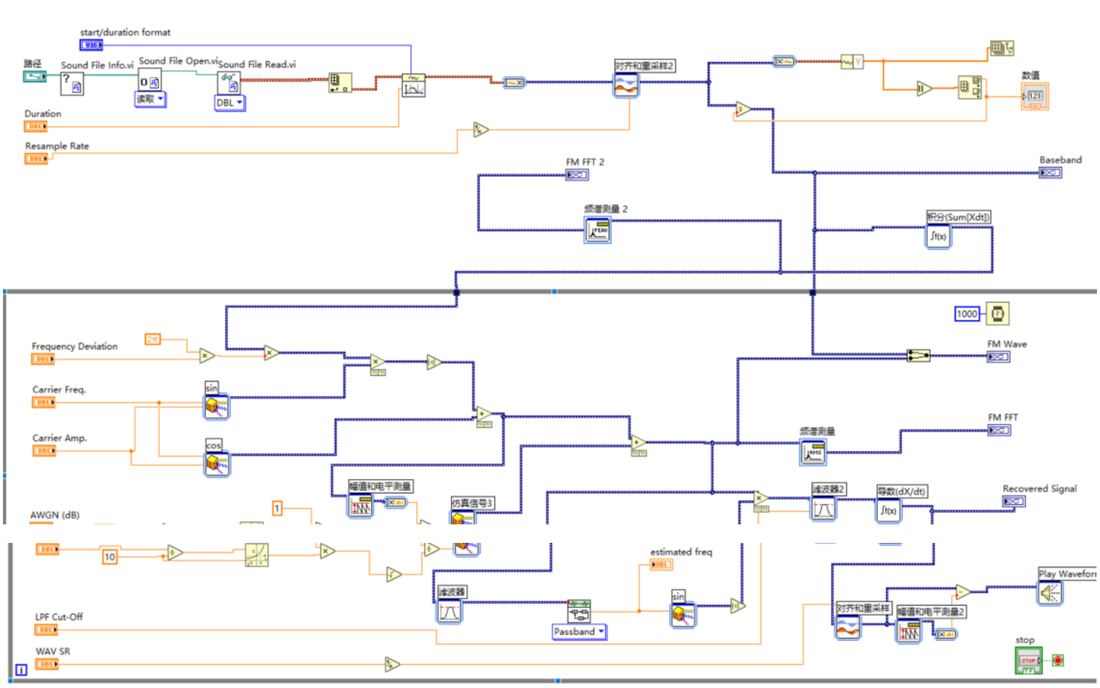
We generate the signal of narrow band frequency modulation by using integral and mixer. After adding Gauss white noise, we take our FM signal into the PLL so that we get the carrier frequency. Then we use DSB demodulation method and a LPF and make signal go through the differential module to rebuild message signal(cosine signal)

(2) Simulation result

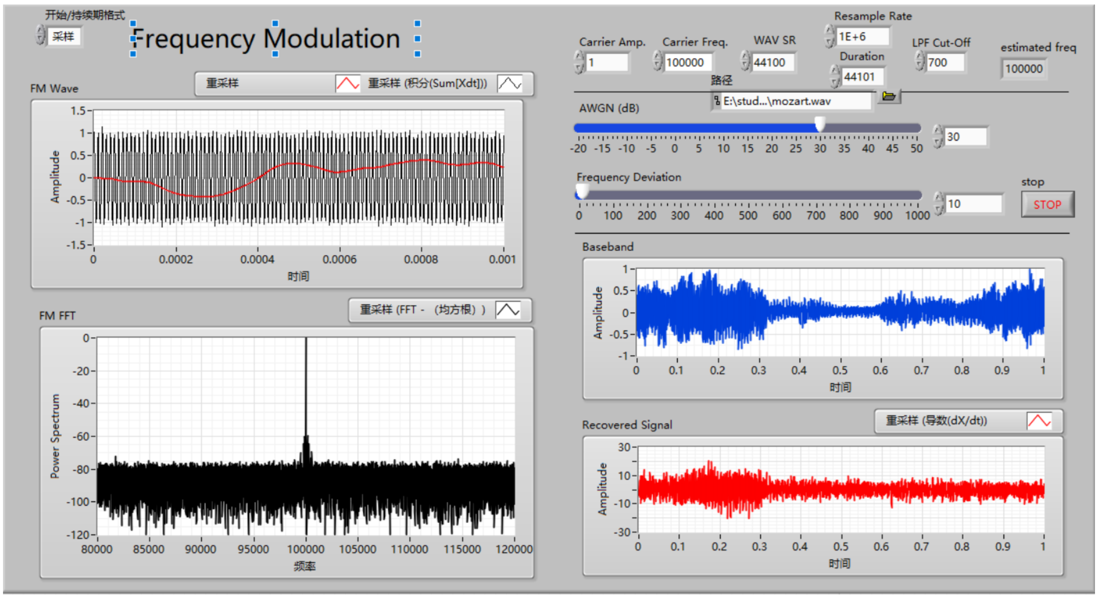


The recovered signal is similar with the baseband signal

ii. **Music signal in NBFM**



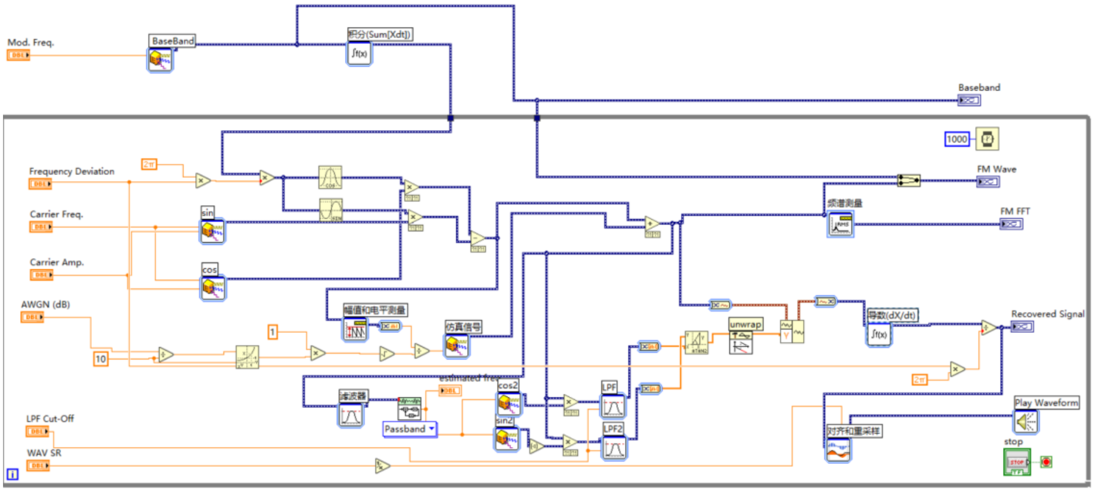
Simulation result



Due to the integral and the differential and demodulate by mixing a sin signal, we get a bad recover result signal in the end.

2. N/WBFM

i. Single tone by arctan2



We use

$$s_{FM}(t) = A_c\cos[2\pi f_c t + 2\pi k_f \int m(t)dt]$$

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

$$s_{FM}(t) = A_c \cos\left[2\pi k_f \int m(\tau)d\tau\right]\cos(2\pi f_c t) - A_c \sin\left[2\pi k_f \int m(\tau)d\tau\right]\sin(2\pi f_c t)$$

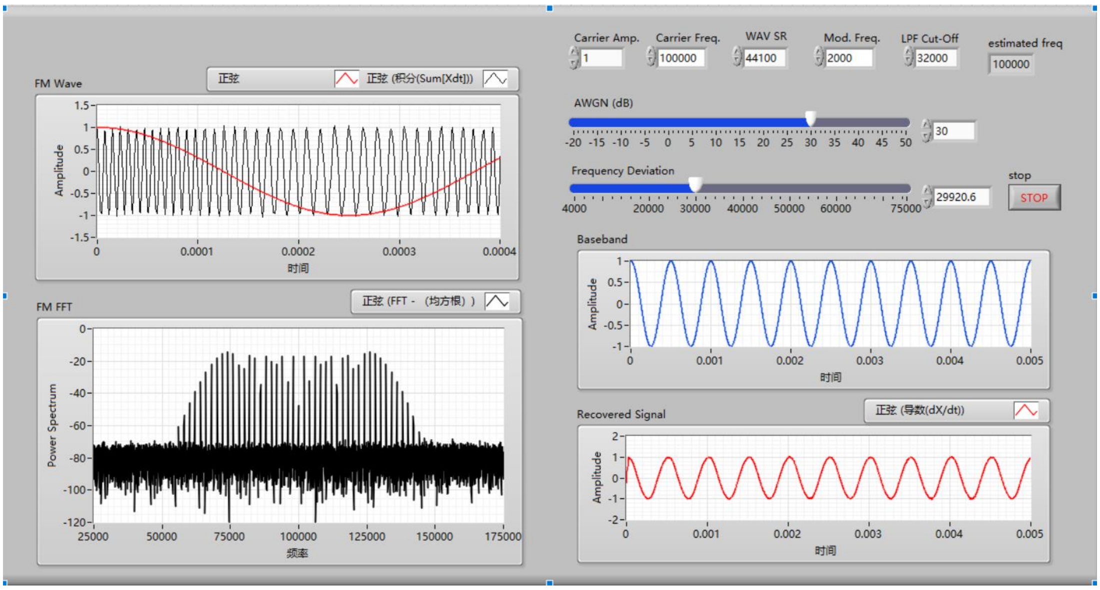
To generate our transmitting FM signal.And go through a BPF and PLL, we get carrier frequency. To get the carrier signal and recover

$$\cos\left[2\pi k_f \int m(\tau)d\tau\right]$$

and

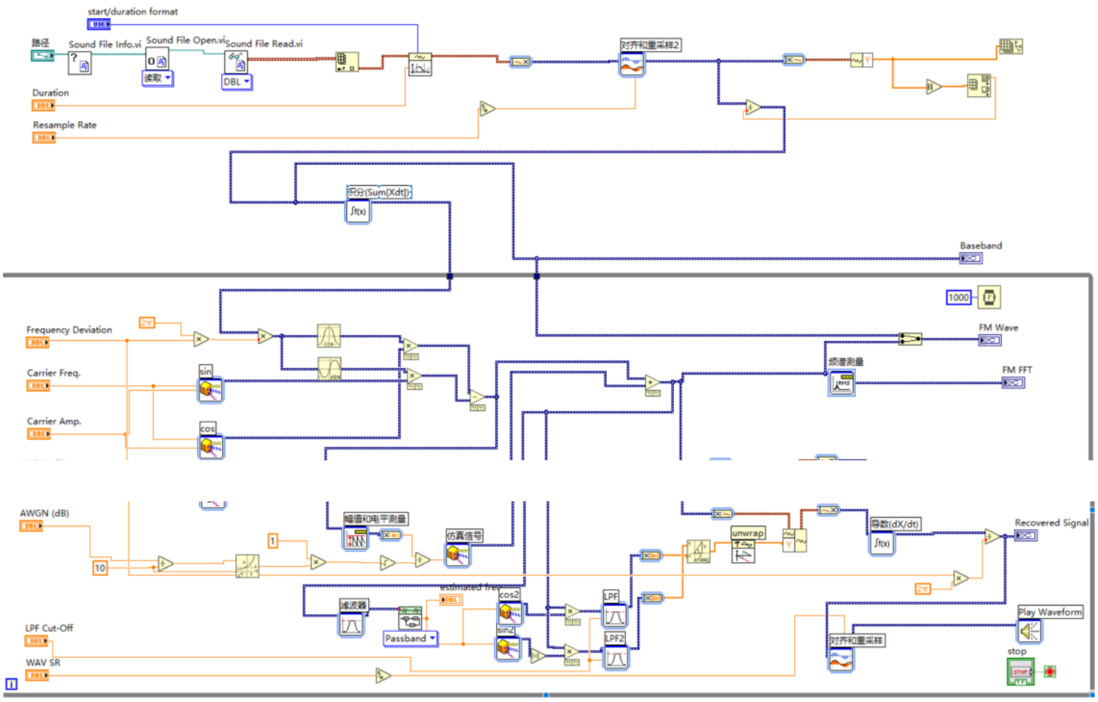
$$\sin\left[2\pi k_f \int m(\tau)d\tau\right]$$

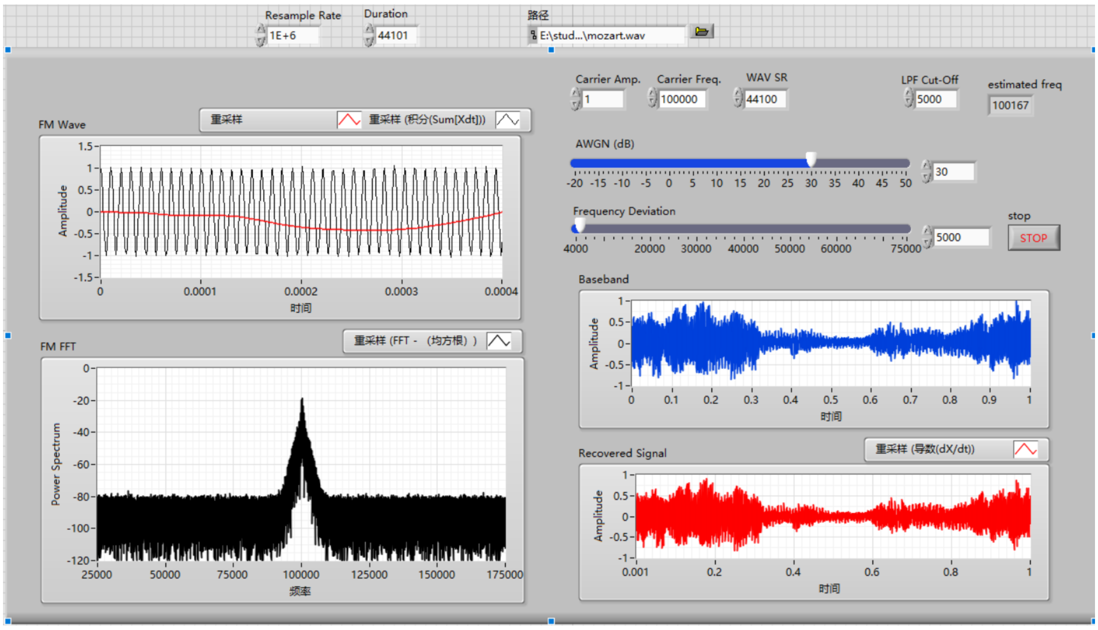
After an arctan2 module and go through the unwrap module so that we get the angle inside the tan. So after differential we can finally get the message signal.



ii. Music signal by arctan2

Block Program

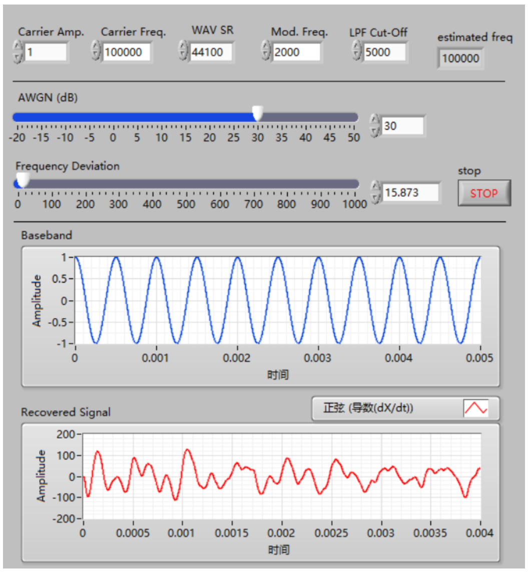




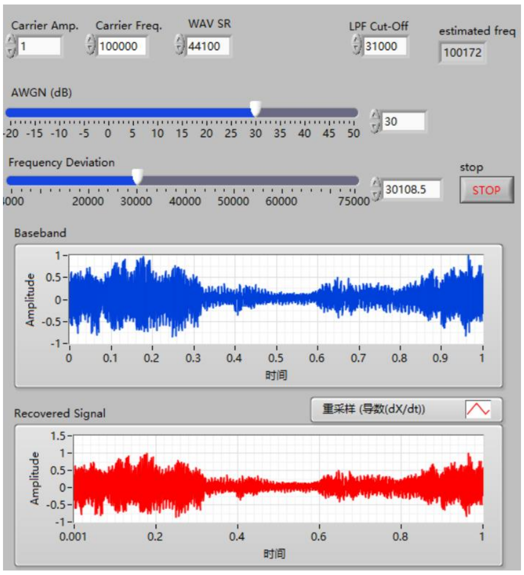
3.Factors affecting the FM modulation system

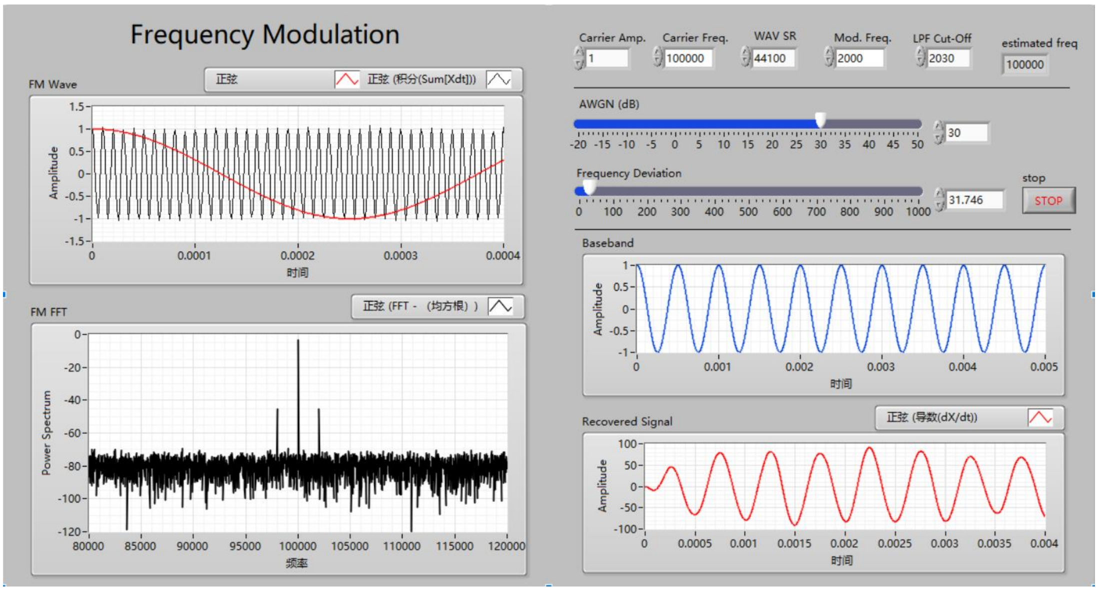
maximum frequency deviation

（1）Because the amplitude of a message signal is normalized to a maximum of 1, it can be approximated as a frequency deviation for k_f. In NBFM, because to ensure the β < 0.2 requirements, so the frequency offset should be as small as possible, for single-frequency signals, f_m certainly, to achieve NBFM, must be the smaller the frequency offset, the better, but because of the actual channel propagation there will be noise interference, such as additive Gaussian white noise, will cause the power of the signal when demodulated close to the power of the noise after passing through the filter, resulting in a relatively small SNR_out, the recovered signal will produce glitches, and the recovery effect is not good.



（2）In the audio test, using WBFM, it is found that the frequency offset and the cutoff frequency of LPF jointly determine the restored signal effect, because with arctangent, the value of the β does not affect the modulation effect, we mix a cos and sin for the FM signal to get the nested sinusoidal function of the message signal again, the recovery of this function depends on the cutoff frequency set by our low-pass filter, f _cut=Δf + f_m, according to Crason's Setting the LPF cutoff frequency will get a better original signal, and the audio signal can be considered to be concentrated within 1000Hz

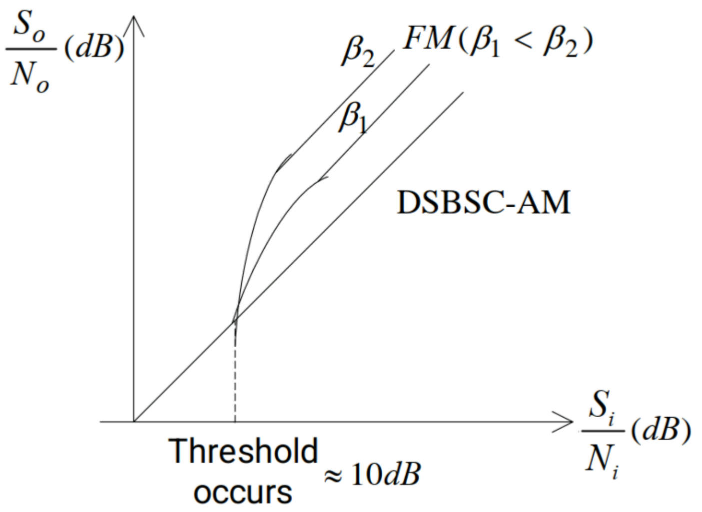




4.Performance Analysis of AM/DSB/FM System

The advantages and disadvantages of AM and DSB signal modulation have been explained in previous reports, FM signals can basically only propagate in a straight line due to their high frequency and short wavelength when the wave propagates.FM broadcasting is to determine a center frequency, and transmit the signal according to the deviation degree of the received actual frequency relative to the center frequency, so it will occupy a large section of frequency, and it is not suitable for AM broadcasting with narrow range and compact resources like AM broadcasting. Instead of using shortwave, ultrashort wave and microwave, which have a larger range and more abundant resources, cannot be reflected by the atmosphere, and will directly penetrate space, so the range is much smaller than that of AM broadcasting. Regarding anti-interference, since FM radio does not rely on the amplitude of radio waves to transmit signals, even if there is an interference source of the same frequency that disturbs the amplitude of radio waves, it will not cause noise to FM radio, and AM will not work, as long as there is an interference source When the amplitude of the radio wave is reached, the signal will be destroyed. Therefore, the sound quality of FM is better than that of AM

In practice, FM can be used for short-distance high-fidelity communication, the output signal-to-noise ratio can be effectively improved, and the restored original signal is clearer, subject to the threshold effect, the signal input signal-to-noise ratio drops to a certain level after the effect drops sharply, and the sound quality effect will be inferior to AM modulation



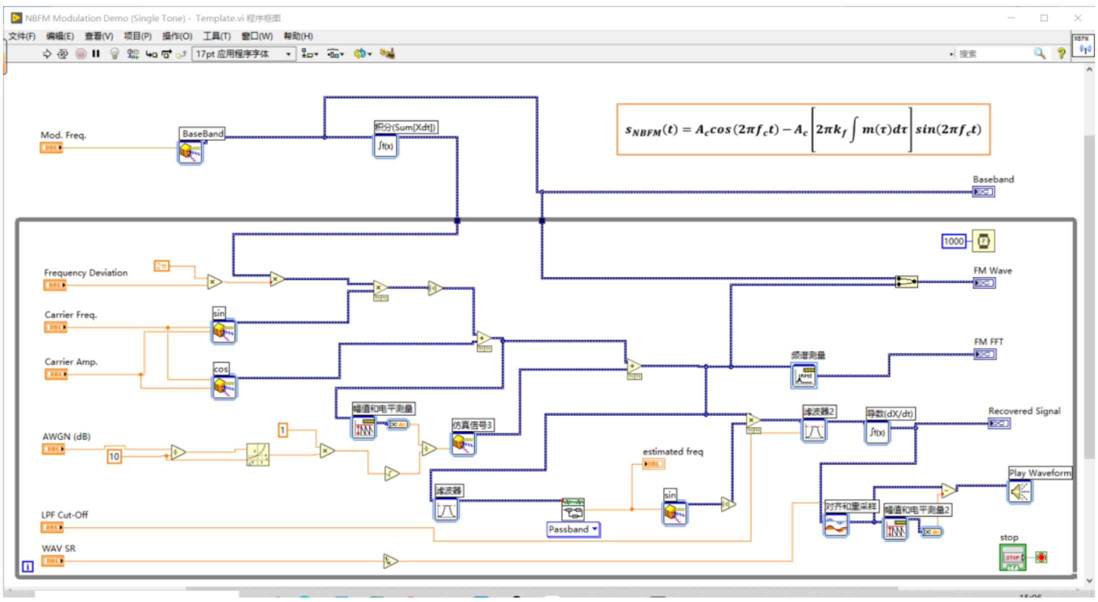
4.The advantages, disadvantages, and social values of FM

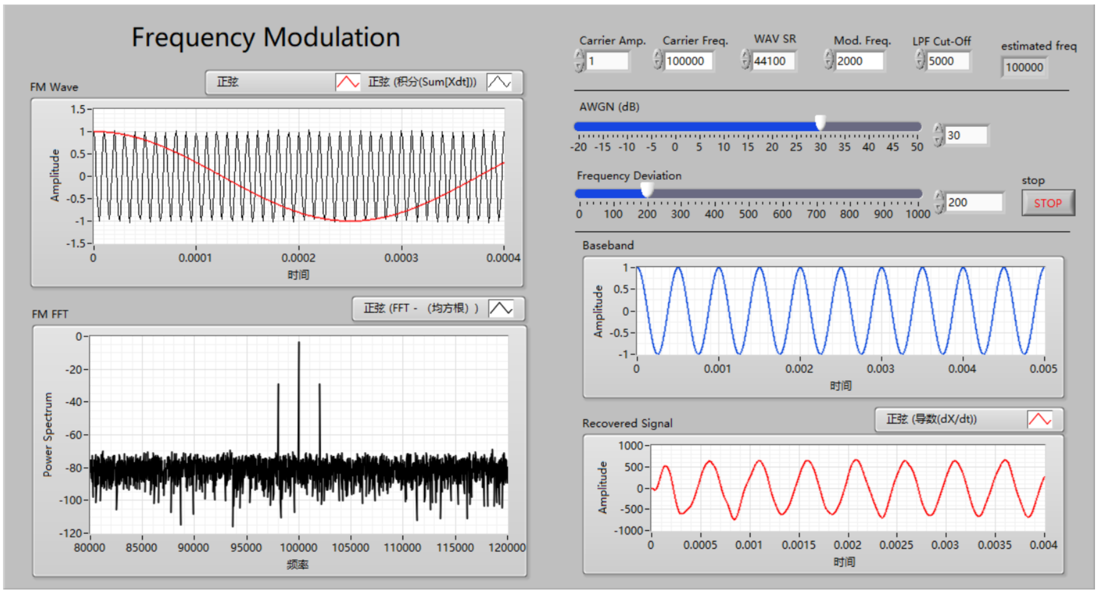
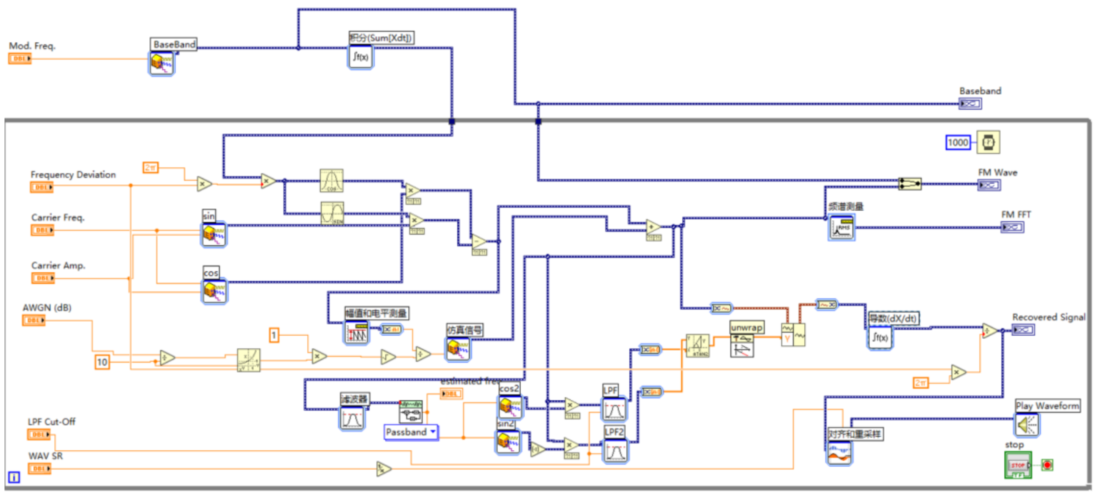
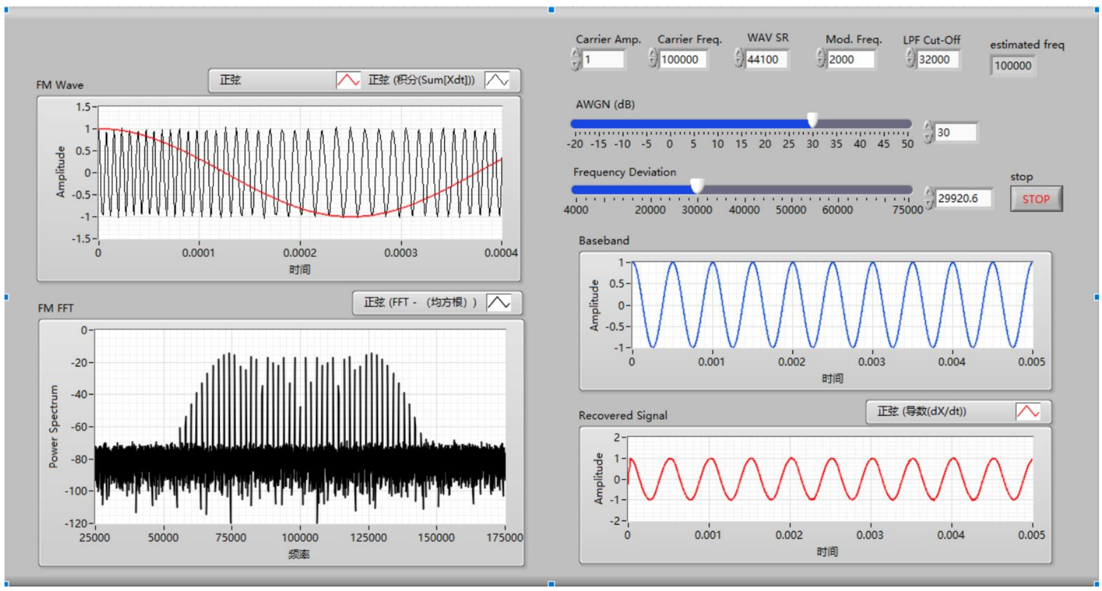
FM has many advantages: 1. Good resistance to interference: FM modulation has high tolerance to noise and interference, enabling clear signal transmission in complex wireless channel environments. 2. High audio quality: FM modulation has low distortion of audio signals, allowing for high-quality audio transmission, making it suitable for applications such as broadcasting and music with high audio quality requirements. There are also many disadvantages of FM system: 1. Larger bandwidth requirement: Compared to AM and DSB modulation, FM modulation requires a larger bandwidth for signal transmission, resulting in lower spectrum utilization efficiency. 2. Higher complexity: FM modulation involves relatively complex modulation and demodulation processes, requiring complex circuits and algorithms for implementation. 3. Higher cost: Compared to AM and DSB modulation, FM modulation has higher implementation costs, including complex circuits and algorithms, adding to the system cost.

FM has many applications and social value. There are some examples: 1. Broadcast Communication: FM broadcasting is a common wireless radio broadcasting method that uses frequency modulation to transmit audio signals with good audio quality and anti-interference performance. FM broadcasting is widely used in radio stations, music broadcasting, news broadcasting, weather forecasting, and other media communication, providing people with diverse information and entertainment content. 2. Wireless Communication: FM modulation is also applied in wireless communication. For example, some walkie-talkies and wireless microphone systems use FM modulation for voice transmission, providing a convenient wireless communication method for public safety, construction, performing arts, and other occasions. 3. Audio and Video Transmission: FM modulation is also used in audio and video transmission. For example, wireless headphones, wireless speakers, and other devices use FM modulation to transmit audio signals wirelessly, providing a wireless audio playback method. In addition, FM modulation can also be used for wireless video transmission, such as wireless cameras and monitoring systems.
Car Radio: FM modulation has widespread application in car radio. Car-mounted FM radio receivers can receive signals from various radio stations, providing drivers and passengers with music, news, traffic information, and other services, offering entertainment and practical functions. 4. Other Applications: In addition to the above applications, FM modulation is also used in other fields. For example, FM modulation can be used for data transmission in remote control devices, wireless sensor networks, and other systems. Moreover, FM modulation is also applied in aviation, aerospace, military communication, and other fields, providing wireless communication and data transmission means for relevant industries.

Experience

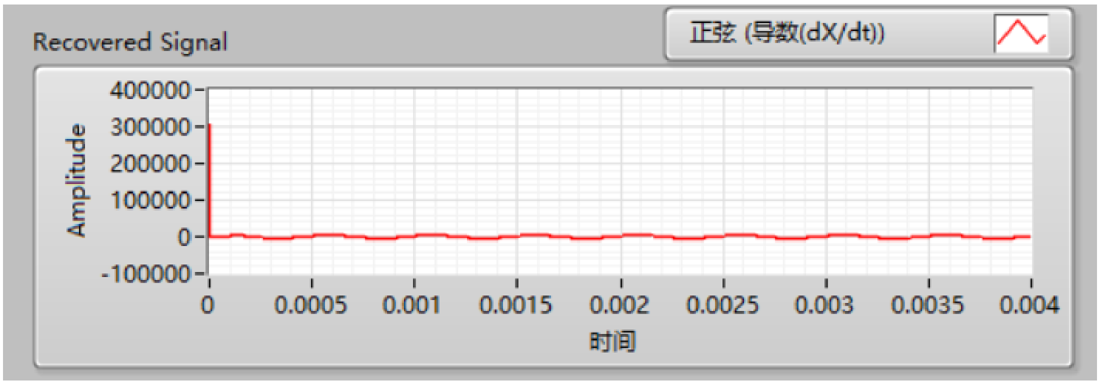
1.

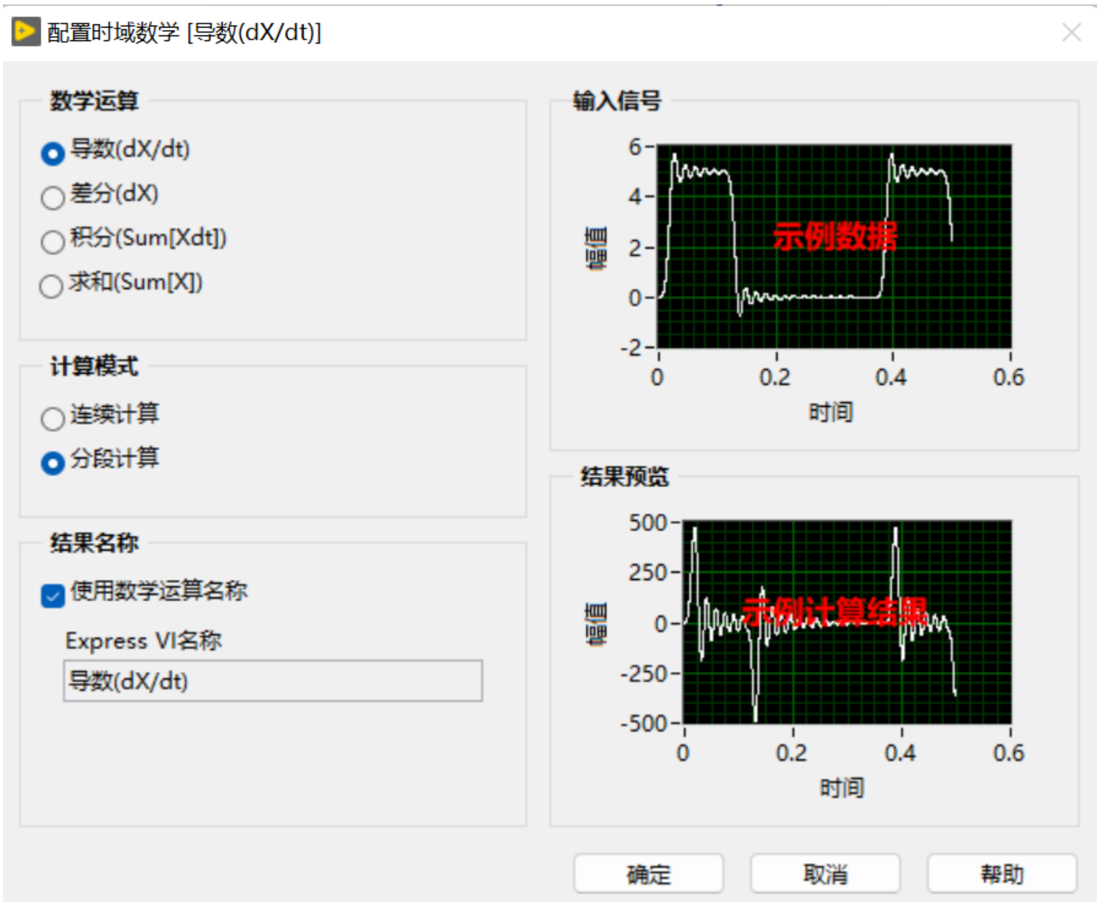



Frequency Modulation
FM Wave
正弦
正弦 (积分(Sum[Xdt]))
Amplitude
时间
FM FFT
正弦 (FFT - (均方根))
Power Spectrum
频率
Carrier Amp.
Carrier Freq.
WAV SR
Mod. Freq.
LPF Cut-Off
estimated freq
1
100000
44100
2000
5000
100000
AWGN (dB)
30
Frequency Deviation
200
stop
STOP
Baseband
Recovered Signal
正弦 (导数(dX/dt))

Mod. Freq.
BaseBand
积分(Sum[Xdt])
Baseband
1000
FM Wave
FM FFT
Frequency Deviation
Carrier Freq.
Carrier Amp.
AWGN (dB)
sin
cos
频谱测量
幅值和电平测量
仿真信号
unwrap
导数(dX/dt)
Recovered Signal
10
滤波器
Passband
estimated freq
cos2
sin2
LPF
LPF2
对齐和重采样
Play Waveform
LPF Cut-Off
WAV SR
stop

FM Wave
正弦
正弦 (积分(Sum[Xdt]))
Amplitude
时间
FM FFT
正弦 (FFT - (均方根))
Power Spectrum
频率
Carrier Amp.
Carrier Freq.
WAV SR
Mod. Freq.
LPF Cut-Off
estimated freq
1
100000
44100
2000
32000
100000
AWGN (dB)
30
Frequency Deviation
29920.6
stop
STOP
Baseband
Recovered Signal
正弦 (导数(dX/dt))

2. Problem we meet

1. In experiment, we found that the figure of our result is very strange like that picture below. After analyzing we found the differentiator needs to choose segmented derivation, which cannot be continuously derived, because a step will be generated from 0 to 1, resulting in a derivative near infinity near t=0.

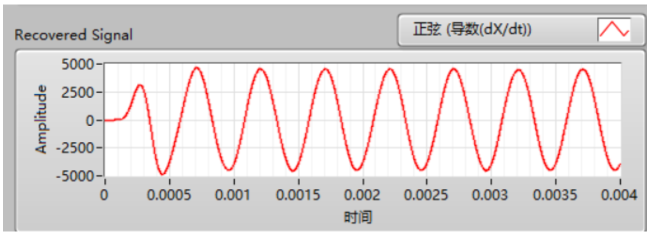




2. When testing audio, it is not easy to determine the f_m, but after measuring the spectrum, it is found that the energy of the audio signal is concentrated within 1000Hz, so it is better to set the f_m to 1000Hz. After adjusting the frequency offset, whether it is a single-frequency signal or an audio signal, the recovered signal effect is not good, and finally it is found that this is due to the cutoff frequency of the low-pass filter is not adjusted, if the cut-off frequency is same as the experiment last

time the effect of result is worse. That is because that the cut-off frequency of LPF should be set to larger because the bandwidth is larger after FM. Because of the Carson's rule the BW can be 2(β+1)f_m. So, the cut-off frequency should be set to larger to approach to 2(β+1)f_m. So that the cutoff frequency is adjusted and the recovery effect becomes better. However, due to the use of PLLs, the bandwidth of the band-pass filter does not need to be adjusted to Δf, and the appropriate value can be selected.

3. During the experiment, we also found a phenomenon: the filter delay will increase with the increase of the order.



3.Contribution

We two finish the whole LabVIEW program together. Song Yihang finish the Analysis of Factors Affecting FM Modulation System. The introduction and analysis of the advantages and disadvantages of FM modulation system were completed, and its social application value was elaborated by Zhang Haodong.

Contribution ratio :50%,50%

Score	97