Project：SDR FM

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Introduction

1.Theoretical basis

(1)FM modulation and demodulation principle

Modulation

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

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

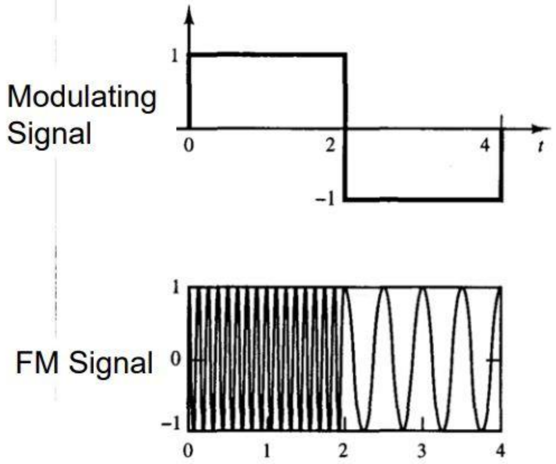
Where the $f_c(t)$ is the original carrier frequency. If the instantaneous carrier frequency $f_i(t)$ is varied linearly with the message signal $m(t)$, i.e., $f_i(t) = f_c(t) + k_f m(t)$ where $k_f m(t)$ is the instantaneous carrier frequency deviation, $\Delta f_i(t)$ and k_f is called as the frequency 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 $\beta \ll 1$ (In general, $\beta \le 0.2$), 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:

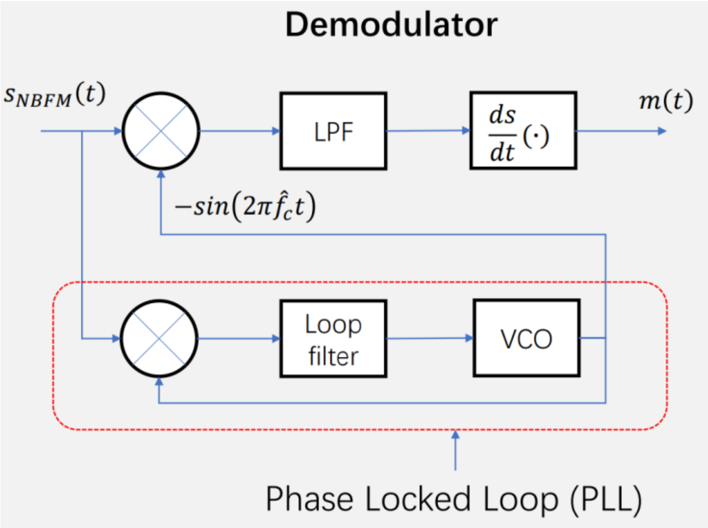
$$\cos(\beta \sin \omega_m t) \approx 1, \quad \sin(\beta \sin \omega_m t) \approx \beta \sin \omega_m t$$

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

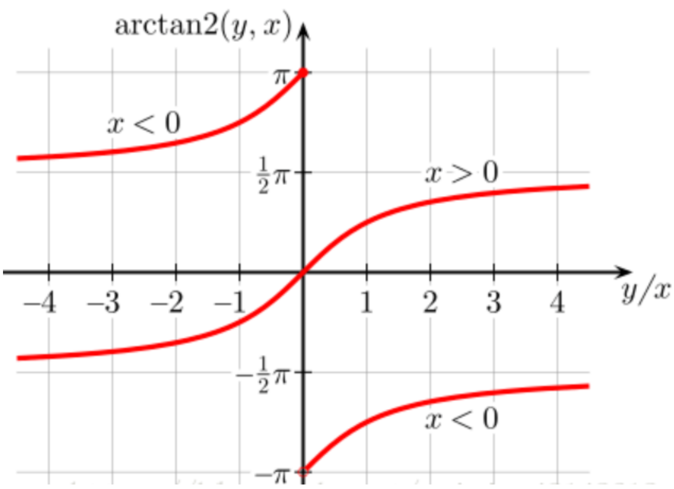
For NBFM the demodulation is easy to implement, it only needs to multiply it by $-\sin(2\pi f_c t)$, then m(t) can be obtained after a LPF. The flow chart is shown as follows where the Phase Locked Loop (PLL) is used to obtain the frequency of the carrier signal which will be introduced on latter.



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

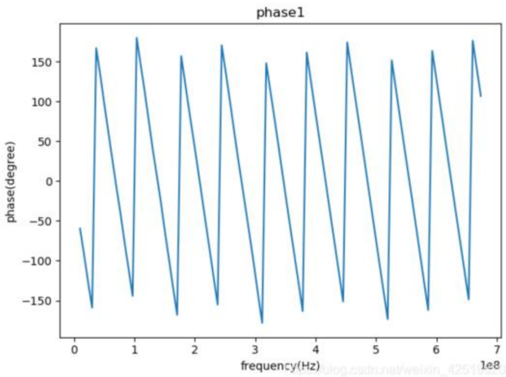
arctan(x) is used to calculate the angle of x=tan(θ), but the range domain of common function arctan(x) is from $-\pi/2$ to $\pi/2$, so it only deals with quadrants one and four. So in this experiment, we will use atan2(y, x) whose range domain is from $-\pi$ to π, so that it can handle any angle in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

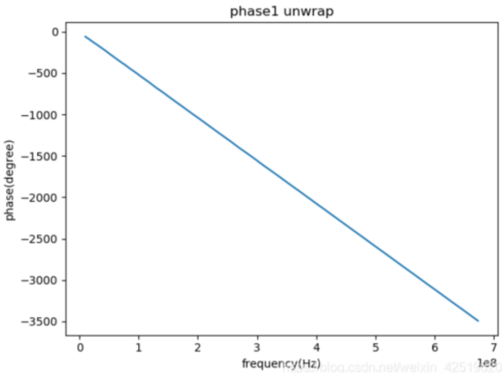


Its method of calculating is: when x>0, atan2(y/x) is namely equal to arctan(y/x), but when x<0 and y≥0, atan2(y/x) is equal to arctan(y/x) $+\pi$, when x<0 and y<0, atan2(y/x) is equal to arctan(y/x) $-\pi$.

(3)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:





(4)Principle of differentiation and integration

Baseband

$$s(nT_s) = cos[2\pi f_c t + 2\pi \int k_f m(nT_s)dt]$$

$$s_I(nT_s) = A_c cos(2\pi \int k_f m(nT_s)dt)$$

$$s_Q(nT_s) = A_c sin(2\pi \int k_f m(nT_s)dt)$$

$$s_l(nT_s) = s_I(nT_s) + js_Q(nT_s)$$

Integration:

Because the frequency is modulated, since the angle is equal to the integral of the frequency multiplied by 2pi, when the modulation signal modulate

$$2\pi \int k_f m(nT_s)dt = atan\left(\frac{s_Q(nT_s)}{s_I(nT_s)}\right)$$

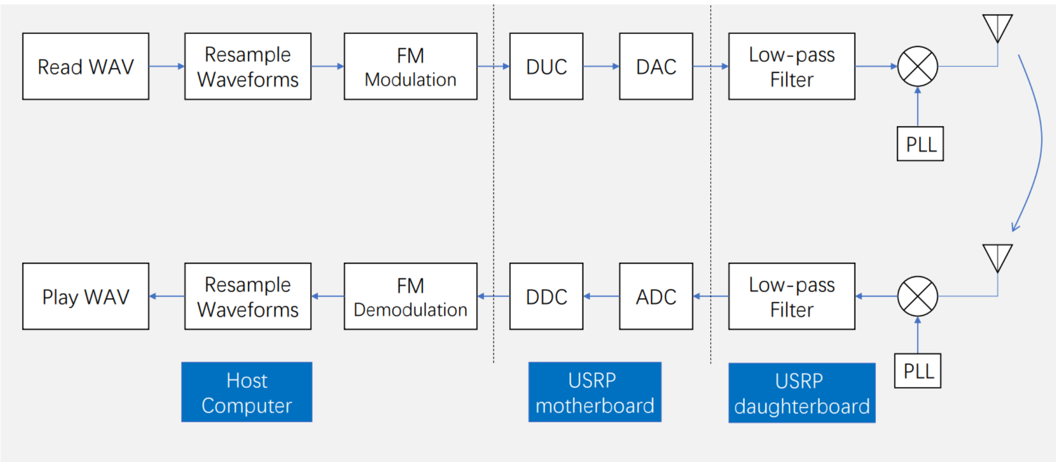
$$m(nT_s) = \frac{1}{2\pi k_f}\frac{d}{dt}\left[atan\left(\frac{s_Q(nT_s)}{s_I(nT_s)}\right)\right]$$

s the carrier signal, it needs to be integrated and added to the phase of the carrier signal.

Differentiation:

Since the transmission signal uses I signal and Q signal, the polar coordinate transformation is used at the signal receiving end RX to extract the angle information of the complex number, and the original modulated signal can be obtained by direct derivation.

(5)Waveform resampling

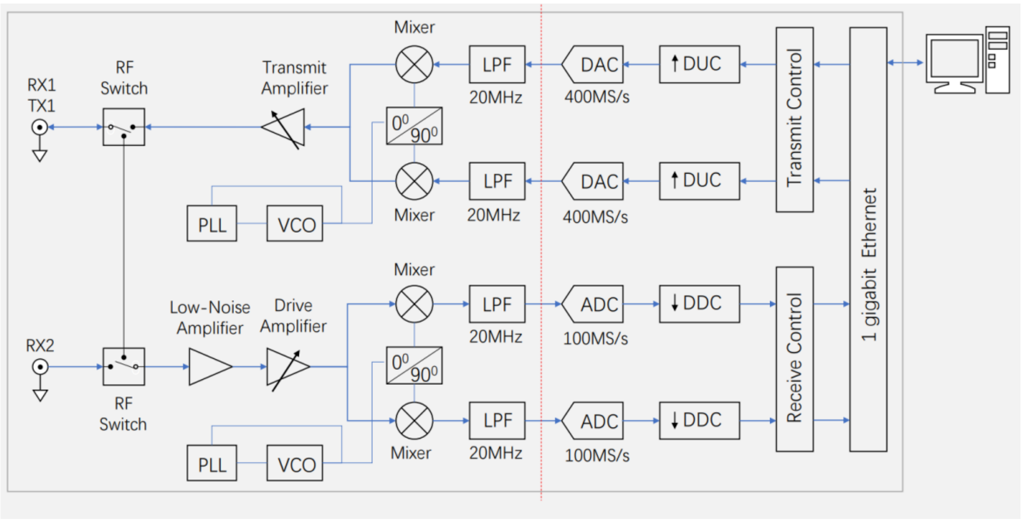


When entering a WAV file, because the frequency of the voice signal is generally 44100Hz, it is necessary to do upsampling so that the sampling rate of the signal matches IQ rate 200kHz, which is convenient for DAC

After the FM signal is demodulated, resampling is required to match the sampling rate of the player, and the frequency is reduced to 44100Hz for unified processing.

2.USRP Basic principle

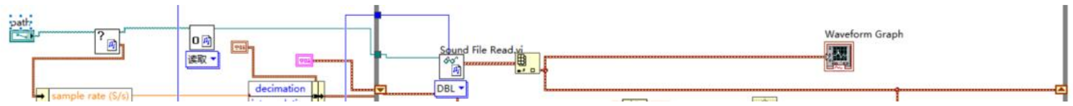
(Universal Software Radio Peripheral)



USRP adopts a typical software-defined radio structure, in the receiving link, the USRP daughter board first performs high-frequency amplification, mixing, and low-pass filtering of the RF signal received by the antenna, and then sends the processed signal (intermediate frequency signal) to the A/D and d/A converters in the motherboard for further processing. The USRP motherboard uses an analog-to-digital converter (ADC) to sample the IF signal, uses the FPGA to perform digital signal processing such as anti-aliasing filtering, downconversion (DDC), and decimation of the sampled digital signal, and finally transmits the extracted I/Q sampling signal to the computer through the Gigabit Ethernet interface. The transmitting process is the reverse of the receiving process.

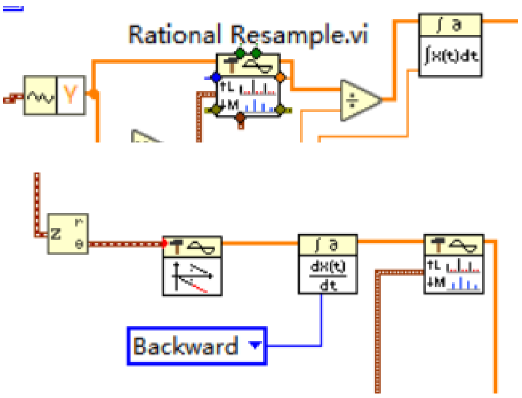
3.LabVIEW module

(1) Sound File Read



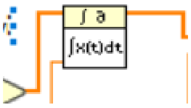
Extract the sound signal from the path and obtain the sampling rate, pass it to Sound File Read, form an array of information, and then do the establishment waveform processing, so that the sound signal forms a processable waveform data

(2) Rational Resample



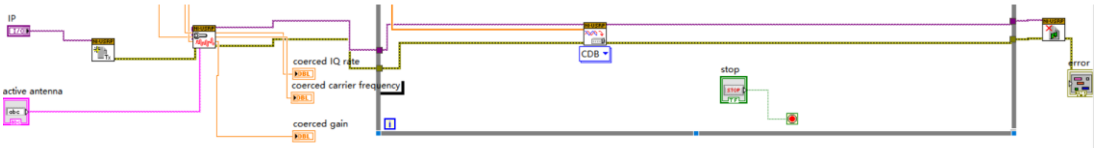
The rational resample module needs bus input data to decide whether to decimation or interpolation, first interpolation to make upsampling when entering voice signals, improve the sampling rate, decimation to make downsampling when restoring voice signals after demodulation, adjust the sampling rate to 44100Hz, adapt to the player frequency

(3) Integral x(t)

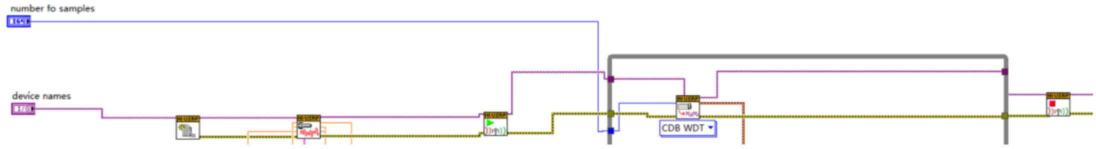


The modulated signal is integrated and embedded in the carrier signal to form an input signal

(4) niUSRP

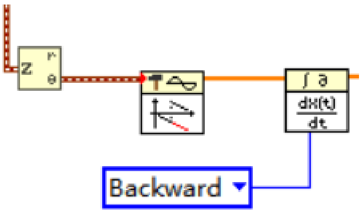


Using the labview module of the USRP that has been built, the DAC and analog signals can be transmitted given the relevant parameters



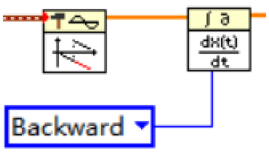
In the receiving section, the given module can form an array of analog signals in the form of I-channel signals and Q-channel signals into the host computer

(5) Unwrap



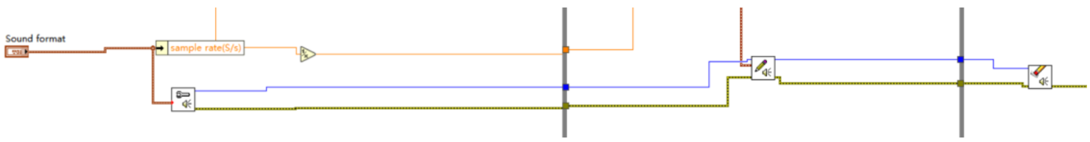
The I-way signal is orthogonal to the Q-way signal, and after the polar coordinate transformation at the receiving end and the angle is derived, the original signal from -pi to pi is obtained, and the phase is unfolded to smooth it

(6) Derivative x(t)



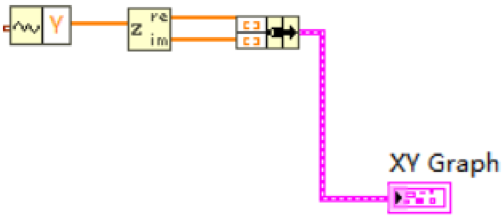
Using the differential block to derive the angle, the original modulated signal can be obtained

(7) Sound Output

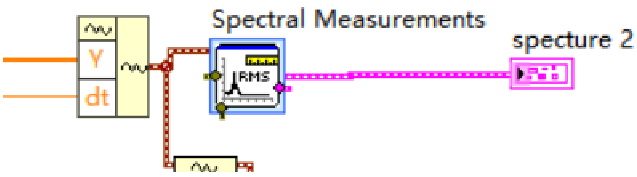


Use the sound playback module to set the sample rate of 44100Hz, and input the resampled sound signal into the module to play the sound

(8) Waveform display module, spectrum measurement module



The I signal and Q signal are decomposed as the real and imaginary parts of the transmission complex numbers, respectively, to make the XY constellation diagram

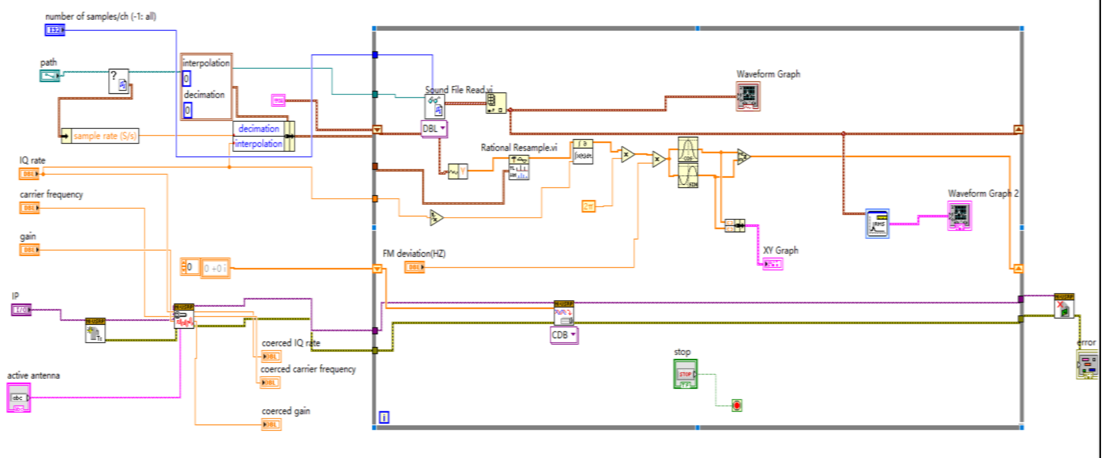


The waveform of the demodulated signal is reconstructed and the dt sampling interval is set to the reciprocal of 44100, and the dB value of the amplitude can be obtained through the waveform diagram module

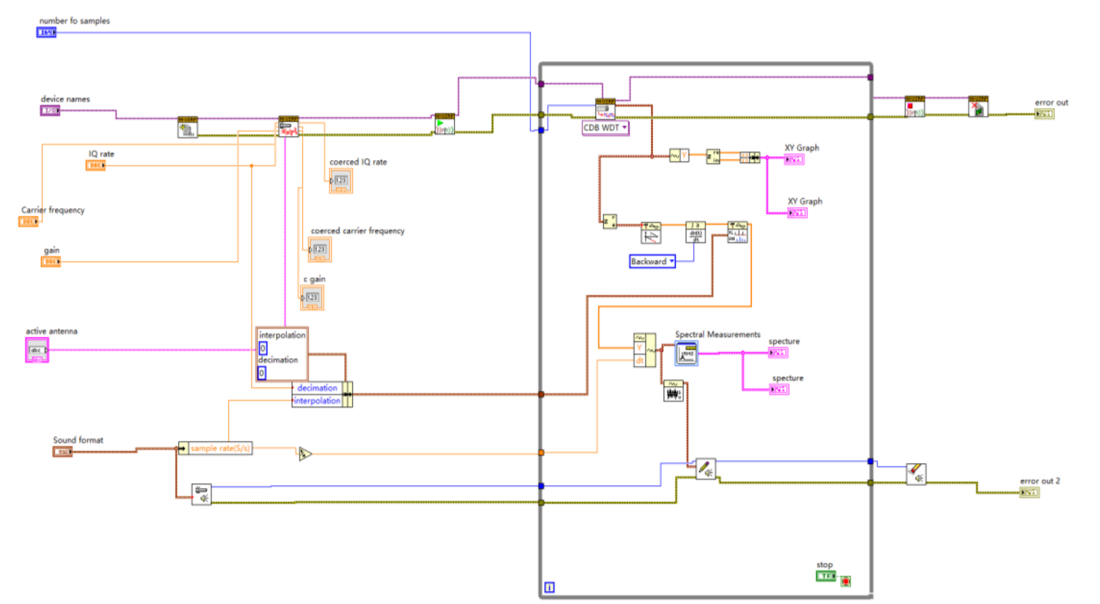
Lab results & Analysis:

Block diagram and result

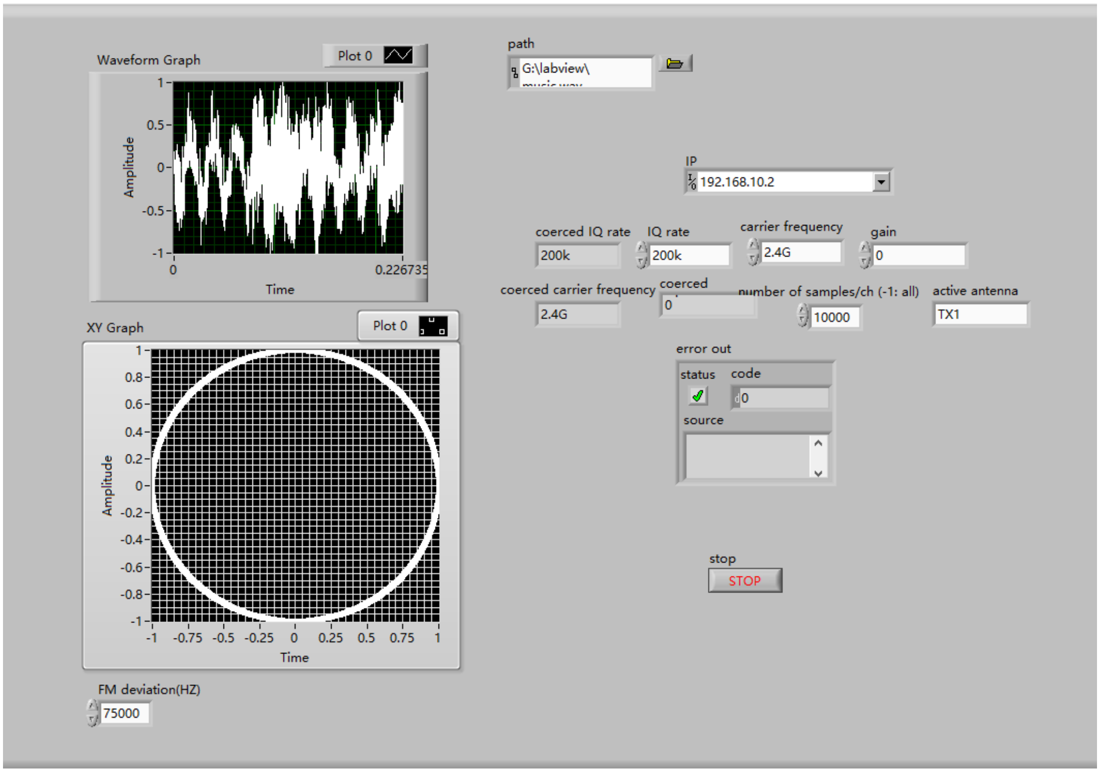
TX:

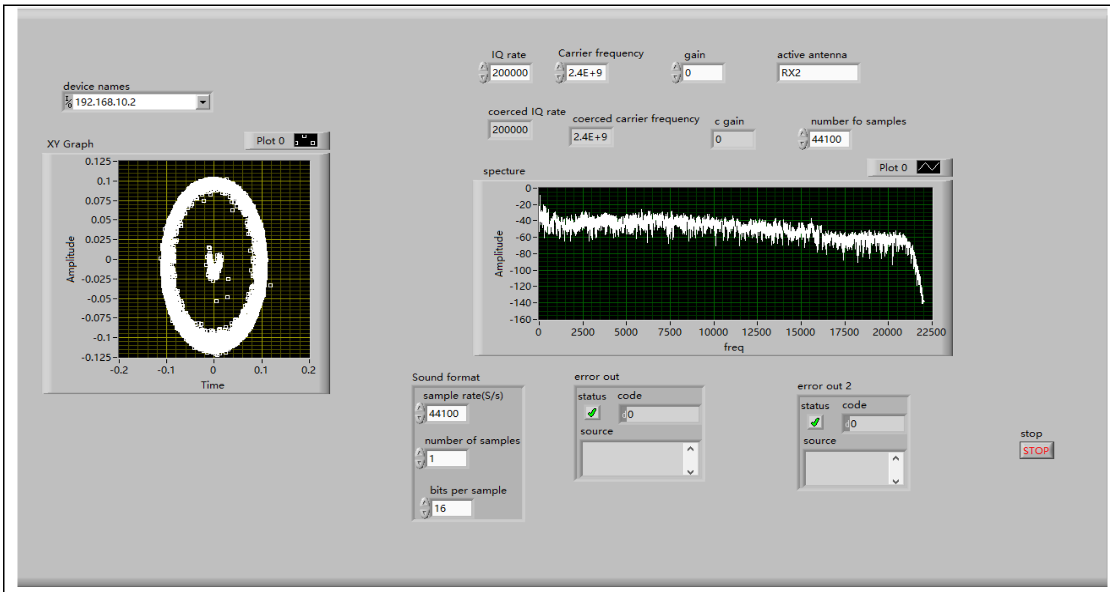


RX:



After choosing a music file to transmit, we can set the IQ rate to 200KHZ, because the frequency of music always is less than 20KHZ, so IQ rate is 200KHZ is enough in this experiment. Then the carrier frequency is set to 2.4GHZ. The result is as follows:

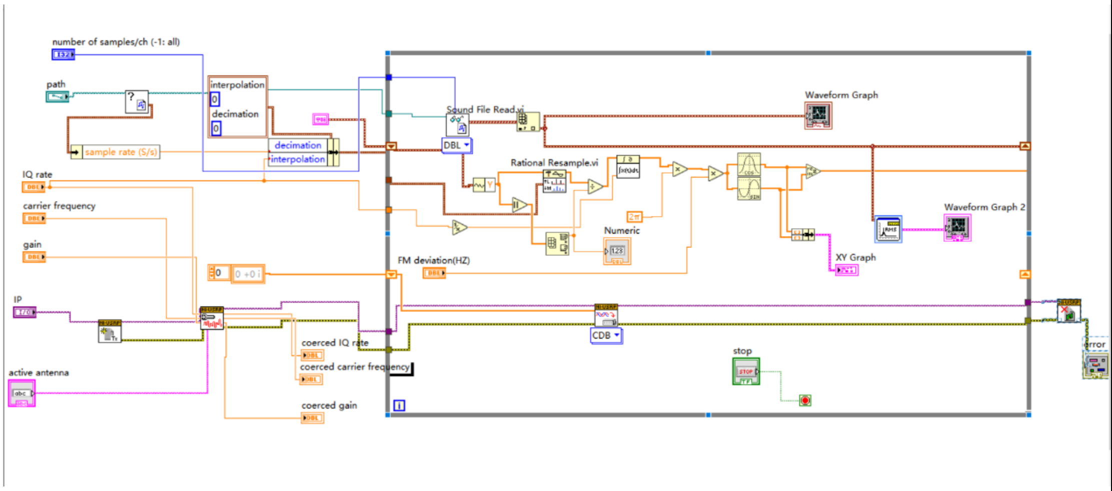




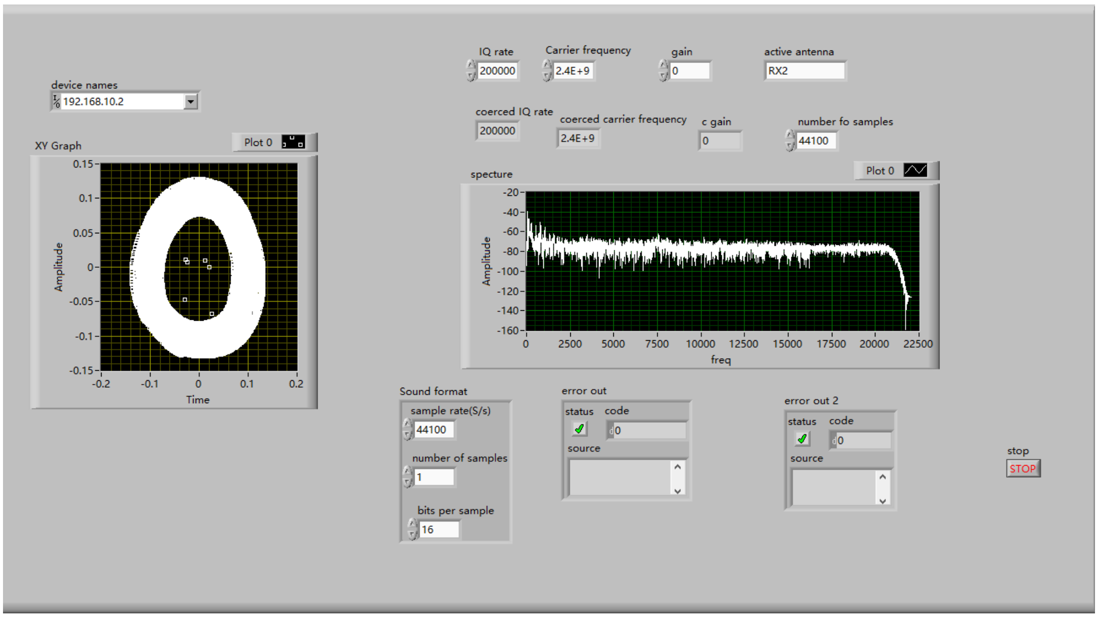
Analysis of factors affecting FM modulation system

The maximum frequency offset has an important impact on the modulation and demodulation of FM system, because the maximum frequency offset is related to the amplitude of message signal, which is difficult to measure and unify, so we can first do a normalization, that is, set the maximum value of message signal as 1, and then the coefficient is equal to the maximum frequency offset. Then the influence on the experimental effect was observed by adjusting the maximum frequency offset.

The program block diagram of normalization processing is shown as follows:

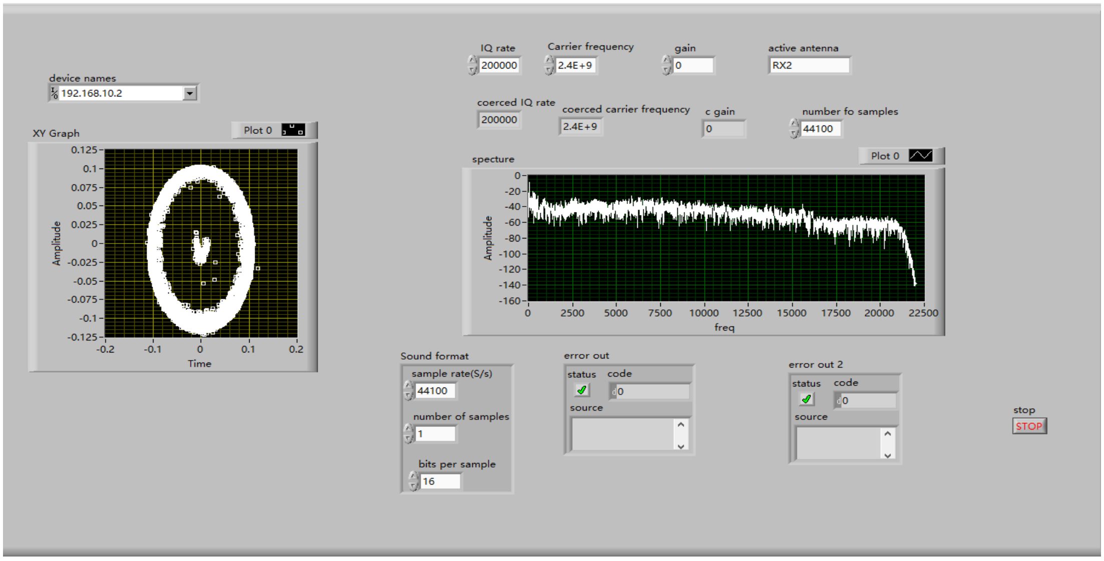


When the maximum frequency offset is 2KHZ, the result is as follows:

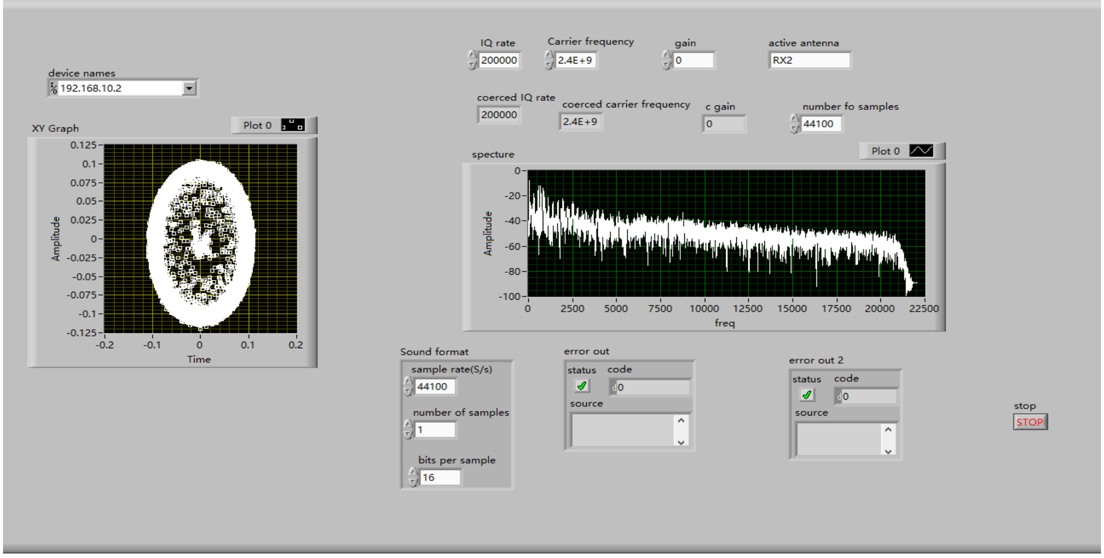


As can be seen from the figure, the effect of demodulation is not good, unable to recover the original signal correctly and completely.

Continuously increasing the value of the maximum frequency offset will improve the effect. When the maximum frequency offset is about 75KHZ, the effect is best. The results are as follows:



When the maximum frequency offset further increases, the noise will be found to increase. When the maximum frequency offset is 100KHZ, the results are as follows:



Bandwidth analysis

In experiment we often use Carson's rule to estimate the bandwidth of the modulated signal, according to the Carson's rule $BW=2\Delta f + 2f_m$, f_m is the frequency of the message signal, the frequency of general music signal is less than 20KHZ, inspecting the frequency spectrum of the message signal, we can find that the bandwidth of message signal is about 20KHZ, beyond which the signal will decay rapidly. Δf is the most frequency deviation, in order to measure Δf, we also firstly do the normalized processing to make the maximum of message signal to 1. Then Δf is the kf we set in panel. So Δf is equal to 75KHZ, then the bandwidth is equal to about 190KHZ.

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.

Extension section

Low-cost implementation: FM modulation/demodulation using FIR module

In this experiment, differentiation is used in the process of demodulation to differentiate the angle function obtained in the process of demodulation to obtain m(t). In this experiment, because it is a digital system, all signals are discrete, so we can use differential operation to replace the derivative operation, which will reduce the complex operation cost in the process of derivation. Thus, low-cost FM demodulation can be achieved. The conversion process is as follows:

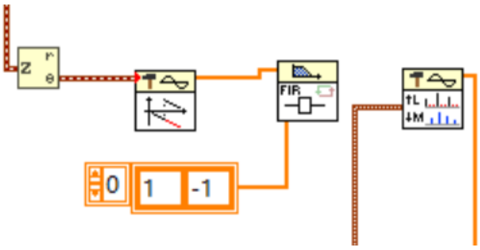
$$\frac{dx(t)}{dt} \xRightarrow[discretization]{} \frac{x[n]-x[n-1]}{1}$$

This suggests that we can use FIR filter to solve this problem, first let's analyze the principle of FIR filter.
The relation between output and input of FIR filter is a form of discrete convolution sum, whose expression is given by LabVIEW as follows:

$$y_i = \sum_{j=0}^{N_b-1} b_j x_{i-j},$$

Where b_j is the coefficient of FIR filter.
As can be seen from the above equation, if we set the coefficients of FIR filters to 1 and -1 respectively, the relationship between output and input will be the same as the above equation, which means that we can use FIR filters to replace the differential module used in the previous program. The modified program block diagram is

shown as the figure below (only the modified derivative part of the program is presented).



In the traditional demodulation program, we will first obtain the Angle function through the inverse tangent method, and then carry out the derivative operation. If the two steps are combined, through the theoretical formula derivation, we can get the following derivation process:

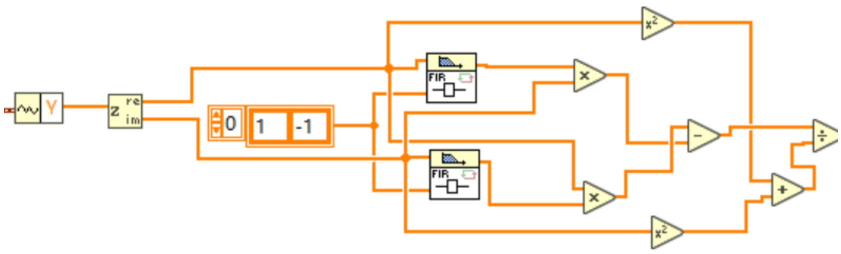
$$m(t) = \frac{d[\arctan(\frac{Q(t)}{I(t)})]}{dt} = \frac{I(t)\frac{dQ(t)}{dt} - Q(t)\frac{dI(t)}{dt}}{I^2(t)+Q^2(t)}$$

That is using discrete difference method:

$$m(n) = \frac{I(n)[Q(n)-Q(n-1)]-Q(n)[I(n)-I(n-1)]}{I^2(n)+Q^2(n)}$$

$$= \frac{I(n-1)Q(n)-I(n)Q(n-1)}{I^2(n)+Q^2(n)}$$

The expression finally obtained tells us that this expression can be realized through FIR filter which means that the arctangent module and the derivative module can be replaced by a FIR filter module, which will greatly reduce the cost. The above expression can be realized by using two FIR filter modules. The block diagram of this part is as follows:

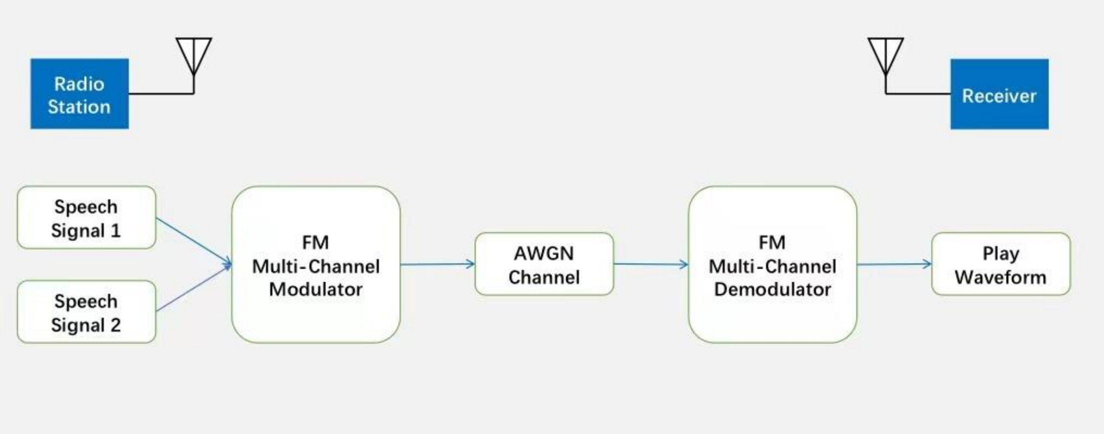


Similarly, we can also use IIR filter to replace the integration module, so as to reduce the complex operation cost of integration operation.

Attempt of FM multi-channel design

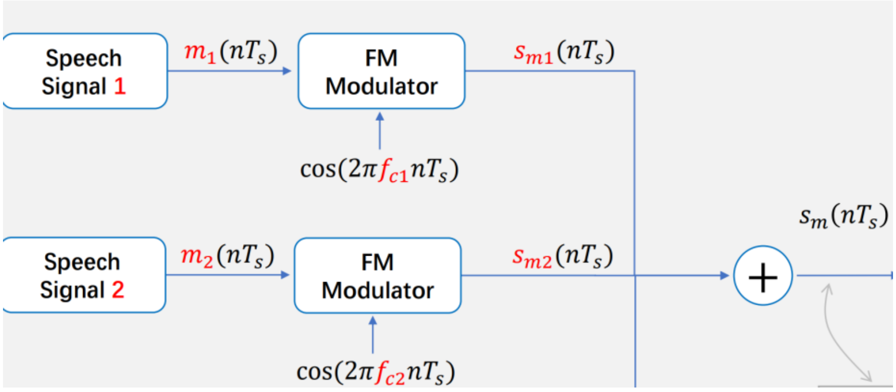
Theoretical analysis

FM multi-channel is to transmit many signals at the same time from the transmitter, and then the receiver will demodulate these signals separately to obtain the only one signal receiver wants. The whole process is presented as follows:

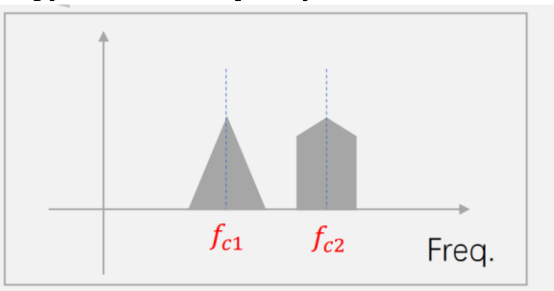


The fundamental principle is using FDM (Frequency Division Multiplexing) to transmit many signals making different signals be transferred to different frequency bands in the frequency domain without overlapping each other by multiplying different carrier signal with different carrier frequency, so that different signals can be transmitted without mutual interference in the frequency domain. At the receiver, it is only need to use a bandpass filter or move different signals back to the low frequency after moving corresponding carrier frequencies. After passing a low-pass filter, in this way, the desired signal can be obtained at the receiver according to different frequency bands.

The modulation process is as follows:



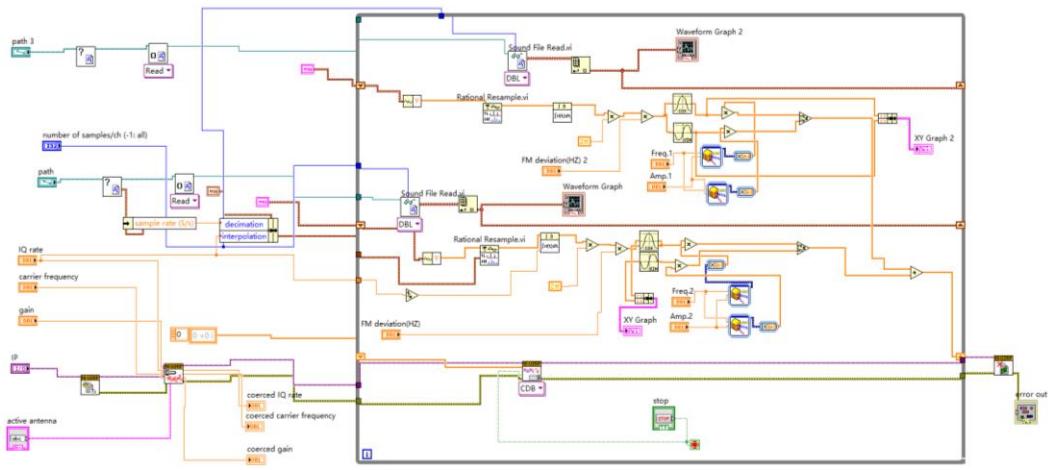
So that the signals occupy different frequency band:



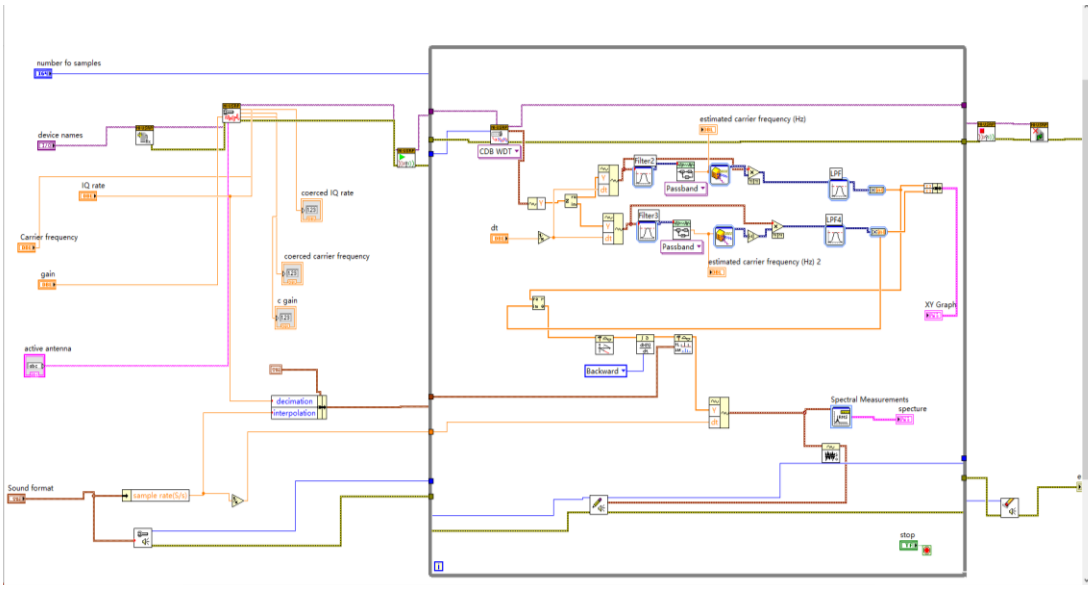
In this experiment, we take the transmission of two signals as an example. First, the program block diagrams are as follows:

Block diagram and result

TX:



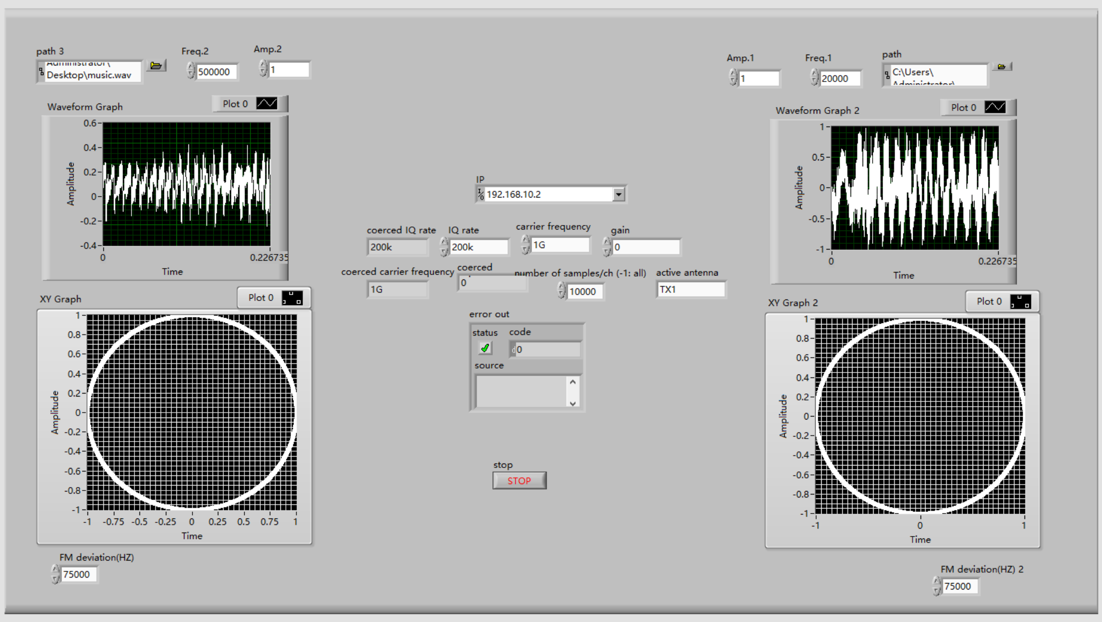
RX:

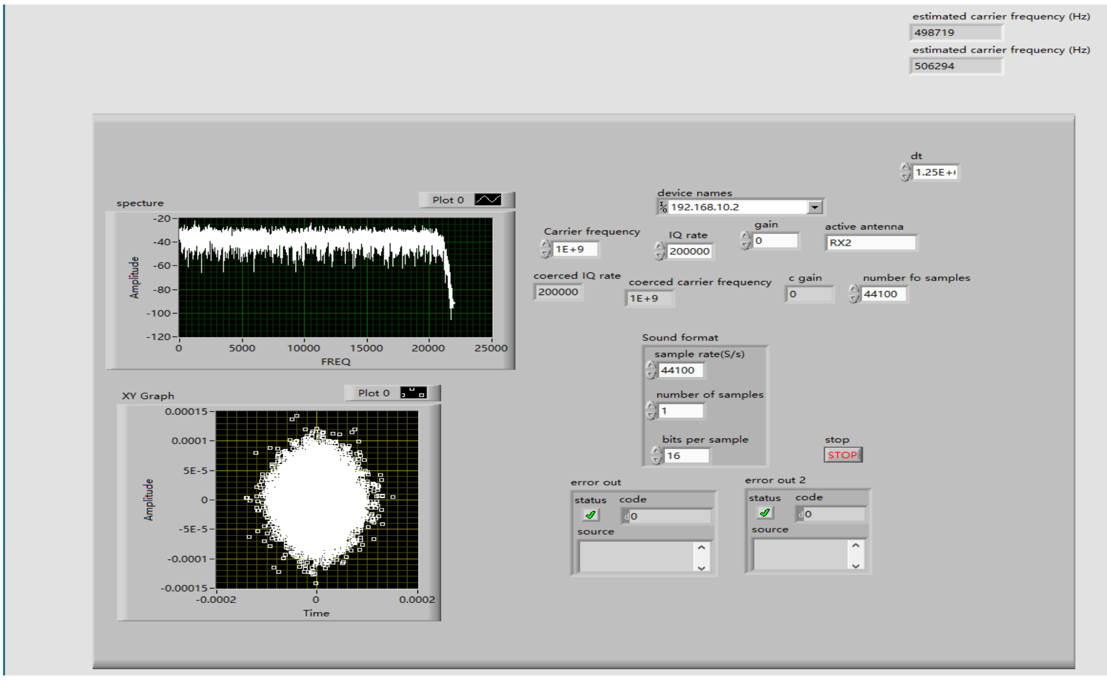


In TX, we take the cosine of the message signal times the carrier signal of the cosine, and we make that the real part, and then we take the sine of the message signal times the carrier signal of the sine, and we make that the imaginary part, and we set the carrier frequencies of the different signals to different values so that we can separate the two signals in the frequency domain. Finally, will be modulated two signals transmitted out, that is, to complete the design of TX.

In RX, we firstly convert the signal data into waveform data, then use complex module to obtain the real and imaginary components, then resample the real and imaginary data to reconstruct the waveform, pass the reconstructed waveform signal through a bandpass filter, then use the phase-locked loop to obtain the carrier frequency, and then multiply the corresponding carrier to transfer the required signal to the low-frequency position. After that, the signal can be obtained by using suitable low-pass filter to complete the demodulation of multi-channel FM system.

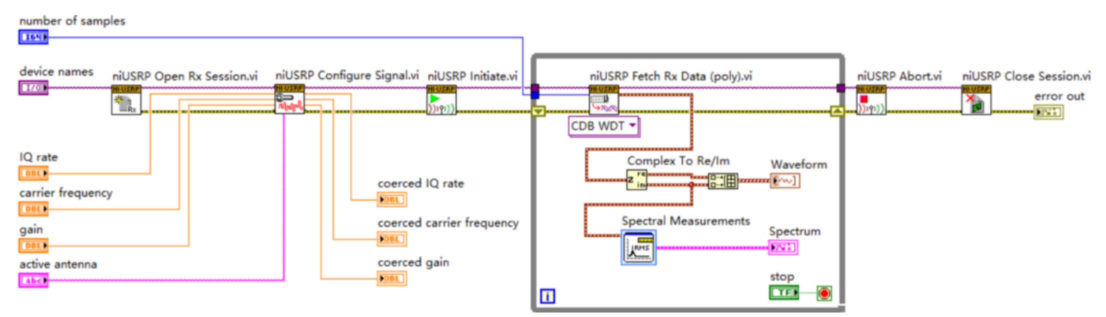
After adjusting the parameters on the panel, the simulation results can be obtained. It is worth noting that we need to set reasonable maximum frequency offset and reasonable parameters of band-pass filter and low-pass filter. Unfortunately, we did not get perfect demodulation and recovery in the end, and the effect is shown in the figure below.

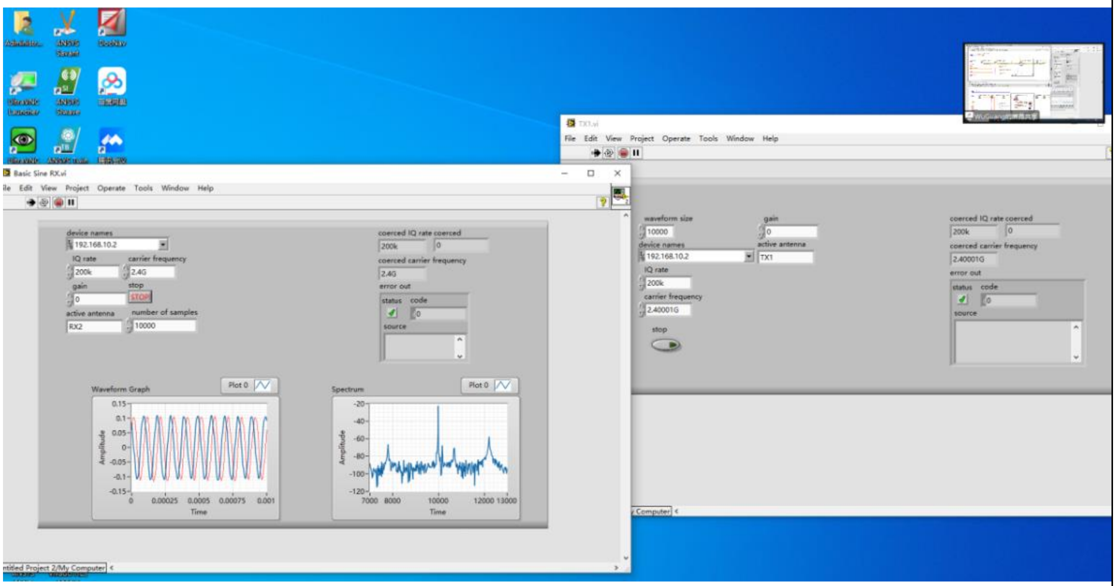




Experience

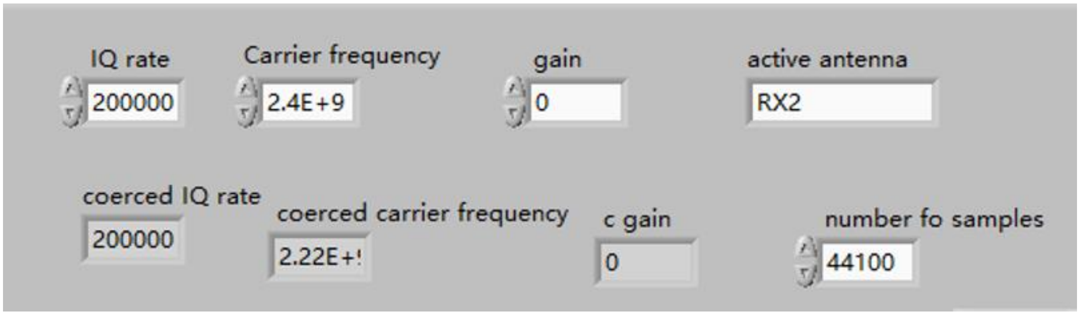
1. The screenshot of in class experiment





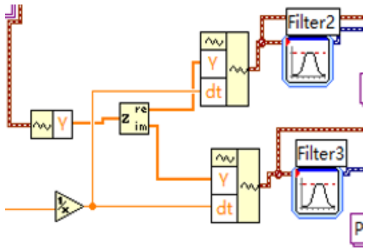
2. Problem we meet

1. First of all, we found in the laboratory that the carrier frequency of 2.4GHZ was not good for experiments, and sometimes the actual carrier frequency was smaller than 2.4HZ. We guessed that this might be because there were too many people operating in the laboratory, and the use of 2.4HZ would cause interference to the signals sent and received in each other's experiments. And the large number of people using the instrument at the same time may make the actual power low and not produce the desired 2.4HZ carrier frequency.

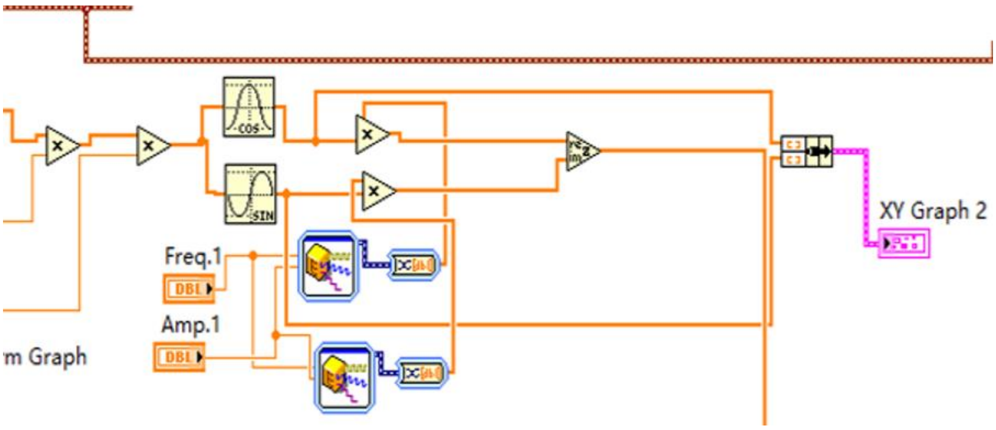


2. During demodulation, if we directly pass the signal through the acquired waveform component module and then through the bandpass filter, the system will report an error, and the error information is that the maximum cut-off frequency of the bandpass filter is less than half of the sampling frequency, because the sampling rate of the directly acquired waveform signal is not appropriate. Therefore, we need

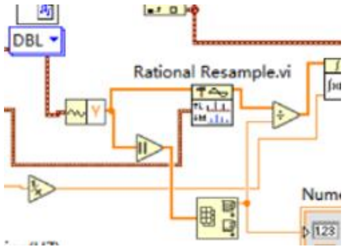
to use the reconstructed waveform module and use the appropriate sampling rate for sampling. The modules taken are shown in the figure below:



3. Before, we directly converted the obtained complex signal into cos carrier in the demodulation program design of FM multi-channel design. Later, we found that there might be some problems in doing so. We traced back to the original formula of FM modulation signal and found that the correct approach should be to modulate the real and imaginary signals by multiplying the cos carrier and sin carrier respectively. So we ended up changing the program to the final version.



4. When we estimated the bandwidth of the modulated signal during debugging, we found that there would be a large error in the estimation. Later, we studied the expression of the bandwidth of the FM modulated signal and found that the maximum frequency offset was the coefficient kf multiplied by the maximum value of message signal, so the real maximum frequency offset was not the parameter we modulated on the panel. However, it is difficult to know the maximum size of the message signal, so we use a normalization operation to solve this problem. We set the maximum size of the message signal to 1, so the coefficient on the panel is directly equal to the maximum frequency offset, so we can directly know the bandwidth.



3.Contribution

We two finish the whole LabVIEW program together. Zhang Haodong finish the Analysis of Factors Affecting FM Modulation System and a part of extension section. The introduction and analysis of the advantages and disadvantages of FM modulation system were completed, and its social application value was elaborated by Song Yihang. **Contribution ratio :50%,50%**	
Score	97