Software Defined Radio Voice Transmission

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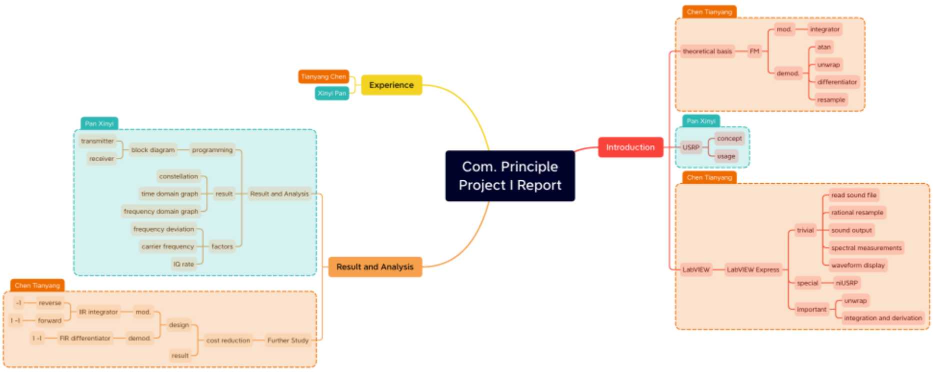
Outline



Introduction

FM Communication Systems

Modulation Process and Utilities

The core concept of FM is to transmit signals by modulating the frequency deviation on a high-frequency carrier signal. The specific method is as follows:

$$f = f_c + k_f m(t)$$

$$\omega = 2\pi f = 2\pi (f_c + k_f m(t))$$

$$\phi = \int \omega d\tau = 2\pi f_c t + 2\pi k_f \int m(\tau) d\tau$$

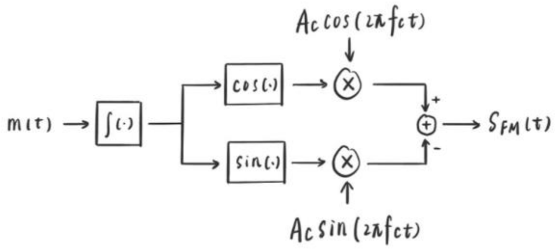
$$f_{FM} = A_c \cdot \cos(2\pi f_c t + 2\pi k_f \int m(\tau) d\tau)$$

1. The frequency of the modulated signal varies linearly with the amplitude of the message;
2. The modulated frequency f is converted into angular frequency ω;
3. The time-domain integration of the angular frequency ω results in the phase Φ;
4. The phase Φ is then incorporated back into the functional expression of the carrier, which produces the modulated signal in the time domain.

Using the properties of trigonometric functions, we can expand the above function into the following series:

$$S_{FM}(t) = A_c \cdot \cos(2\pi k_f \int m(\tau) \cdot d\tau) \cos(2\pi f_c t) - A_c \sin(2\pi k_f \int m(\tau) d\tau) \sin(2\pi f_c t)$$

Based on this expansion series, we can design a modulation scheme:



1. Perform time-domain integration;
2. Take the sine and cosine of the resulting signal and multiply each with the corresponding carrier signal;
3. Add the two resulting signals together to obtain the modulated signal.

In the modulation system, we utilized an integrator:

Demodulation Process and Utilities

In this experiment, the demodulation method differs from the frequency discrimination procrss. As we all know that for angle modulation, the signal is loaded onto the phase of the modulated signal.

Therefore, we use the following steps to recover the original message signal:

$$S_{FM}(t) = A_c \cdot \cos(2\pi k_f \cdot \int m(\tau) \cdot d\tau) \cos(2\pi f_c t) - A_c \sin(2\pi k_f \cdot \int m(\tau) d\tau) \sin(2\pi f_c t)$$

$$(S_{FM}(t) \cdot \cos(2\pi f_c t))_{LPF} = \frac{1}{2} A_c \cos(2\pi k_f \int m(\tau) d\tau)$$
$$(-S_{FM}(t) \cdot \sin(2\pi f_c t))_{LPF} = \frac{1}{2} A_c \cdot \sin(2\pi k_f \int m(\tau) d\tau)$$

Arctangent: Phasor Recovery

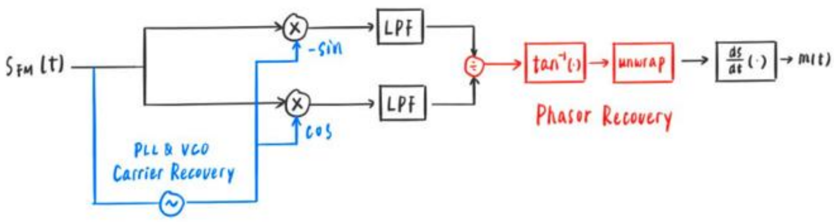
$$\frac{\frac{1}{2} A_c \sin(2\pi k_f \int m(\tau) d\tau)}{\frac{1}{2} A_c \cos(2\pi k_f \int m(\tau) d\tau)} = \tan(2\pi k_f \cdot \int m(\tau) d\tau)$$

$$\xrightarrow[\text{unwrap}(\cdot):\ \to (-\pi, \pi)]{\arctan(\cdot)} 2\pi k_f \int m(\tau) d\tau$$

$$\xrightarrow{\frac{d}{dt}(\cdot)} 2\pi k_f\, m(t)$$

1. We use the arctangent method to directly obtain the phase of the signal.
2. After phase unwrapping, we differentiate with respect to time.

In this way, we can recover the original message signal. The flowchart of this process is shown below.

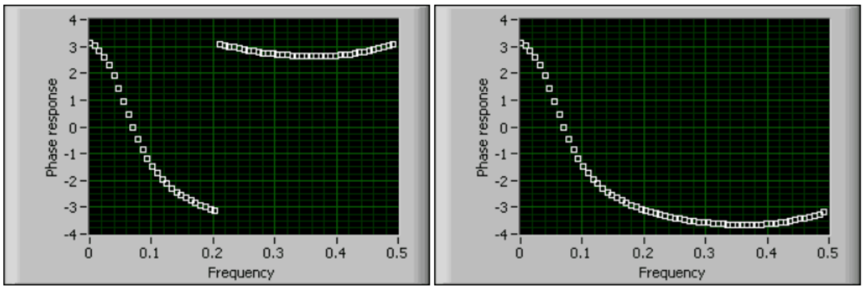


The arctangent method used in this experiment differs slightly from the one used in the previous experiment. Based on our knowledge of complex variable functions, we treat the signal directly as a complex number, and the phase angle of the complex number is what we need to obtain.

In the modulation system, we utilized complex to polar function, unwrap phase function, a differentiator and rational resample function.

For complex to polar functions: the function converts rectangular coordinates of a complex number z = a + bi to its polar components r and θ, where r is the magnitude and θ is the phase angle. The magnitude r is computed as the square root of the sum of the squares of the real and imaginary parts of z. The phase angle θ is computed as the arctangent of the imaginary part divided by the real part of z, using the arctan2 function to correctly handle the quadrant in which the angle lies. When matrix data is wired as input, this function is replaced by a VI that works with matrix data type, and the resulting VI has the same icon but uses a matrix-specific algorithm. If the input data type causes a basic math operation to fail, such as division by zero, the function returns an empty matrix or NaN. Disconnecting the matrix input(s) will restore the original function.

For unwrap phase function: The Unwrap Phase VI is a function that eliminates discontinuities in a Phase array by unwrapping it. This is achieved by using an equation to calculate the unwrapped phase whenever the difference between two adjacent values in Phase exceeds pi. The phase unit is specified as radians in and radians out. For other units specified in phase unit, similar equations are used to calculate the unwrapped phase. The purpose of this VI is to correct phase discontinuities that occur in signals due to phase wrapping. When phase values wrap, it becomes difficult to interpret the signal correctly. Unwrapping the phase makes it easier to analyze and interpret the signal accurately.



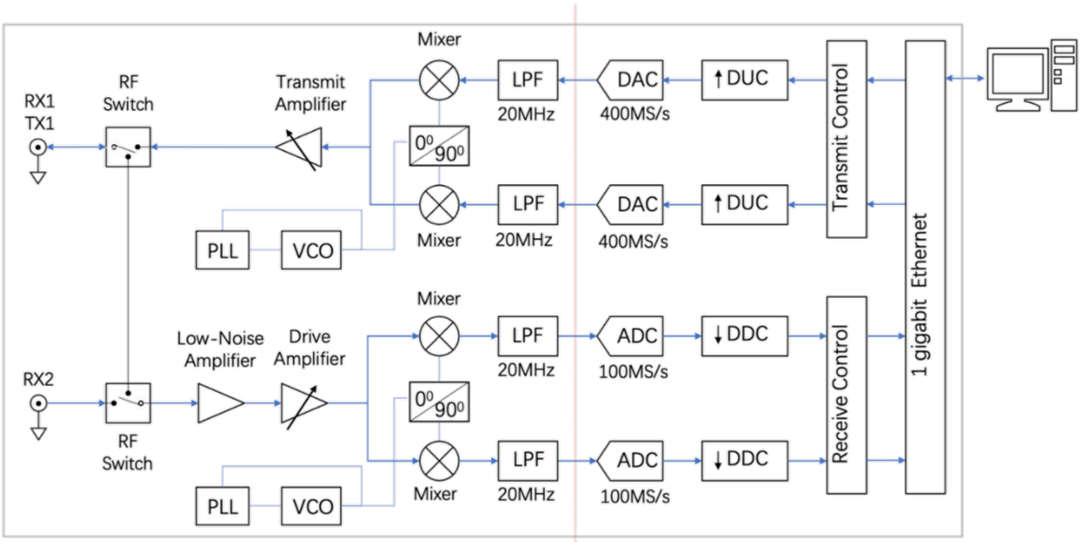
Two graphs above are presented to demonstrate the effects of unwrapping the phase. The first graph shows the original phase before unwrapping, while the second graph shows the phase after unwrapping. The unwrapped phase graph shows a continuous curve that is easier to analyze and interpret compared to the wrapped phase graph, which shows a discontinuous curve due to phase wrapping.

For differentiators: in signal processing, a differentiator is a device that approximates the derivative of a continuous-time signal by computing the difference between two adjacent samples. The differentiator is commonly used in applications that require the detection of sharp transitions in signals or the measurement of signal rates. Differentiation can be implemented using various methods such as the 2nd Order Central, 4th Order Central, Forward, and Backward methods, which use initial and final conditions to minimize the error at the boundaries.

For rational resample function: the function is used for resampling an input signal X. The function resamples the signal by changing the time interval between adjacent samples. The process involves inserting zeros between every two adjacent samples of the input signal to increase the number of samples by an interpolation factor, producing an interpolated signal X1. A Finite Impulse Response (FIR) filter is applied to the interpolated signal starting from the start index, producing the first output sample. The filter is then moved to the position start index + decimation, and filtering is repeated to obtain the second output sample. This process continues with each new decimation value being added until there are not enough samples in X1 for filtering. The final output samples are stored in internal states, and the function waits for the next signal block. The time interval between two adjacent samples in the output signal is equal to decimation/interpolation, and t0 specifies the time of the first output sample. The function uses a combination of up sampling, filtering, and down sampling techniques to resample the input signal to a desired rate while minimizing aliasing and phase distortion.

NI USRP

NI USRP (National Instruments Universal Software Radio Peripheral) is a software-defined radio (SDR) platform that enables researchers, engineers, and educators to rapidly prototype and implement wireless communication systems. It offers a versatile and reconfigurable architecture, making it suitable for a wide range of applications, including signal processing, spectrum analysis, and real-time communication. With support for various RF frequency ranges and a flexible interface to software such as LabVIEW, the NI USRP provides a comprehensive and powerful solution for SDR development.



LabVIEW Express

Special Modules

The following modules are special for this experiment:

1. The **niUSRP modules** in LabVIEW Express provide driver functions for controlling the Universal Software Radio Peripheral (USRP), which is a flexible software-defined radio platform for wireless communications and signal processing applications.

Significant Modules

The following modules play important roles in this experiment:

1. The **Integral module** in LabVIEW Express performs time-domain integration of an input signal x(t) over a specified time interval and generates the corresponding output signal;

2. The **derivative module** in LabVIEW Express performs time-domain differentiation of an input signal x(t) over a specified time interval and generates the corresponding output signal;

3. The **unwrap module** in LabVIEW Express is used to remove the phase jumps in a phase signal caused by periodic discontinuities and generate a continuous output phase signal.

4. The **rational resample module** is used to resample a waveform signal to a different sample rate while maintaining the original waveform shape.

Trivial Modules

The following modules are trivial and have been applied in the previous experiments:

1. The **sound file module** is used for reading audio waveform data from a file and storing it in a LabVIEW waveform format;

2. The **sound output module** is used for playing audio waveform data through a specified audio output device;

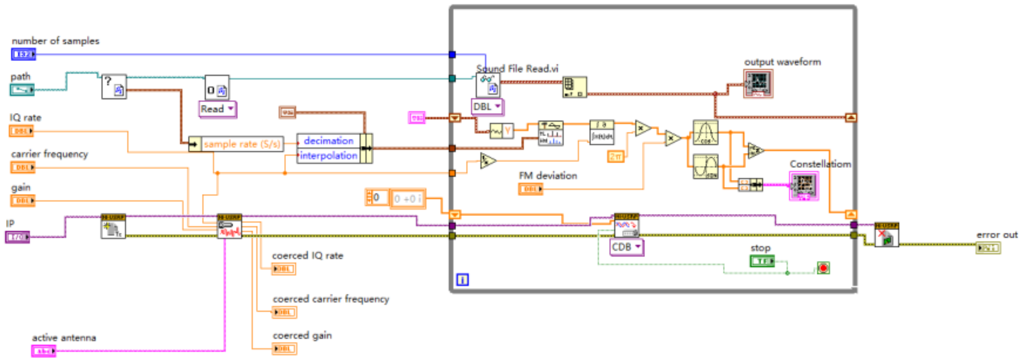
4. The **waveform display module** is used for displaying waveform data as a graph in a LabVIEW front panel;

5. The **spectral measurements module** is used for performing spectral analysis of a waveform signal to obtain information about its frequency content.

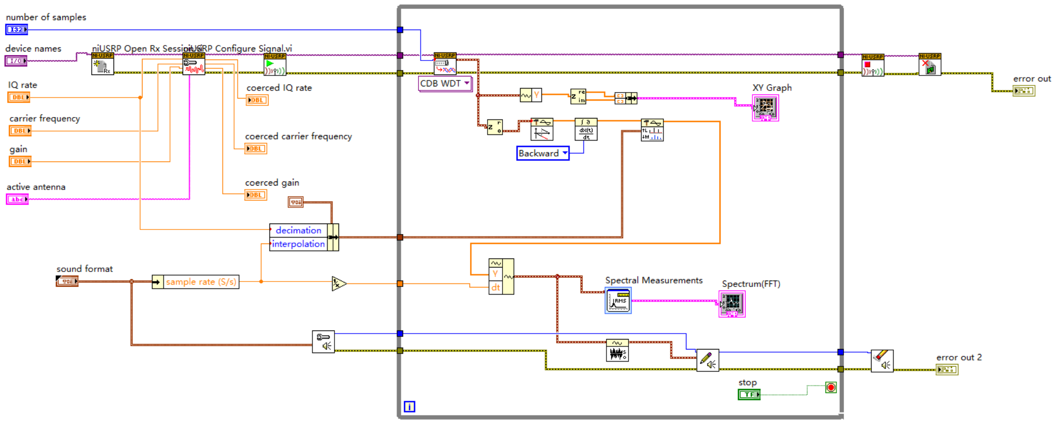
Result and Analysis

LabVIEW Programming

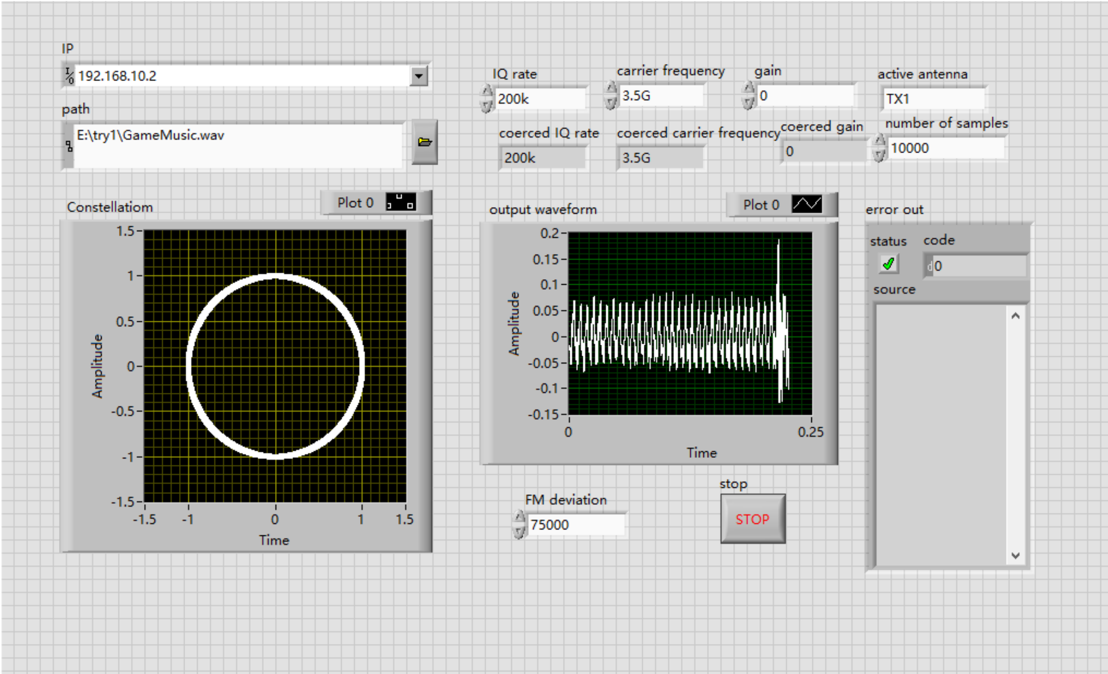
Modulation Program Block Diagram



Demodulation Program Block Diagram

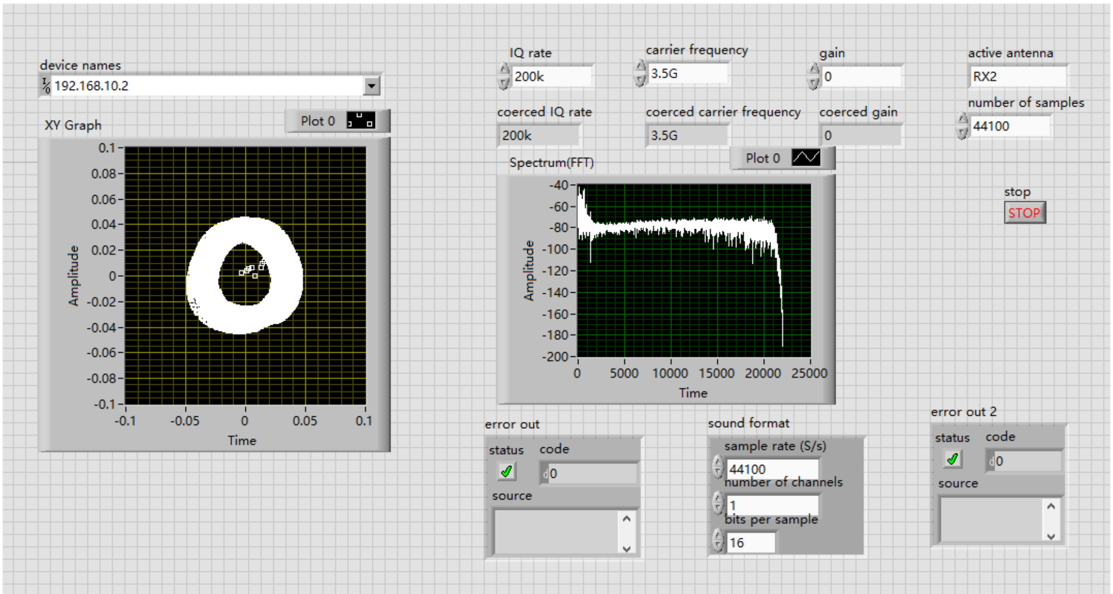


Simulation Result



Transmitter simulation result

carrier frequency=3.5G, FM deviation=75000Hz, IQ rate=200k



Receiver simulation result

carrier frequency=3.5G, FM deviation=75000Hz, IQ rate=200k

Influencing Factors of FM Modulation System

Carrier Frequency

The carrier frequency is the frequency of the unmodulated carrier signal used in FM modulation.

The frequency of the modulated FM signal is varied according to the information signal while remaining centered around the carrier frequency.

In practice, the choice of carrier frequency impacts the propagation characteristics of the FM signal, which in turn affects the coverage area and signal quality. Higher carrier frequencies tend to have a more limited coverage area and are more susceptible to obstacles such as buildings and terrain. On the other hand, lower carrier frequencies can propagate over longer distances and are less affected by obstacles.

The selection of carrier frequency is influenced by factors such as regulatory frequency allocations, intended coverage area, and the presence of other communication systems. The selection should balance the need for coverage, signal quality, and efficient use of the frequency spectrum.

In this experiment, we selected three different carrier frequencies of 2.4G, 3.5G and 10G. As shown in Figure 4.1.1, 2.4G is the carrier frequency provided by the teacher, so there are a large number of users, resulting in more noise in the received signal and the phenomenon of crosstalk frequency. As shown in Figure 4.1.2, 3.5G is used by fewer people, so the music received is very clear. As shown in Figure 4.1.3,10G is already outside the USRP carrier frequency range, so we actually have a carrier frequency of 4.42G, and the sound received is very clear.

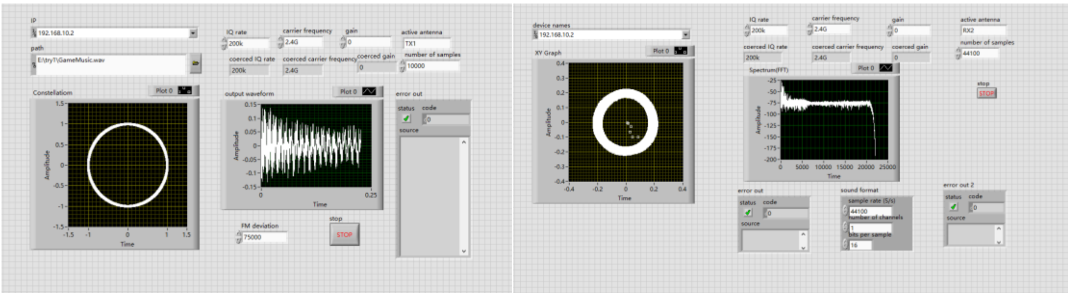


Figure 4.1.1 carrier frequency=2.4G, FM deviation=75000Hz, IQ rate=200k

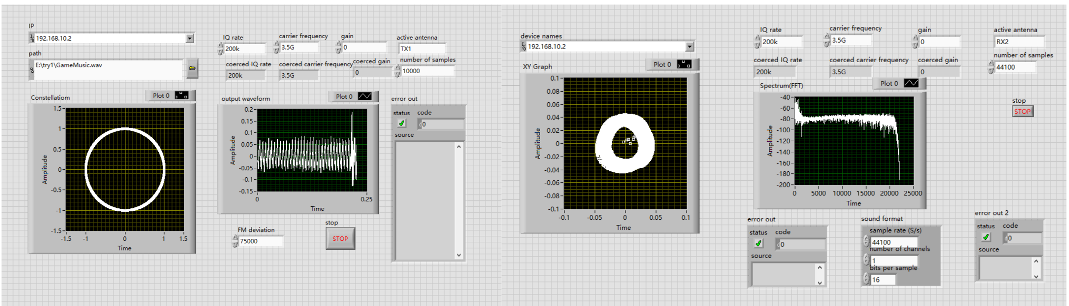


Figure 4.1.2 carrier frequency=3.5G, FM deviation=75000Hz, IQ rate=200k

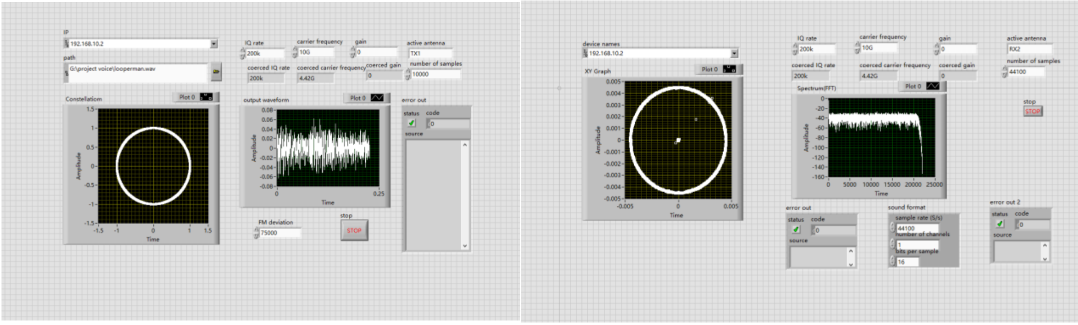


Figure 4.1.3 carrier frequency=10G, FM deviation=75000Hz, IQ rate=200k

Frequency Deviation

In frequency modulation (FM) systems, the maximum frequency deviation refers to the maximum change in frequency from the carrier frequency due to modulation by the information signal. This deviation is often denoted as Δ*f*. Carson's formula is a widely used equation for estimating the bandwidth of an FM signal. According to Carson's formula, the bandwidth of an FM signal is given by:

$$BW = 2(\beta + 1) f_m \quad = 2(\Delta f + f_m)$$

According to Carson's formula, a larger maximum frequency deviation will generate a wider signal bandwidth, thus improving the overall fidelity, but it may introduce more noise and occupy more spectrum resources, so it is necessary to reasonably plan spectrum allocation. A larger maximum frequency deviation will also result in a higher modulation index, possibly resulting in overmodulation.

At the same time, the low maximum frequency offset will produce a relatively narrow signal bandwidth, resulting in the overall fidelity is not high, high-frequency signal loss, limited transmission information. And the smaller modulation index will lead to under modulation, resulting in poor signal quality.

In this experiment, FM deviation= 750000,75000,5000 Hz was tried. When FM deviation= 75000Hz (Figure 4.2.1), the signal is relatively clear and complete. When FM deviation=750000Hz (Figure 4.2.2), there was too much noise and overmodulation. When FM deviation=5000Hz (Figure 4.2.3), the high frequency signal is obviously lost and the degree of signal restoration is not high.

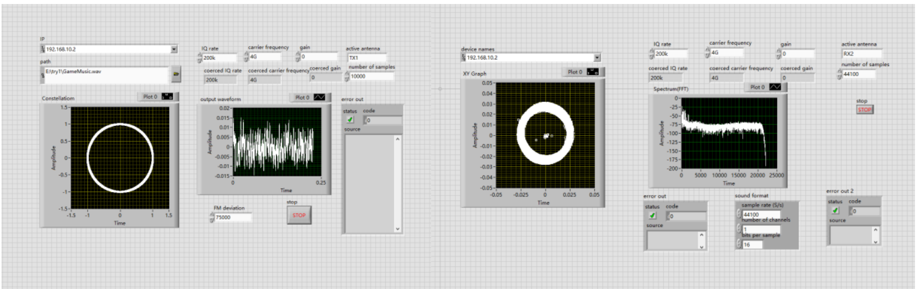


Figure 4.2.1 carrier frequency=4G, FM deviation=75000Hz, IQ rate=200k

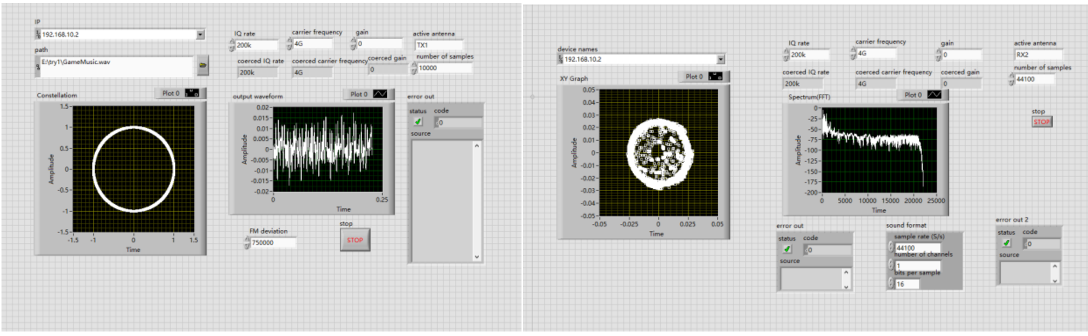


Figure 4.2.2 carrier frequency=4G, FM deviation=750000Hz, IQ rate=200k

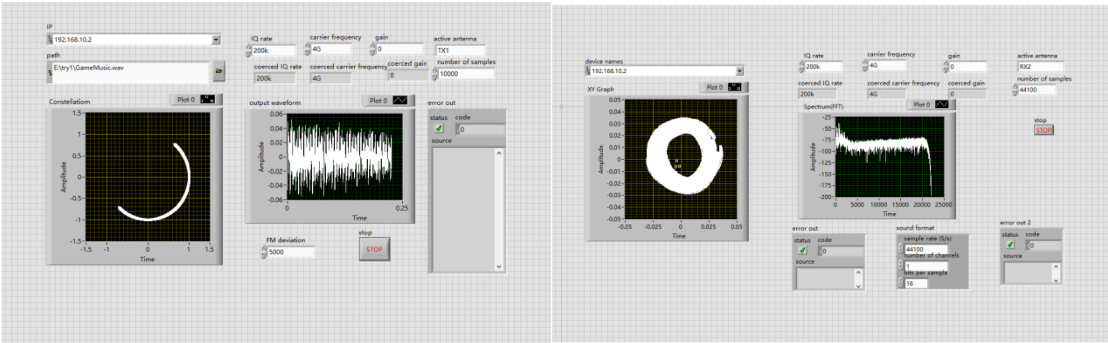


Figure 4.2.3 carrier frequency=4G, FM deviation=5000Hz, IQ rate=200k

IQ Rate

The IQ rate, or the rate of In-phase and Quadrature samples, has a significant impact on the performance of an FM (Frequency Modulation) modulation system. In this experiment, we tried an IQ rate of 50k, 200k and 2M. A low IQ rate (=50 k) can lead to insufficient frequency resolution, which may result in aliasing and degradation of signal quality. On the other hand, an excessively high IQ rate(=2M) can introduce redundant information and increase computational complexity, leading to inefficient resource utilization. Therefore, it is essential to choose an appropriate IQ rate that balances frequency resolution and computational efficiency.

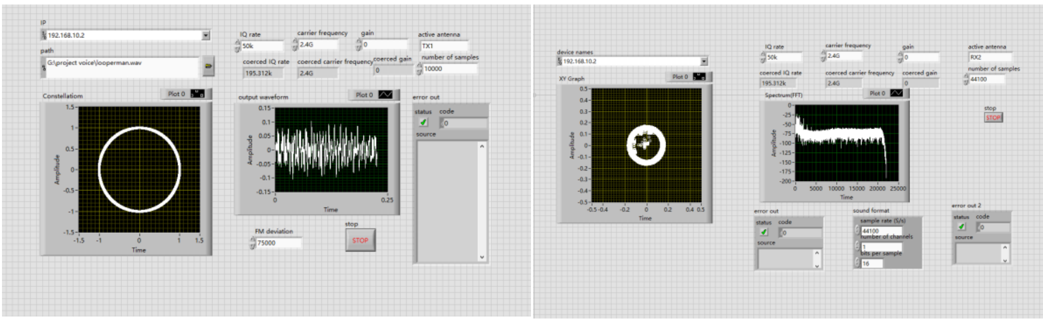


Figure 4.3.1 carrier frequency=4G, FM deviation=75000Hz, IQ rate=50k

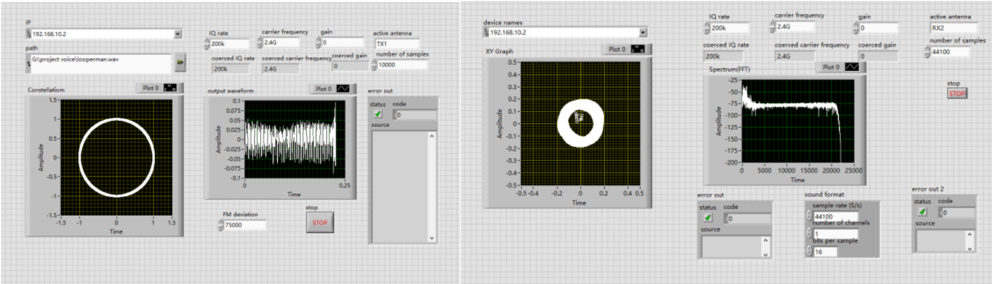


Figure 4.3.2 carrier frequency=4G, FM deviation=75000Hz, IQ rate=200k

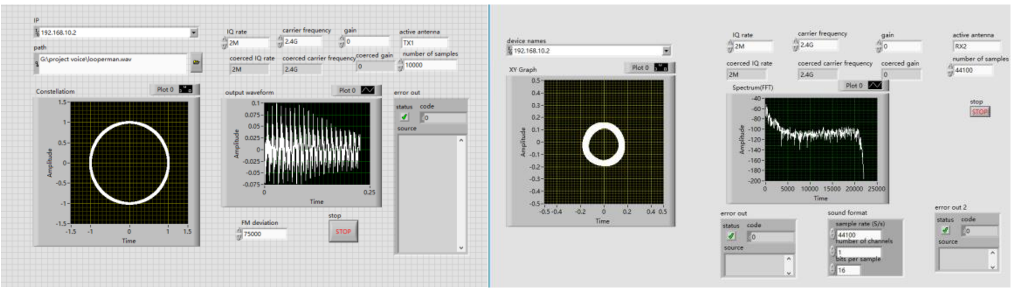


Figure 4.3.3 carrier frequency=4G, FM deviation=75000Hz, IQ rate=2M

About FM Communication Systems

Advantages and Disadvantages

Advantages of FM:

Immunity to Noise: FM modulation is more resistant to noise and interference compared to AM modulation. The information is encoded in the frequency of the carrier, making it less sensitive to noise that affects the amplitude.

Better Sound Quality: FM modulation provides better sound quality for audio transmission, making it suitable for music and broadcasting.

High Fidelity: FM modulation is capable of transmitting high-fidelity signals with a wide frequency range.

Disadvantages of FM:

Larger Bandwidth: FM modulation requires a larger bandwidth than AM modulation. This makes it less efficient in terms of spectrum usage.

Complexity: FM modulation and demodulation circuits are more complex than those used for AM modulation.

Social Applications

FM has been widely used in various fields such as radio broadcasting, telecommunications, and wireless communication.

FM radio broadcasting provides high-quality audio signals to the public.

In addition, FM modulation is used in two-way radio communication systems and mobile communication networks.

Bonus: Low-cost Implementation of FM Systems

In FM modulation, integration is required for modulation, while differentiation is required for demodulation. However, the cost of implementing integrators and differentiators in circuits is relatively high, which is not conducive to low-cost implementation of FM systems.

As an alternative, the cost of filters in circuits is relatively low, so filters could be designed to replace the integrator and differentiator, thereby reducing the implementation cost of the entire circuit.

In the bonus section of this project, we investigate the use of FIR/IIR filter modules as a substitute for integrators and differentiators to achieve FM modulation/demodulation systems with lower cost.

Theoretical Analysis

During the study of Signals and Systems, we realize that differentiation is essentially a high-pass filtering process, and the integration process can also be analyzed in the frequency domain.

Effect of Differencing

- $J(i,j) = |M(i,j) - M(i+1,j+1)| + |M(i+1,j) - M(i,j+1)|$

Based on this, it is feasible in theory to design specific filters to implement differentiation and integration functions by clarifying the frequency response of differentiation and integration.

Hence, replacing the differentiators and integrators in the circuit with filters using this approach is possible.

For differentiation process, a first-order difference can be used as an approximation. Subsequently, the theoretical analysis of using the FIR filter in the signal processing module library to replace the differentiator can be conducted. Suppose x[n] is a discrete signal obtained by sampling x(t). The derivative of the signal, which is the slope of the tangent line, can be approximated by the following equation:

$$\frac{dx(t)}{dt} \simeq \frac{x[n] - x[n-1]}{1} = 1 \cdot x[n] + (-1) \cdot x[n-1]$$

It can be observed from this equation that differentiation in signal processing can be achieved by using a finite impulse response filter.

For integration process, it can be viewed as an infinite impulse response (IIR) process. The theoretical analysis of using IIR filters from the signal processing module library to replace integrators is carried out below.

Assuming T_s is the signal sampling interval, the integration of the original function can be approximated by the sum of the product of the sampled function amplitude and time:

$$\int_0^t x(\tau)\,d\tau \simeq \sum_{k=0}^{n} x\cdot(kT_s)\cdot T_s$$

Now, let y[n] denote the sum, and write the above equation into an autoregressive form, we get:

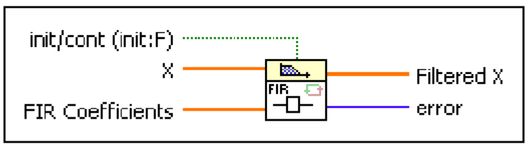
$$1\,y[n] + (-1)\,y[n-1] = (-1)\,x(n\cdot T_s)\cdot T_s$$

In summary, the integration signal processing can be implemented using infinite impulse response (IIR) filters. Implementation in LabVIEW.

LabVIEW Implementation

FIR Differentiator: Due to LabVIEW help, the construction of FIR filters only needs one coefficients array:

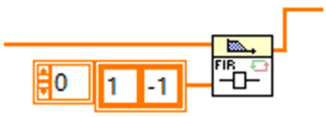
FIR Filter (DBL)



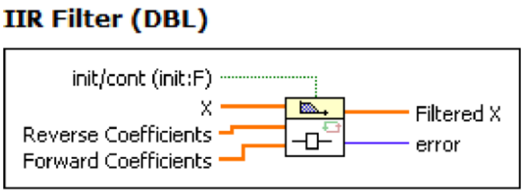
The FIR Coefficients are marked as red in the function:

$$\frac{dx(t)}{dt} \simeq \frac{x[n] - x[n-1]}{1} = 1\cdot x[n] + (-1)\cdot x[n-1]$$

Pass the array and we can construct the FIR differentiator:



IIR Integrator: Due to LabVIEW help, the construction of IIR filters needs reverse coefficients array and the forward coefficients array:



And the mathematical expression for IIR filters in LabVIEW is:

$$y_i = \frac{1}{a_0}\left(\sum_{j=0}^{N_b-1} b_j x_{i-j} - \sum_{k=1}^{N_a-1} a_k y_{i-k}\right)$$

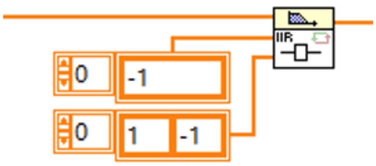
Recall that the autoregressive mathematical form for IIR integrator is:

$$1\, y[n] + (-1)\, y[n-1] = (-1)\, x(n\cdot T_s)\cdot T_s$$

Convert it into the LabVIEW defination form, we could get:

$$y[n] = \frac{1}{\underbrace{1}_{a_0}}\cdot\left(\underbrace{(-1)}_{b_1} x[n-0]\cdot 1 - \underbrace{(-1)}_{a_1} y[n-1]\right)$$

Pass the arrays and we could construct the IIR Integrator:



Simulation Result

As mentioned earlier in the presentation, in addition to completing the simulation of a low-cost FM communication system, I also made some optimizations to the graphical user interface (GUI). Now, users can modulate IQ rate and carrier frequency through sliders, and adjust the gain using a knob. This change makes the program more in line with the design theme of "old-fashioned transceivers", while also enhancing user-friendliness and aesthetics.

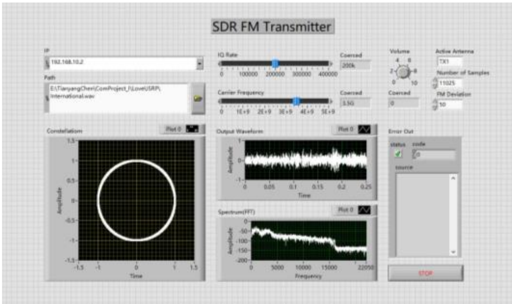
Similar to the functional verification in the basic section, we verify the FM communication system with FIR/IIR filters as follows:

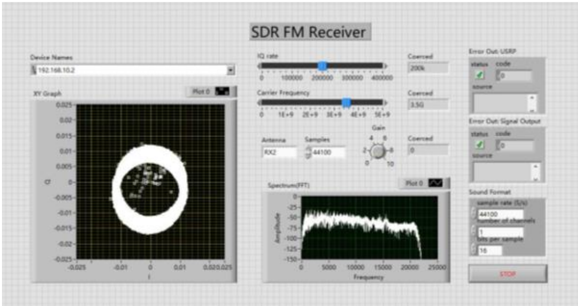
1. Turn on the receiver, without turning on the transmitter, we can only hear noise.
2. Turn on the transmitter, the receiver can play music normally. The sound does have a

"low-cost" feel.

3. Turn off the transmitter, and the receiver returns to noise.

Screenshots of this experimental process are shown below.



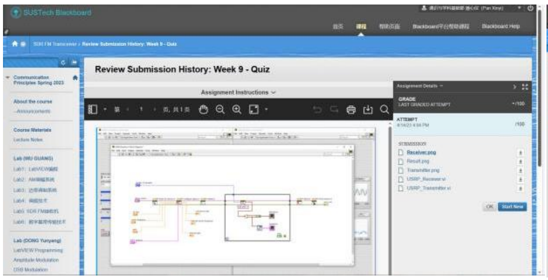
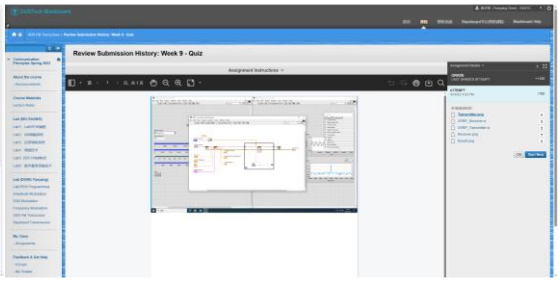


Scan the QR Code to download the screencast of this experiment:

In summary, the FM modulation/demodulation using FIR/IIR modules was successfully simulated in this experiment, achieving the goal of low-cost FM SDR implementation.

Experience

Screenshots of Class Participation

Experience

陈天阳：

1. All source codes, results, presentation resources, and reports of this experiment have been synchronized to https://github.com/EliotChen8/EE206_Lab/tree/main/Project_I. You can access our repository by scanning the QR code below.

This is an essential attempt. My teammate and I will deepen our collaboration based on GitHub to improve version control and collaboration mechanisms of the projects.

2. We encountered many difficulties in the experiment: when designing filters in the bonus part, I did not fully understand the meaning of input and output at first, which caused the system to only receive noise no matter how I adjusted the parameters. Later, I rederived the theory and redesigned the filter parameters, finally the system worked properly. This made me feel that my knowledge of Signals and Systems is not solid enough and needs to be further improved. In the process of improving the results, I found that it is difficult to discuss the impact of changing filter parameters on the system through conventional time-domain and frequency-domain analysis methods. The project has been stuck out of the qualitative research stage, and because the results can only be subjectively described, I believe that the scientific nature of this part is insufficient and should not be included in this report. However, as soon as I find an effective analysis method in the future, I will update the research results on GitHub, so let's wait and see.

3. Nevertheless, in this experiment, we have gained a lot: we clarified the concept and usage of USRP; we successfully built a transmitter-receiver through USRP and software, and constructed our first wireless communication broadcasting system; we successfully verified the signal transmission function of SDR; we successfully implemented the integration and differentiation functions using FIR and IIR filters, and built a low-cost FM SDR system. These experiences have

great significance for our understanding of communication principles, theoretical knowledge, and future communication system design.

潘心仪：

In this experiment, we encountered many problems. For example, transmitter cannot transmit signals, receiver can only receive noise, because the network problems cannot match and so on. This suggests that we should be more careful in the process of experiment.

This experiment also made me have a deeper understanding of USRP, and explored the factors affecting FM modulation system and how to optimize it. I also studied the wide application and value of FM modulation system in reality.

Self-rating and Contribution

Chen Tianyang (100/100): Debugging of the program and the designing of the bonus part, writing of the Introduction section on the principles and LabVIEW Express of the report, writing of the bonus part in the Result and Analysis section, integration and finalizing of the report;

Pan Xinyi (100/100): Background research, Construction of the basic part of the program, writing of the Introduction section on USRP in the report, writing of the basic part in the Result and Analysis section of the report, writing of the corresponding part in the Experience section.