**Project 1: FM Receiver Project**

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| **Introduction**  In this project, we use Producer-Consumer design pattern to design the FM receiver, in this progress we use producer pattern to read the pre-recorded signal and use the consumer pattern to demodulate FM signals.  The FM receiver model is below:  Notes: As being limited by hardware conditions, this FM receiver project was based on the LABVIEW simulation.  **Theoretical Analysis of FM transmitter and receiver**  Consider a sinusoidal carrier wave given by where means the Amplitude of carrier signal, means the frequency of the carrier signal and is the message signal.  Then the instantaneous carrier frequency is ,using the relationship between angle and frequency, , where means frequency sensitivity.  At the receiver, use arctangent demodulation to recover the message signal. As baseband signal was consist of real part and imaginary part, which can be expressed as . Where and .  By using arctangent method, we can recover message signal as follows:  **Lab results & Analysis**  **Implementation of FM receiver by Labview**   1. **Block diagram**     The figure shown above is the block diagram of the FM Receiver implemented by LabVIEW.   1. **Producer-Consumer Design Pattern**   Producer-Consumer Design Pattern is the most basic design pattern in the multithread programming, in another word, it is the composite design pattern that combines event handlers and message handlers. This design pattern includes a producer loop and several consumer loops. In addition, each different loop can run synchronously at different rates.  We create a Producer-Consumer template through “**File -> New -> From Template -> Frameworks -> Design Patterns**”. In this project, we use Producer-Consumer Design Pattern (Data) to implement the FM receiver.   1. **Producer loop**     The figure shown above is the figure of the producer loop we used in the implementation of the FM receiver.  In the producer loop of the Producer-Consumer Design Pattern Data), we usually use it to perform signal acquisition.  In the producer loop, we first use **Read Waveform.vi** module to get the pre-recorded signal data. Then use first in first out (FIFO) queue to control the input and release data which can keep the loop running easily and smoothly. After getting data, the producer loop will transfer the data to consumer loop.   1. **Consumer loop**     The figure shown above is the figure of the consumer loop we used in the implementation of the FM receiver.  In the consumer loop of the Producer-Consumer Design Pattern (Data), we usually use it to perform signal processing and display.  In the consumer loop, we use “arctan” method to build the demodulator. We first get the imaginary and real part of the signal, and then we transform them into exponential form / polar form so that we can get the phase of the signal. And then, we use the unwrap the phase and take the derivative of the signal to get the demodulated signal. After that, we re-sample the demodulated signal and normalize it. Finally, we input the output waveform into the **Sound Output Write.vi** module to implement the sound play.   1. **Demodulation**       The figure shown above is the demodulation part in the consumer loop of the FM receiver. In the demodulation, we use the “arctan” method to demodulate the pre-recorded signal. First, we use the **Decimate 1D Array** module to get the imaginary and real part of the signal, then we use the **Re/Im To Complex** module and the **Complex To Polar** module to transform the complex signal to its polar form. After the **Re/Im To Complex** module, we get the XY Graph to display the imaginary and real parts of the signal respectively. And we use the **Unwrap Phase.vi** module to eliminate discontinuous phase points and take the derivative of the signal, then we can get the demodulated signal. After that, we need to re-sample the demodulated signal use **Rational Resample.vi** module so that we can play it on the audio player side.  **Implementation of FM receiver by DLL**  **Using Dynamic Link Library to implement FM demodulation**  Dynamic Link Library (DLL): a dynamic link library is a program module that can be shared by multiple software, and it has well encapsulation internally. Labview provides powerful external interface capabilities to call programs written in other languages.  In the project of the FM receiver, the FM arctangent demodulation method is implemented using MATLAB, the dynamic link library is generated through the CODER toolbox, and then the DLL is imported into the LABVIEW programming software, and then a custom subvi file is generated and imported into the project to be used.  The progress model is as follows.  **Step1:**  Write arc-tangle demodulation function in MATLAB, which include atan function, unwrap and differentiate function to demode the transmitted signals, recall the analysis of demodulation progress by arctangent method.  FMDemoRTLSDR\_DLL function in MATLAB:    **Step2:** Using MATLAB CODER toolbox to generate the DLL by reviewing code generation, defining the input type and checking the running-time.  **Step3**: import the DLL generated bellow in LABVIEW, then it can create a subvi in your user library. After done this, we can add this subvi in our project. But remember to correct the default setting of the parameters of the input type, otherwise the music played out will sound card.  The block diagram:    **Evaluation of the result**   1. **The influence of IQ Rate**   From the project, what can be known is that IQ rate should be among the range of [275000,315000]. In this range, the music signal can be demodulated relatively perfectly. Otherwise, the music played is unstable and distortional.    ***IQ rate=300k Hz***    ***IQ rate=230k Hz IQ rate=400k Hz***   1. **The influence of numbers of channel& sample rate**   After the experiment, a phenomenon what can be found is that the music signal can be perfectly played only when the setting of channel and sample rate are (1,44100) or (2,22050).    ***channel=1 sample rate=44100 channel=2 sample rate=22050***    ***channel=3 sample rate=14700***   1. **The control of music length that played**     What can be found is that this module can control the length of music played.  As the number increases, the length gets longer.    ***constant=40***    ***constant=100***    ***constant=160***  **User Interface**    This is our design of user interface. We find a picture of FM radio interface which serves as the background. What’s more, time indicator is added to Producer-Consumer Design Pattern so that it can display the time in real time when running.  **Design of multi-channel system**    The basic idea for the multi-channel system is that we modulate a single signal use same type of carrier signal with different carrier frequency separately, and then add them up and pass them into the channel. In the receiver, we use bandpass filter with different passband to separate the composite signals and demodulate them separately to get the recovered signal.   1. **Block diagram** 2. **Single tone signal**     The figure shown above is the single tone test of the multi-channel FM system. In this test, we designed three channels for testing.   1. **WAV signal**     The figure shown above is the WAV signal test of the multi-channel FM system with three channels for testing.   1. **Modulation** 2. **Single tone signal**     The figure shown above is the modulation part of the multi-channel FM. We first use **Cluster** data structure to store the parameters needed for modulation, and use **Unbundle By Name** module to input the cluster elements whose names have been specified to the **FM subVI** to modulate the input signal. After modulation, we add three modulated signals and AWGN together.    The figure shown above is the internal block diagram of the **FM subVI**. This subVI has four inputs which are three modulation parameters Frequency Deviation, Carrier Frequency and Carrier Amplitude, and the baseband signal. And its output is the modulated signal.      For the process of the frequency modulation, we begin by narrow-band modulation of the baseband signal to get the NBFM signal. First, integrate the signal, take its sine and cosine, and multiply them by the sine and cosine carrier signals respectively. Then, add them together to get the NBFM signal.  We can get the WBFM signal by increasing the frequency deviation which can improve the anti-noise performance but need to consumes more bandwidth.   1. **WAV signal**   The process of the modulation of the WAV signal is same as the single tone test, so we don't repeat it here.   1. **Demodulation** 2. **Single tone signal**     The figure shown above is demodulation of the modulated signal. In the process of the demodulation in the multi-channel system, we use the case structure and a bandpass filter with different passband to extract only one signal from the composite signal for demodulation. In order to be able to implement the above process, we create a numeric control to ensure that we can extract different signal components in different cases. And we set the upper cutoff frequency to the carrier signal frequency plus half of the baseband signal frequency. And similarly, we set the lower cutoff frequency to the carrier signal frequency minus half of the baseband signal frequency. For demodulation, we use the “arctan” method to first get the in-phase component and the antiphase component of the modulated signal. And then use the arctan function to get the instantaneous phase, then take the derivative of it to get the recovered signal.    The figure shown above is the internal block diagram of the **FM Demodulation subVI**. This subVI has four inputs which are Waveform(of the modulated signal), Dynamic Data(of the modulated signal), estimated carrier frequency(from the phase locked loop) and LPF Cut-Off. And its output is the demodulated signal.    For the process of the demodulation, we use the phase loocked loop to recover the carrier frequency. We first use the phase locked loop to get the estimated carrier frequency and create one sine signal and cosine signal whose frequency is what we get from PLL. And then multiply the modulated signal with those two sinusoidal signals respectively and pass through a low pass filter to get the in-phase component and the antiphase component, then use the “arctan” method (1. Get the arctan of I and Q. 2. Unwrap phase) and take the derivative of the result from the “arctan” method to get the recovered signal.    Phase locked loop (PLL) is a negative feedback regulation system. It consists of phase discriminator, loop filter and voltage controlled oscillator (VCO). VCO can output sine wave with different frequency according to the amplitude of the input voltage. In the PLL, in short, it is to obtain the desired system model parameters by constantly observing and adjusting the estimated parameter values. In this situation, we want to use PLL to get the parameter values of the received signal r(t), so when we get the maximum DC component, i.e., v(t)=v(t)max, and doesn't change over time, which means we get the parameter value which is closest to r(t).   1. **WAV signal**     The process of the demodulation of the WAV signal is same as the single tone test, so we don't repeat it here. Something different is that we directly set the upper cutoff frequency to the carrier signal frequency plus 1KHz, and set the lower cutoff frequency to the carrier signal frequency minus 1KHz.  **Analysis for multi-channel system**   1. **Waveform of baseband signal, FM frequency spectrul, and demodulated signal**  * **Single tone signal**             Some parameters of modulation is shown above. We can know that we set modulation frequency=2KHz, LPF cut-off=5KHz, WAV Sample Rate=4.41KHz, signal to noise ratio SNR=50, and the modulation frequency for three baseband signals are all 2KHz. For channel 1, we set the carrier amplitude=1, carrier frequency=100KHz, and frequency deviation=20KHz. For channel 2, we set the carrier amplitude=1, carrier frequency=200KHz, and frequency deviation=20KHz. For channel 3, we set the carrier amplitude=1, carrier frequency=300KHz, and frequency deviation=20KHz.    Unit of frequency: Hz  The FM signal’s power spectrum with AWGN in frequency domain is shown above. From the power spectrum, we can know that each WBFM signal takes up a lot of bandwidth, but the center of their spectrum is at their carrier frequency.   * **WAV signal**               Some parameters of modulation is shown above. We can know that we set LPF cut-off=5KHz, Resample Rate=1MHz, Duration=44101, WAV Sample Rate=4.41KHz, and signal to noise ratio SNR=100. And we read three different audio files on three channels. For channel 1, we set the carrier amplitude=1, carrier frequency=100KHz, and frequency deviation=30KHz. For channel 2, we set the carrier amplitude=1, carrier frequency=250KHz, and frequency deviation=20KHz. For channel 3, we set the carrier amplitude=1, carrier frequency=400KHz, and frequency deviation=20KHz.    Unit of frequency: Hz  The FM signal’s power spectrum with AWGN in frequency domain is shown above. From the power spectrum, we can not only know that the WAV signal has more than one frequency components, and thereby we can guess that many sounds in our lives consist of many single tone signals with different frequencies. But also we can find that different WAV signals have different power spectrum. In the multi-channel system, we move different WAV signal to different bands of the channel for transmission.   1. **Performance Evaluation**   In this part, we take channel 2 as an example to do the performance evaluation in the multi-channel(3 channels here) system.   1. **SNR**   Unit of time: s  Unit of frequency: Hz   * **Single tone signal** * **SNR=-10**      * **SNR=0**      * **SNR=30**      * **SNR=50**     We set fcut-off=5KHz, and frequency deviation=20KHz. From the figures above, we can find that when SNR is low, the spectrum of FM signal will be seriously disturbed by noise. And as SNR increases, the spectrum becomes better and better. If we listen to the demodulated signal, as SNR increases, the sound is getting more and more clear; on the contrary, as SNR decreases, the sound is getting more and more unclear. When SNR=0, we can almost hear nothing but noise, which means the demodulation is fail, and this is because of the low SNR so that the noise is dominant.   * **WAV signal** * **SNR=-10**      * **SNR=0**      * **SNR=50**      * **SNR=100**     We set fcut-off=5KHz, and frequency deviation =30KHz. From the figures above, we can find that when SNR is low, the spectrum of FM signal will be seriously disturbed by noise, and we can hardly see the presence of the modulated signals in the spectrum. And as SNR increases, the spectrum becomes better and better, and base noise in the spectrum become less and less. If we listen to the demodulated signal, as SNR increases, the sound is getting more and more clear; on the contrary, as SNR decreases, the sound is getting more and more unclear. When SNR=0, we can hardly hear the sound but noise, when SNR=-10, we cannot hear anything at all but noise, which means the demodulation is fail, and this is because of the low SNR so that the noise is dominant.   1. **Cutoff Frequency**   Unit of frequency: Hz   * **Single tone signal** * **fcut-off=5KHz**      * **fcut-off=10KHz**      * **fcut-off=15KHz**     We set SNR=50, and frequency deviation=20KHz. From the figures above and also by listening the demodulated signal, we can find that as the cutoff frequency increases, there are more noise in the demodulated signal. We can also confirm it by playing the sound, and we can conclude that the lower cutoff frequency, the performance is better. However, we need to notice that the cutoff frequency should not be too low since the filter is not ideal LPF, and also as the WBFM signal takes up a lot of bandwidth. Its frequency response below its cutoff frequency is not flat. Thus, if we set the cutoff frequency too low, there will be an attenuation in the demodulated signal.   * **WAV signal** * **fcut-off=5KHz**      * **fcut-off=10KHz**      * **fcut-off=15KHz**     We set SNR=100, and frequency deviation=20KHz. From the figures above and also by listening the demodulated WAV signal, we can find that as the cutoff frequency increases, there are more noise in the demodulated WAV signal. We can also confirm it by playing the sound, and we can conclude that the lower cutoff frequency, the performance is better. However, we need to notice that the cutoff frequency should not be too low since the filter is not ideal LPF, and also as the WBFM signal takes up a lot of bandwidth. Its frequency response below its cutoff frequency is not flat. Thus, if we set the cutoff frequency too low, there will be an attenuation in the demodulated signal.   1. **Frequency deviation**   Unit of frequency: Hz   * **Single tone signal** * **Frequency deviation=0**      * **Frequency deviation=20000**      * **Frequency deviation=40000**     We set SNR=50, fcut-off=5KHz. In the FM FFT waveform graphs, we can see that when frequency deviation=0, the spectrum of the modulated signal just has one frequency component which is at its carrier frequency, and the quality of the sound we heard is not very good either. And as we increase the frequency deviation, the quality of the sound we heard is better and better. So we can conclude that as the increase of the frequency deviation, the FM system can have a better anti-noise performance. However, something we need to notice is that, as the frequency deviation increasing, i.e., as the bandwidth occupied by the modulated signal increases, the frequency component at its central location, i.e., its carrier frequency, may be smaller than the adjacent frequency component. Then, the quality of the sound we hear is correspondingly worse because we set the upper cutoff frequency of the bandpass filter to the carrier signal frequency plus half of the baseband signal frequency, similarly, we set the lower cutoff frequency of the bandpass filter to the carrier signal frequency minus half of the baseband signal frequency.   * **WAV signal** * **Frequency deviation=0**      * **Frequency deviation=30000**      * **Frequency deviation=50000**     We set SNR=100, fcut-off=5KHz. In the FM FFT waveform graphs, we can see that when frequency deviation=0, the spectrum of the modulated signal just has one frequency component which is at its carrier frequency, and we cannot hear any music sound. And as we increase the frequency deviation, the quality of the sound we heard is better and better. So we can conclude that as the increase of the frequency deviation, the FM system can have a better anti-noise performance. However, something we need to notice is that, as the frequency deviation increasing, i.e., as the bandwidth occupied by the modulated signal increases, the frequency component at its central location, i.e., its carrier frequency, may be smaller than the adjacent frequency component. Then, the quality of the sound we hear is correspondingly worse because we set the upper cutoff frequency of the bandpass filter to the carrier signal frequency plus 1KHz, and set the lower cutoff frequency of the bandpass filter to the carrier signal frequency minus 1KHz. | |
| **Experience**   1. In this project, I carry out the implement of FM demodulation by DLL and the evaluation of results. From this progress, I experience the strong power of the interface of LABVIEW. And it is so convenient to use DLL to generate the user library subvi in LABVIEW, it can help to deal with too much problems. Also, in the progress of this project, I learn a lot about the team work, how to divide team division of labor and the importance of communication.   ---杨佳怡(FM receiver by DLL, Performance evaluation, PPT making, report writing)   1. In this project, I designed the UI interface and consulted the background knowledge. It is convenient to design the UI interface because of powerful graphical interface design capabilities of Labview. Also, I help the design of multiple channels and performance evaluation.   ---(张旭东 User Interface, Background, Performance evaluation, PPT making, report writing)   1. In this project, I was responsible for the implement of FM receiver by Labview and FM multi-channel system. It’s convenient to use the Producer-Consumer Design Pattern to conduct system simulation because of its ingenious design, which is use queue data structures to buffer data, prevent data blocking, and multithreaded programming. In the FM receiver, we find that if we put the demodulated signal directly into the sound player from **Build Waveform** module, we cannot play the sound clearly. Thus, we add a **Normalize Waveform.vi** module after it, and the demodulated signal can be played successfully. In this project, we find that there are many repeated part in the FM receiver and FM multi-channel system, so we create these repeated parts as subVI, such as FM modulator and FM demodulator.   ---(孙逸涵 FM receiver by Labview, FM multi-channel system, PPT making, report writing)   1. The screenshots of real-time completion of Tencent class are shown below: | |
| **Score** | 100 |