**Project 1：FM receiver**

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| **Implementation of FM receiver**   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.  **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 signal 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. | |
| **Experience** | |
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