Journal of Electronic Research and Application, 2024, Volume 8, Issue 6
http://ojs.bbwpublisher.com/index.php/JERA
ISSN Online: 2208-3510
ISSN Print: 2208-3502

Theoretical Analysis and Design Implementation of FM Broadcast Receiving System based on SDR

Yalan Yin*, Qi Wang, Jianrui Li, Chunliu He

Electronic Information Engineering College of Sanjiang University, Nanjing 210029, China

Corresponding author: Yalan Yin, yi_yalan@sju.edu.cn

Abstract: ADALM Pluto is a software-defined radio product. Based on theoretical analysis, this paper designs a Frequency Modulation (FM) broadcast receiving system using ADALM Pluto as a wireless receiver on the Simulink platform. The simulation results show that the theoretical analysis is correct and the parameter settings are reasonable, and the Software-Defined Radio (SDR) has the characteristics of high performance, high ease of use, and low cost.

Keywords: SDR; FM broadcasting; Modulation and demodulation; Simulink

Online publication: December 2, 2024

1. Introduction

Software-Defined Radio (SDR) technology has transformed many areas of modern communications and networking. Traditional hardware-based radios are designed for one or a small number of applications, and the functionality of a single SDR can be modified through firmware updates and according to changes in back-end signal processing to enable multiple system applications. Therefore, the application of SDR to modern communication systems is a topic worthy of in-depth research and exploration.

2. The mechanism of China's FM broadcasting receiving system

The frequency band used in China's FM broadcasting regulations ranges from 87.5 MHz to 108 MHz, with a total bandwidth of 20.5 MHz. The maximum frequency offset that can be used is 75 kHz, and 25 kHz is left between the channels as a guard interval. Therefore, the baseband signal bandwidth for FM broadcasting is 100 kHz, and the station searches for different channels generally in steps of 100 kHz.

3. The overall design of the FM broadcast receiving system

3.1. Block diagram of the overall design

The design environment of the system is the Simulink platform of MATLAB. The design model of the FM broadcast receiving system is composed of the ADALM Pluto Receiver module, filter 1 module, FM Demodulator Baseband module, filter 2 module, down-sampling module, and Audio Device Writer module.

3.2. Signal processing model analysis of the ADALM Pluto wireless receiver

The FM signal time domain expression is given by

$$s_{\mathrm{FM}}(t) = A\cos\left[\omega_c t + K_f \int_{-\infty}^{t} m(\tau)d\tau\right] \tag{1}$$

Where, $S_{FM}(t)$ is the received FM broadcast signal, which is the modulated signal; A is the amplitude of the carrier wave; ω_c is the carrier angular frequency; K_f is frequency modulation sensitivity; and $m(\tau)$ is baseband audio signal. FM signal time domain expression exponent form is

$$s_{\mathrm{FM}}(t) = \frac{A}{2}\left[e^{j[\omega_c t + K_f \int_{-\infty}^{t} m(\tau)d\tau]} + e^{-j[\omega_c t + K_f \int_{-\infty}^{t} m(\tau)d\tau]}\right] \tag{2}$$

The received signal is first multiplied with the local carrier to obtain: $e^{-j\omega_c t}$, $s_{\mathrm{A}}(t)$

$$\begin{aligned} s_{\mathrm{A}}(t) &= s_{\mathrm{FM}}(t)\cdot e^{-j\omega_c t} \\ &= \frac{A}{2}\left[e^{j[\omega_c t + K_f \int_{-\infty}^{t} m(\tau)d\tau]} + e^{-j[\omega_c t + K_f \int_{-\infty}^{t} m(\tau)d\tau]}\right]\cdot e^{-j\omega_c t} \\ &= \frac{A}{2}\left[e^{j[K_f \int_{-\infty}^{t} m(\tau)d\tau]} + e^{-j[2\omega_c t + K_f \int_{-\infty}^{t} m(\tau)d\tau]}\right] \end{aligned} \tag{3}$$

$S_A(t)$ is the signal passed through a low-pass filter that filters out the high frequencies, leaving the baseband signal: $2\omega_c$, $y(t)$

$$\begin{aligned} y(t) &= LPF\{s_{\mathrm{FM}}(t)\cdot e^{-j\omega_c t}\} \\ &= \frac{A}{2} e^{j[K_f \int_{-\infty}^{t} m(\tau)d\tau]} \end{aligned} \tag{4}$$

Equation (4) can also be written as

$$y(t) = \frac{A}{2} e^{j\varphi(t)} \tag{5}$$

Where this means that the input audio signal is a multiple of the derivative of the phase $\varphi(t) = K_f \int_{-\infty}^{t} m(\tau)d\tau$, $m(t)$, $\varphi(t)$. In the following, the frequency domain processing process of the received signal of FM broadcast by ADALM Pluto based on SDR is analyzed. The frequency domain expression of an FM broadcast signal is given by

$$s_{\mathrm{FM}}(f) = \mathrm{Y}(f + f_c) + \mathrm{Y}(f - f_c) \tag{6}$$

Where, $Y(f)$ is the frequency domain of the baseband modulated signal, $y(t)$.

Since the Fourier transform is given by $e^{-j\omega_c t}$:

$$e^{-j\omega_c t} \Leftrightarrow 2\pi\,\delta(f + f_c) \tag{7}$$

The multiplier pair of ADALM Pluto is multiplied with in the time domain, then, is convolved with in the frequency domain. Therefore, the frequency domain expression of the signal after the multiplier is $S_{FM}(t)$, $e^{-j\omega_c t}$,

$S_{FM}(f)$, $2\pi\delta(f+f_c)$, and $S_A(t)$.

$$s_A(f) = 2\pi Y(f+f_c+f_c) + 2\pi Y(f+f_c-f_c) = 2\pi Y(f+2f_c) + 2\pi Y(f) \tag{8}$$

The signal is passed through a low-pass filter to obtain the frequency domain of the low-pass signal $2\pi Y(f)$. Note that the baseband signal is not the frequency domain of an FM broadcast baseband audio signal $m(t)$ and $m(f)$.

3.3. Filter 1 processing of the signal

The function of filter 1 is to filter out the out-of-band noise. The output signal of this filter can be regarded as the same as the input signal, and is

$$y(t) = \frac{A}{2}e^{j\varphi(t)} \tag{9}$$

3.4. FM modulator baseband processing of the signal

The FM modulator baseband module recovers the audio baseband signal from the input signal using the baseband delay demodulation method $y(t)$ and $m(t)$. FM modulator baseband module input signal $y(t)$ is

$$y(t) = \frac{A}{2}e^{j\varphi(t)} \tag{5}$$

Pass $y(t)$ through a delay module that delays a sampling time interval T, and the delayed signal $y(t)$ is

$$y_1(t) = \frac{A}{2}e^{j\varphi(t-T)} \tag{11}$$

The signal obtained after conjugation of the pair is $y_1(t)$ and $y_2(t)$.

$$y_2(t) = \frac{A}{2}e^{-j\varphi(t-T)} \tag{12}$$

Will be multiplied by $y_2(t)$, $y(t)$, and $w(t)$.

$$\begin{aligned} w(t) = y(t)\cdot y_2(t) &= \frac{A^2}{4}e^{j\varphi(t)}\cdot e^{-j\varphi(t-T)} \\ &= \frac{A^2}{4}e^{j[\varphi(t)-\varphi(t-T)]} \end{aligned} \tag{13}$$

After sampling the signal, it is obtained

$$w_n = w(nT) \tag{14}$$

Substitute into **Equation (13)** to obtain $t = nT$

$$w_n = \frac{A^2}{4}e^{j[\varphi_n-\varphi_{n-1}]} \tag{15}$$

After passing the angle() module, you get

$$v_n = \varphi_n - \varphi_{n-1} \tag{16}$$

Signal Vn is the approximate derivative of, therefore φ_n is

$$m_n \approx \varphi_n \tag{17}$$

Where, m_n is the sampling value of the FM broadcast audio baseband signal. In this way, the output of the FM modulator baseband module is the sampled value of the FM broadcast audio baseband signal. Reduce the rate of the signal appropriately, and then use the Audio Device Writer module to use the audio equipment of the computer to play the FM broadcast signal.

4. FM broadcast receiving system module parameter settings

4.1. ADALM Pluto receiver module

The center frequency is set to 105.8 MHz (receiving Nanjing Music Station). The baseband sample rate, according to the Nyquist low-pass sampling theorem, is at least twice the baseband signal bandwidth. Since the baseband signal bandwidth of the FM broadcast signal is 100 kHz, the sampling rate is at least 200 kHz. The baseband sampling rate is required to be an integer multiple of the audio sampling rate. The audio sampling rate of the Audio Device Writer module is 48 kHz, so the baseband sampling rate needs to be an integer multiple of 48 kHz. To sum up, the baseband sampling rate is set to 240 kHz.

For the samples per frame, the frame size is required to be no higher than two times the audio sampling rate. For this reason, the frame size can be set to a maximum of 96 kHz to achieve maximum real-time performance in signal processing.

4.2. Filter 1

Filter 1 is used to filter out the out-of-band noise of the baseband signal. The Filter is implemented by the Digital Filter Design module of the Simulink module library. Since the signal coming out of the ADALM Pluto Receiver module is the baseband signal of the FM broadcast signal, the type of the digital filter is set to low-pass. The bandwidth requirement is not less than 100 kHz of the baseband signal bandwidth of the FM broadcast signal, so the sampling rate is set to 240 kHz.

4.3. Baseband FM demodulator

The Baseband FM Demodulator is implemented using the FM Demodulator Baseband module of the Simulink module library. Since China stipulates that the maximum frequency deviation allowed by a radio station is 75 kHz, the frequency deviation is set to 75 kHz.

4.4. Filter 2

Filter 2 is implemented using the Digital Filter Design module of the Simulink module library. Its function is to get the FM broadcast mono audio signal and filter out the out-of-band noise at the same time. Therefore, its type is set to low-pass type. The bandwidth is the same as the bandwidth of the FM broadcast baseband audio signal, $m(t)$. Since the mono audio signal spectrum is between 30 Hz and 15 kHz, the cutoff frequency (Fstop) can be set slightly above 15 kHz.

4.5. Down-sampling module

The down-sampling module is implemented with the Downsample module of the Simulink module library, which resamples the input by discarding K-1 consecutive samples after each sample of the output. Since the baseband sampling rate of 240 kHz is five times the audio sampling rate of 48 kHz, the "Downsample factor, K" parameter of the Downsample module is set to five.

4.6. Audio Device Writer module parameter settings

This module requires the input frame size of the ADALM Pluto Receiver to be less than or equal to twice the sampling rate of this module. The sampling rate of this module is 48 kHz, therefore, the input frame size in the ADALM Pluto Receiver module is set to 96 kHz.

5. Simulate the FM broadcast receiver system

Through the analysis and study of the signal processing process of the designed FM broadcast receiving system based on SDR, the optimal parameter values for each module of the system are determined. These optimal parameters are applied to each module, and after simulating the system, the program of Nanjing Music Station (105.8 MHz) is successfully received. By analyzing the quality of the received audio broadcast signal and observing the oscilloscope and spectrum analyzer images, it can be concluded that the quality of the received audio broadcast signal is excellent.

6. Concluding remarks

Through the simulation operation of the designed FM broadcast receiving system based on SDR, it is concluded that the received signal quality of FM broadcast is excellent, which verifies the correctness of the theoretical analysis and the rationality of the parameter setting, and shows that SDR has the characteristics of high performance, high ease of use and low cost.

Funding

Jiangsu Excellent Engineer Education and Training Program 2.0 Professional Construction Project (Project No. 46), Jiangsu University Brand Construction Project Phase II (Third Batch) Project (Project No. 452), and Teaching Construction and Reform Project of Sanjiang University (Project No. J23007).

Disclosure statement

The authors declare no conflict of interest.

References

[1] Wang QD, et al., 2017, Principles of Communication (2nd edition). Publishing House of Electronics Industry, Beijing, 2017: 86–87.

[2] Xiang J, 2022, Practical Tutorial of Communication Principles—Simulation and Analysis using MATLAB. Tsinghua University Press, Beijing, 2022: 103–111.

[3] Gao YY, Wei YM, Guo MX, et al., 2020, Principles of Communication (3rd Edition). China Machine Press, Beijing, 2020: 53–60.

[4] Yu YB, et al., 2017, Digital Signal Processing. Southeast University Press, Nanjing, 2017: 09.

[5] Huang B, 2020, Simulation and Implementation of FIR Digital Filter Based on Frequency Sampling Method. Science and Technology Innovation, 2020(33): 122–123.

[6] Hua X, 2020, Speech Signal Processing Based on Matlab. Modern Information Technology, 4(17): 76–78.

[7] Shuo J, Li Z, et al., 2021, Noisy Speech Signal Processing Based on Matlab. Electroacoustic Technology, 45(07): 7–9.

[8] Zhong C, Chong C, et al., 2017, Design of Control System Based on MSP430 Single Chip Microcomputer. Tsinghua University Press, Beijing.

[9] Liu C, Chen L, Yuan CL, et al., 2019, Development Status and Innovation of Radio communication Technology. Electronic Technology and Software Engineering, 2019(24): 31–32.

[10] Yin YW, Ma JT, Shi L, et al., 2021, Design and Application of IIR Digital Filter Based on Matlab. China Modern Educational Equipment, 2021(21): 31–33.

[11] Chen JL, Kang L, Li GW, et al., 2020, FIR Digital Filter Design Based on Matlab and Its Application in Receiver. Armamentarium Automation, 39(12): 33–35.

[12] Yu LL, Yang MZ, Yu Y, et al., 2021, Design of Communication Principle Experimental Platform based on Software Defined Radio. Jiangsu Communication, 37(01): 46–48.

[13] Zhang D, Han S, et al., 2019, Design of IF Signal Processing System Based on Software Defined Radio Platform. Computer Measurement and Control, 27(11): 4.

[14] Zhang SY, Xu WQ, et al., 2012, Communication Principle and MATLAB/Simulink Simulation. Posts and Telecommunications Press, Beijing, 201209. 322.

[15] Wang JX, Tian XP, Cao HM, et al., 2011, Foundations of Information Theory and Coding. Posts and Telecommunications Press, Beijing, 201109.434.