Optimized design of QPSK-OFDM based audio transmission system

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ABSTRACT

With the rapid development of wireless communication technology, wireless audio transmission plays an increasingly important role in modern life. This article designs and optimizes an audio transmission system based on QPSK modulation and OFDM baseband modulation using eLabRadio software and the eNodeX 10F-DS hardware device. The system incorporates key techniques such as CVSD (Continuous Variable Slope Difference), convolutional coding, and differential coding, with optimized parameter adjustments. Test results show that, after optimization, the system can achieve high-quality, real-time audio transmission at distances of 0-7 m with a transmit attenuation of 10 dB and a receive gain of 50 dB. At a fixed distance of 2 m, the best audio transmission is achieved with a receive gain in the range of 50-67 dB and a transmit attenuation of 10 dB.

Keywords: Audio transmission system, QPSK, OFDM, eLabRadio software, eNodeX 10F-DS hardware

1. INTRODUCTION

With the rapid development of information technology, wireless communication plays a crucial role in daily life[1]. Wireless audio transmission, a key application, finds wide use in scenarios like wireless headphones and microphones[2]. However, wireless channels are susceptible to multipath effects, fading, and other factors that significantly impact signal transmission quality[3,4]. Investigating techniques that guarantee real-time audio transmission and high transmission quality in complex wireless environments is thus of paramount theoretical and practical importance[5].

Quadrature Phase-Shift Keying (QPSK) and Orthogonal Frequency-Division Multiplexing (OFDM) are two modulation techniques widely used in wireless communication systems[6]. QPSK offers high spectral and power efficiencies, effectively mitigating channel fading and multipath effects[7]. On the other hand, OFDM divides a high-speed data stream into multiple low-speed sub-streams and applies QPSK modulation to each, significantly enhancing the system's resistance to multipath and improving spectral efficiency[8]. Therefore, integrating QPSK and OFDM techniques in designing an audio transmission system based on OFDM baseband modulation is expected to achieve high-quality audio transmission in wireless environments.

The aim of the article is to provide an effective solution based on QPSK and OFDM baseband modulation for research and engineering applications in the field of wireless audio transmission in order to achieve the goal of high-quality audio transmission in complex wireless environments.

2. SYSTEM INTRODUCTION

The system is built on the software radio platform by Wuhan Lintec Electronic Technology Co., Ltd., featuring the eNodeX 10F-DS hardware and eLabRadio software. Both use standard interfaces, facilitating integration with other general-purpose software and SDR hardware platforms.

2.1 Hardware introduction

The eNodeX 10F-DS device has an aluminum portable design exterior and houses the AD9363 RF front-end and the Xilinx ZYNQ-7010 baseband processing unit. The whole hardware device is divided into the following parts:

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(1) RF front-end: it integrates a 12-bit resolution digital-to-analog converter (DAC) and an analog-to-digital converter (ADC), and is equipped with one transmitter and one receiver channel; its RF operating frequency range is from 70 MHz to 3 GHz, and the maximum channel bandwidth that can be adjusted is up to 20 MHz[9].

(2) A/D and D/A high speed conversion

(3) Signal processing unit

(4) FPGA baseband processing unit: Xilinx's AP SoC combines a dual-core Cortex-A9 ARM processor with the ZYNQ-7010 FPGA. The baseband processing unit supports comprehensive system design with integrated memory, audio I/O, bi-directional USB, Ethernet, and an SD card slot[10].

2.2 Communication system design software

eLabRadio software uses a graphical interactive interface, contains more than 100 algorithmic knowledge particles, supports hardware and software to build a variety of wired and wireless communication systems; the main functional modules are as follows:

(1) Hardware interface module: the system selects the RF transceiver module based on Pluto driver, which is up and down converter supported by USB interface through IQ; software radio devices include Pluto-driven eNodeX10F series and ADI's ADALM-PLUTO[11].

(2) Virtual instrumentation: contains three instruments: oscilloscope, BER analyzer and spectrum analyzer.

(3) Source compilation code: contains two types of PCM and CVSD.

(4) Channel compilation code: contains modules such as Hamming compilation code, differential compilation code and convolutional compilation code.

(5) Modulation and demodulation: contains analog modulation and demodulation, binary modulation and multivariate modulation and demodulation.

(6) Filter: Various types of filters can be selected, such as extraction filter and interpolation filter.

3. SYSTEM DESIGN AND IMPLEMENTATION

3.1 Simulation program construction

The simulation scheme of the audio transmission system is built in the eLabRadio software, and the system topology is shown in Figure 1. In the software, the data reading module is selected as the source, the specified audio file is loaded and CVSD coding is chosen. Convolutional and differential coding are used for channel coding, QPSK modulates the signal, and the baseband signal is input to the OFDM modulation module to generate the IQ signal, which is digitally upconverted and sent through the CIC interpolation filter. The RF signal at the receiver end is digitally downconverted to generate the IQ signal, extracted and filtered by the CIC, and undergoes OFDM demodulation, QPSK demodulation, channel and source decoding, and is finally output in eLabRadio. After building the simulation scheme, CIC filters are added at the transmitter and receiver ends, and parameters are adjusted to ensure rate matching.

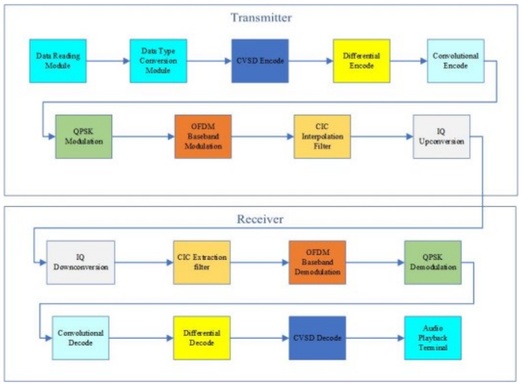

Figure 1. Simulation scheme system topology.

3.2 Self-transmit-self-receive system

The self-transmit-self-receive system modifies some modules in the simulation scheme and uses one 10F-DS hardware device for standalone transmission and reception, as shown in Figure 2. The IQ upconversion module is replaced with two modules: real-to-complex and eNodeX10F wireless transmission. The IQ downconversion module is replaced with two modules: eNodeX10F wireless reception and complex-to-real. After modifying the modules, rate matching and parameter adjustments are required. The original simulation scheme's module parameters remain unchanged, only the new wireless transmitter and receiver module parameters need to be set.

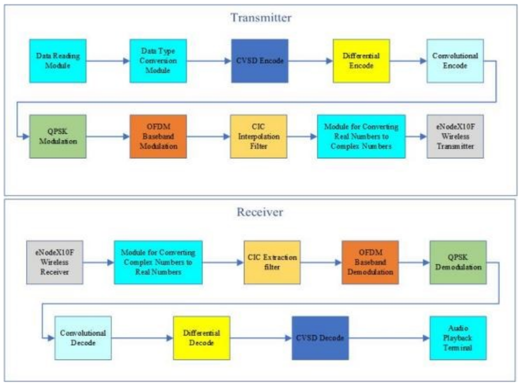

Figure 2. Topology of self-sending and self-receiving system.

3.3 One-transmit-one-receive system

The one-transmit-one-receive system operates similarly to the self-transmitter-self-receiver system but with finely tuned transmit and receive frequencies. All modules and parameters remain unchanged. The key distinction is that transmission and reception occur on separate computers using two 10F-DS devices. A glue stick antenna is connected to the transmitting device (TX2) and the other to the receiving device (RX2). The 10F-DS devices are linked to PC 1 and PC 2 via USB cables. Once the system is powered on and the system program is running on both computers, the chosen audio file can be transferred from PC 1 and played back in real time on PC 2.

4. SYSTEM TESTING AND ANALYSIS

4.1 Waveform analysis

Using an oscilloscope to observe signal waveforms at the transmitter and receiver ends, Figure 3 compares the waveform before CVSD encoding and after CVSD decoding. At the transmitter end, the signal waveform before CVSD encoding appears sinusoidal, and similarly, at the receiver end, the waveform recovered from CVSD decoding also exhibits a sinusoidal shape.

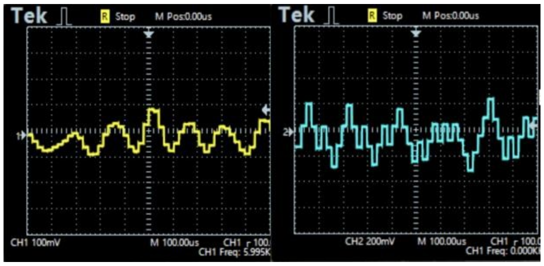

Figure 3. CVSD pre-encoding and post-decoding waveforms.

Figure 4 shows the comparison between the signal waveform before differential encoding and after differential decoding. The waveform before differential encoding is represented as a sequence of rectangular pulses. Post differential decoding, the waveform remains a sequence of rectangular pulses, indicating successful recovery of the signal waveform after differential decoding.

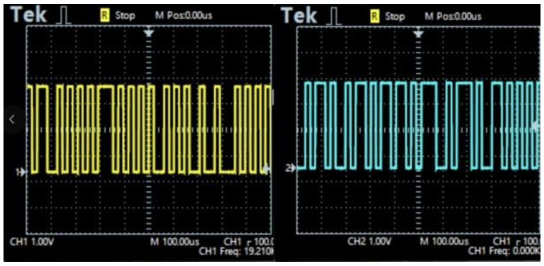

Figure 4. Differential pre-encoding and post-decoding waveforms.

Figure 5 shows the constellation diagram after QPSK modulation, the left side of the figure is a vector form constellation diagram, which shows the vector trajectory corresponding to each symbol in the QPSK modulation process, i.e., the phase change path from one symbol to the next; the right side of the constellation diagram is a point form constellation diagram, in which the four dots represent the four phase states of the QPSK modulation, and each dot is relatively clear and centralized, which indicates that the signal is relatively stable and good signal quality.

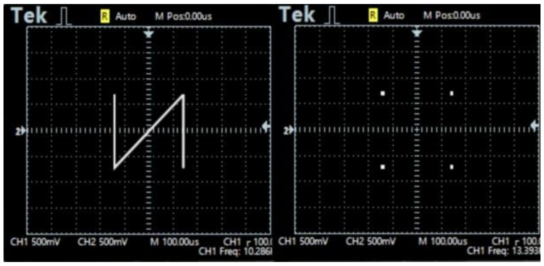

Figure 5. QPSK modulation constellation diagram.

4.2 Different transmission distance effect test

Fix the transmit attenuation of the transmitting device to 10 dB and the receive gain of the receiving device to 50 dB; the position of the transmitting device is fixed and unchanged, and the position of the receiving device is moved to test the audio transmission effect at different distances, and the test results are shown in Table 1. As the distance between the transmitting device and the receiving device becomes farther, the transmission effect of the audio gradually becomes worse; when the distance is in the range of 0-7 meters, the transmission effect of the audio is the best, and when the distance continues to increase, there is a smaller noise and the transmission effect becomes worse.

Table 1. Audio playback effect test at different transmission distances.

Distance/m	Noise situation	Transmission effect
0	No noise	Great
1	No noise	Great
2	No noise	Great
3	No noise	Great
4	No noise	Great
5	No noise	Great
6	No noise	Great
7	No noise	Great
8	Less noise	Mediocre
9	Less noise	Mediocre

4.3 Different receiving gain effect test

The distance between the transmitting device and the receiving device is fixed at 2 meters, and the transmit attenuation is set to 10 dB to test the transmission effect at different receive gains, and the test results are shown in Table 2. The results show that the system has a good transmission effect when the receiving gain is in the range of 50-67 dB.

Table 2. Audio transmission effect at different receiving gains.

Receive gain/dB	Noise situation	Transmission effect
67	No noise	Great
64	No noise	Great
60	No noise	Great
56	No noise	Great
53	No noise	Great
50	No noise	Great
47	Low noise	Mediocre
44	Low noise	Mediocre
41	Low noise	Mediocre
38	Low noise	Mediocre
35	Low noise	Mediocre
30	Low noise	Mediocre
20	High noise	Terrible

4.4 System real-time test

In order to evaluate the real-time performance of the system, the test distance is fixed at 2 meters, the transmit attenuation is set to 10 dB, the receive gain is set to 50 dB, and the transmission time of the whole audio file is measured. Firstly, the simulation scheme is evaluated, which involves only the simulation design of the audio transmission system without any physical hardware, and thus does not need to consider any transmission delay; then the audio file transmission time of the self-transmit-self-receive system and the one-transmit-one-receive system are tested respectively, and the results of the tests are shown in Table 3. The results show that compared with the simulation scheme, there is a delay of 1.9 seconds for the self-transmit-self-receive system and 2.6 seconds for the one-transmit-one-receive system, which both have better real-time performance.

Table 3. System real-time test results.

Category	Audio transfer time/s
Simulation program	20
Self-transmit-self-receive system	21.9
One-transmit-one-receive system	22.6

5. CONCLUSION

In the article, an audio transmission system based on QPSK and OFDM baseband modulation is designed and implemented, and its design is optimized. Through detailed system design and simulation, the effectiveness and real-time performance of the system in audio transmission are verified. The simulation results show that the system can maintain good transmission effect under different transmission conditions, and exhibits small delays of 1.9 s and 2.6 s in both the self-transmit-self-receive and one-transmit-one-receive systems, which satisfy the real-time requirements in practical applications. The research in the article not only verifies the feasibility of QPSK and OFDM based modulation techniques in audio transmission, but also provides a valuable reference for the optimized design of future wireless communication systems.

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