

# The development of transmission demonstration software of speech signal in channel based on MATLAB

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**Abstract:** In view of the concept abstraction in the teaching process of 'Communication Principle' and the difficulty of students understanding of the propagation process of signal in the channel, in order to intuitively show the change state of signal transmission in the channel, this paper takes the classical speech signal as the research object, and combines the key and difficult points in the teaching process to develop and design a set of demonstration experiment software based on MATLAB / App Designer. The article first introduces the basic principle and overall design scheme of speech signal transmission in the channel, and then carries out interface design, algorithm simulation function design and demonstration function design. The software integrates specific demonstration experiments into abstract theoretical teaching, which greatly improves the teaching effect, helps deepen students' understanding of theoretical knowledge, helps teachers reduce the teaching difficulty of course teaching, and improves teaching efficiency.

**Keywords:** Communication Principle, Demonstration Software, MATLAB, speech Signal

## 1. Introduction

'Communication Principle' is a theoretical and practical course[1]. Developing a set of practical demonstration software to assist theoretical teaching can help students deeply understand the basic physical concepts, clarify the basic physical processes, firmly grasp the basic analytical methods and flexibly use theoretical tools to solve practical problems. We take the classic speech signal as the research object, combined with the key and difficult points in the teaching process, developed and designed a set of demonstration experiment software based on MATLAB / App Designer. The software developed by can quickly collect the relevant data generated during the transmission of speech signals and automatically draw their waveforms in real time. Through the visual demonstration process, the key and difficult points of " speech signal transmission in the channel " are vividly displayed, including the sampling theorem, filter design, PCM, etc., which are not easy for students to understand, so as to facilitate students to deeply understand its concept and transmission process.

## 2. Software Design Method

### 2.1. MATLAB Simulation Platform

The name of MATLAB is composed of the first three letters of MATrix and LABoratory[2]. In the 1970 s, Cleve Moler, then director of the Department of Computer Science at the University of New Mexico, designed a set of 'easy-to-use' interfaces for students to call LINPACK and EISPACK matrix software toolkit library programs in order to reduce the burden of students' programming. Today's MATLAB has richer data types and structures, more friendly object-oriented, faster and more sophisticated graphics visualization, more extensive mathematics and data analysis resources, and more application development tools[3]. There are SLMULINK simulation module and MATLAB function in MATLAB communication toolbox, forming a collection of operation function and simulation module, which is used for research, development, system design and simulation in the field of communication. The modules in the communication toolbox can be used directly, and allow modification. It is very convenient to use, so it can fully meet the needs of user design and operation.

## 2.2. Characteristics of Software Technology

This software is based on solid communication theory knowledge, takes students as the center, and encourages students to explore continuously. In the code writing, according to the basic communication principles of A-law 13-line nonlinear PCM coding, BPSK signal modulation, band-pass filtering, BPSK demodulation, low-pass filtering, PCM decoding, sampling decision and so on, the continuous speech signal is discretized. Combined with the appropriate data structure, the computer is used to simulate the relevant data generated by the speech signal in the transmission process of the channel in real time and display it to the students in real time. The developed software is developed based on MATLAB R2020a. The App Designer function is used to modularly develop each interface, and the original \*.mlapp file is designed. On this basis, the \*.mlapp file is compiled by MATLAB Compiler, which can be compiled into a \*.exe file that can be executed independently from the MATLAB environment. The software has the characteristics of human-computer interaction, simple operation and stable operation in the user interface. After the software is opened, only the user clicks on the required function, and the waveform change of the voice signal during the channel transmission can be visualized. The operation of the software is based on the input parameter drive, and the running memory is small. The software is designed based on the object-oriented programming method, which has strong portability and can realize the expansion of functions.

## 3. Software Function Realization

### 3.1. Overall Design Scheme

The software developed by is a system that demonstrates the changing state of the recorded speech signal transmitted in the channel. Its main modules include speech signal processing module, PCM codec module, BPSK modulation and demodulation module, Channel noise module, etc.

The structure of the software system is shown in Figure 1.

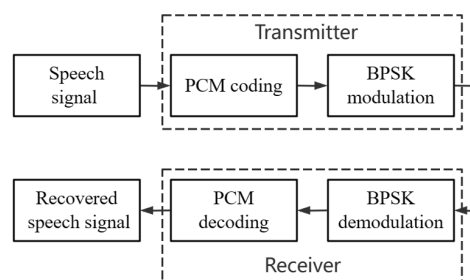


Figure 1: The structure diagram of 'speech signal transmission in the channel' demonstration software system.

### 3.2. Software Module Design

#### 3.2.1. Speech Signal Processing Module

In MATLAB, the audiorecorder function can be used for recording operation, and the speech signal data can be obtained by getaudiodata, and then the speech signal can be processed. The waveform of the original speech signal collected by the experiment is shown in Figure 2.

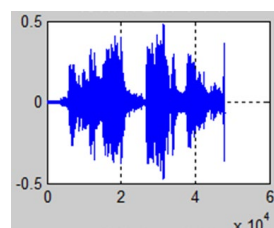


Figure 2: The waveform of the original speech signal collected by the experiment.

#### 3.2.2. PCM Coding

The formation of the PCM signal is realized by the three steps of sampling, quantization and coding.

Sampling refers to the conversion of time-continuous analog speech signals into time-discrete sampling signals according to the sampling theorem[4]. Quantization is to discrete the amplitude of continuous sampling speech signal. Coding refers to the use of binary code groups to represent the quantized sample pulse. Choose 13 broken line coding. The designed software system uses the A-law 13-line non-uniform quantization method. The PCM quantization coding results are shown in the Figure 3.

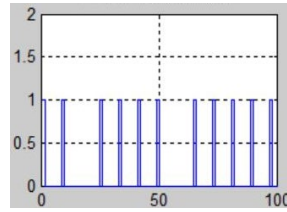


Figure 3: The result diagram of PCM coding for the collected original speech signal.

### 3.2.3. BPSK Modulation

In binary digital modulation, when the phase of the sinusoidal carrier changes discretely with the binary digital baseband signal, a binary phase shift keying (2PSK) signal is generated[5]. Usually,  $0^\circ$  and  $180^\circ$  of the modulated signal carrier are used to represent 1 and 0 of the binary digital baseband signal, respectively. The time domain expression of the binary phase shift keying signal is:

$$e_{2PSK}(t) = \left[ \sum_n a_n g(t - nT_s) \right] \cos W_c t \quad (1)$$

In the formula,  $g(t)$  is a rectangular pulse with pulse width of  $T_s$  and height of 1. When the transmission probability is  $P$ ,  $a_n=1$ , the time domain expression of the binary signal is:

$$e_{2PSK}(t) = \cos W_c t \quad (2)$$

When the transmission probability is  $1-P$ ,  $a_n=-1$ , the time domain expression of the binary signal is:

$$e_{2PSK}(t) = -\cos W_c t \quad (3)$$

When sending binary symbol '0', the modulated signal  $e_{2PSK}(t)$  takes  $0^\circ$  degree phase. When sending binary symbol '1',  $e_{2PSK}(t)$  takes  $180^\circ$  degree phase.

The designed software system uses a relatively simple analog modulation method to modulate the signal. The analog modulation method is shown in Figure 4.

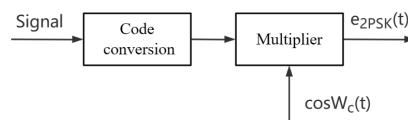
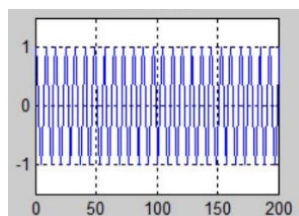
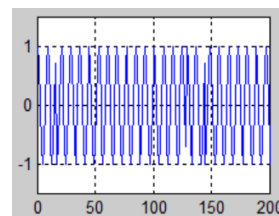


Figure 4: Analog modulation modulation principle diagram.

The high frequency carrier waveform used in the experiment is shown in Fig. 5 (a), and the signal waveform after BPSK modulation is shown in Figure 5 (b).



(a) High frequency carrier waveform.



(b) Signal waveform after BPSK modulation.

Figure 5: Signal waveform in BPSK modulation process.

### 3.2.4. BPSK Demodulation

The BPSK signal belongs to the DSB signal, and its demodulation can no longer use the envelope detection method, only coherent demodulation. The process of coherent demodulation of BPSK signal is actually the process of polarity comparison between the input modulated signal and the local carrier signal, so it is often called polarity comparison demodulation. The demodulation principle of BPSK

signal is shown in Figure 6.

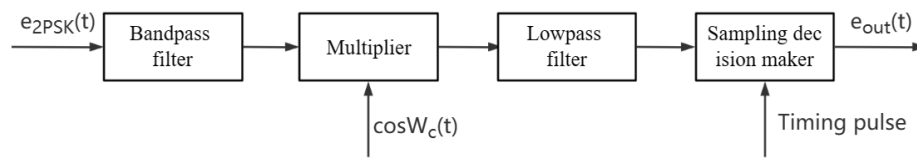


Figure 6: BPSK signal demodulation schematic diagram.

As shown in Figure 6, two filters are used in the designed system, a low-pass filter and a band-pass filter, both of which are designed to be filtered by an elliptical filter. Elliptic filter, also known as Cauer filter, is a kind of filter with ripples in passband and stopband. Compared with other types of filters, elliptical filters have the smallest passband and stopband fluctuations under the same order. The function of the band-pass filter is to filter out the out-of-band noise, and the function of the low-pass filter is to filter out twice the frequency component to obtain the speech signal without PCM decoding.

The waveform of BPSK signal after coherent demodulation is shown in Figure 7.

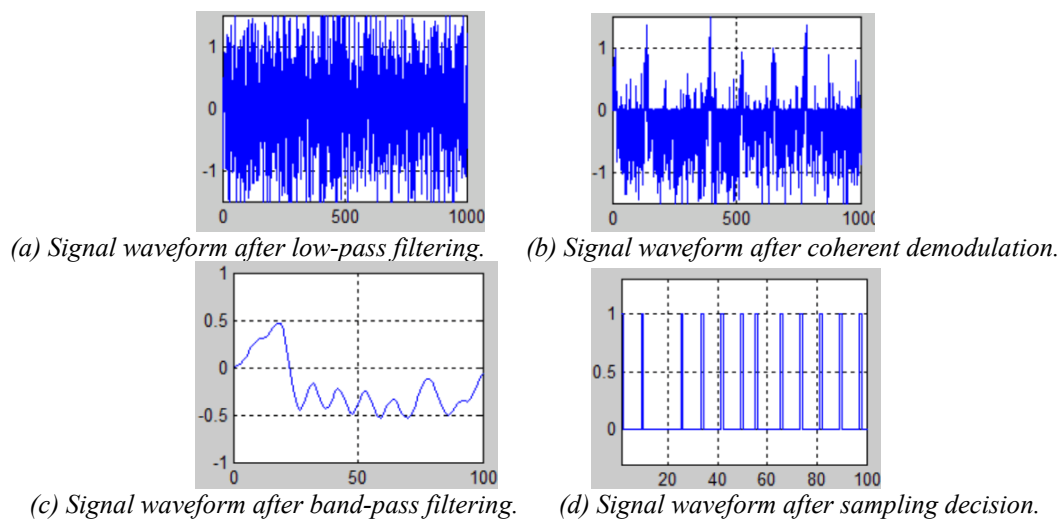


Figure 7: The waveform of BPSK signal after coherent demodulation.

### 3.2.5. Noise Processing

When the speech signal is transmitted in the channel, it will be disturbed by noise. In the simulation experiment, the speech signal is subjected to noise processing. In the experiment, the waveform of the BPSK modulated signal after noise processing is shown in Figure 8.

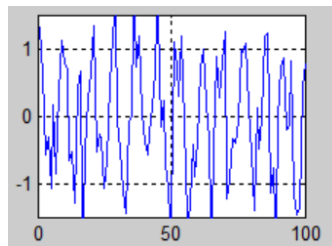


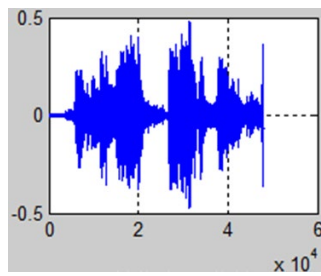
Figure 8: Signal waveform of the BPSK modulated signal after noise processing.

## 4. Experimental results and analysis

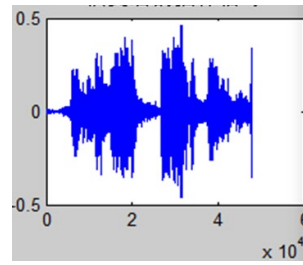
### 4.1. Waveform Comparison

The input speech signal waveform of the designed software should be similar or similar to the speech signal waveform processed by the software. The input speech signal waveform and the recovered speech signal waveform are shown in Figure 9 (a)(b) respectively. It can be seen from the naked eye in Figure 9 that the two are not much different and meet the system functional

requirements[6].



(a) The input speech signal waveform.



(b) The recovered speech signal waveform.

Figure 9: Comparison of speech waveforms before and after software processing.

#### 4.2. System Bit Error Rate Analysis

The generation of error is due to the fact that in signal transmission, the decay changes the voltage of the signal, causing the signal to be destroyed during transmission and generating error. Noise, impulses caused by alternating current or lightning, transmission equipment failures, and other factors can lead to error codes. The bit error rate curve drawn by MATLAB is shown in Figure 10. It can be seen from the bit error rate curve that the design of the software system meets the actual theoretical needs and can express the specific image of the teaching content[7].

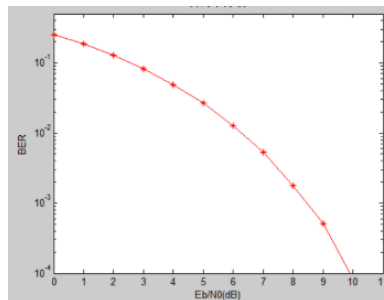


Figure 10: Curve of error rate.

#### 5. Conclusions

The development of "communication principle" course demonstration software based on MATLAB provides a good auxiliary tool for boring theory teaching. The use of demonstration program combines theory teaching and practice demonstration organics, improves the efficiency of the classroom and the interest of students, the design is still in the primary stage, and more functional modules need to be embedded into the system in the future.

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