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Research of pcm coding and decoding system based on simulink

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Abstract. PCM (Pulse Code Modulation) is a common method to convert analog signals to digital signals. Simulink is used to model and simulate PCM communication system then compare the receiving signal waveform output from ideal channel and non-ideal channel separately. The three steps of signal digitization: sampling, quantization and coding are simulated. Through experimental simulation and analysis, it can be seen that Simulink has advantages in communication system modeling and simulation, and it also provides a simple and intuitive method for understanding PCM.

Key words. Pulse Code Modulation, sampling, quantization, simulink.

1. Introduction

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Simulink is a toolkit for dynamic systems modeling simulation and analysis provided by MATLAB. A module specially designed to display the output signal can be used to observe the simulation results in simulation process. Through Simulink's storage module, the simulation data can be conveniently saved into workspace or file. Code can be organized into modules and the modules can be organized into hierarchical subsystems by Simulink[1,2]. Based on the above advantages, Simulink is a generic simulation modeling tool.

PCM is a method of using a set of binary digital code instead of a continuous

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signal to realize digital communication. This kind of communication system has strong anti-interference ability, and is widely used in voice transmission and telemetry system[3,4]. Using Simulink to simulate PCM decoding process, and it can deepen understanding PCM. This kind of method can also be applied to classroom teaching and experimental teaching.

2. PCM coding principle

Pulse code modulation is the process of transferring a continuous analog signal into a discrete digital signal. It is completed by sampling, quantifying and coding.The receiver receives the signal through decoding and a low pass filter[5].

2.1. Sampling

The so-called sampling is the periodic scanning of the analog signals, and the continuous signals in time domain become discrete signals in time domain. After sampling, the analog signal should also contain all information of the original signal, which means that the original analog signal can be restored without distortion.

The lower-limit of sampling rate is determined by the sampling theorem.Set the highest frequency is less than $f_h$of a continuous analog signal.The condition of signal recovery is that the sampling frequency f_s is not lower than $2f_h$ That is:

$$f_s \geq 2f_h \tag{1}$$

2.2. Compression and quantification

The analog signal is sampled to be a discrete signal in time domain, but it is still an analog signal. It has to be quantified to become a digital signal. Quantification is the process of converting the analog signal of amplitude continuous into a digital signal represented by a finite bit binary number.

The quantization is divided into uniform quantization and non-uniform quantization. In practice, non-uniform quantization is often used. The quantization interval is small for small signal. Instead, the quantization interval is large for large signal[6,7]. In the United States, u-law compression is adopted, but A-law compression is adopted in both China and Europe. Therefore, PCM adopts A-law compression[8].A-law compression formula is as following:

$$y = \begin{cases} \frac{Ax}{1+\ln A} & 0 < x \leq \frac{1}{A} \\ \\ \frac{1+\ln Ax}{1+\ln A} & \frac{1}{A} < x \leq 1 \end{cases} \tag{2}$$

x is the normalized input voltage of the compressor and y is the nomalized output voltage of compressor. A is a constant and it determines the degree of compression. In practical terms, A is 87.6.In practice, the compression-expansion characteristics of 13 fold lines (A=87.6) are usually approximated by A-law compression. The PCM encoding used in this design takes use of this compression-expansion feature.

2.3. Coding

The so-called coding is to transform the quantized signal into code, and the reverse process is called decoding. In the realization of circuit, the quantizer and encoder often constitute an inseparable coding circuit, which has different implementation schemes, and the most commonly used one is the successive comparison method. 8 bit PCM coding is usually used to ensure satisfactory communication quality in voice communication.

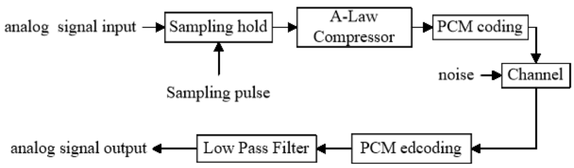

Fig. 1. PCM principle diagram

3. Coding and decoding simulation system

PCM coding and decoding system based on Simulink adopts sinusoidal signal source, and includes three parts, namely encoder, channel, decoder.First, sampling signal takes uses of zero order holder, and then signal is quantized by A-law compression and quantizer, and then it is coded.The encoded digital signals are transmitted by binary channel and decoded by the receiving terminal. The decoding process is the opposite of coding process.Decoding signal passes through low pass filter. The receiving signal waveform is obtained and compared with the original input waveform.

A scope module is used to display the output signal during the experiment in Simulink. The key output signals in this communication system include sampling signal, compressing and quantizing signal, encoding signal, decoding signal and filtering signal.

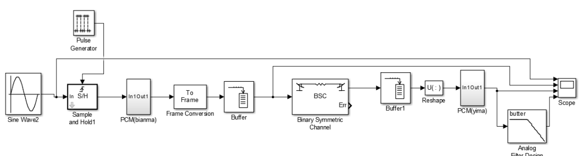

Fig. 2. Simulation diagram based on simulink

3.1. Sampling module

Periodic sampling can turn time and amplitude continuous signal into amplitude continuous and time discrete signal. Then sampling voltage is maintained for a certain time through a sample and holed circuit.According to the sampling theorem, the sampling frequency should not be less than double highest frequency of the analog signal. The frequency of the simulated sinusoidal signal is 2* Pi, that is, the period is 0.1592 seconds. If the uniform sampling theorem is established, the sampling interval must be less than 0.08 seconds, otherwise the signal will be difficult to recover and the distortion will occur. The sampling interval here is 0.04 seconds.

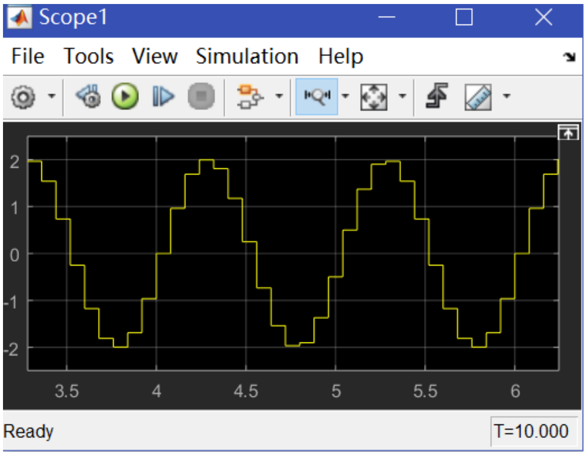

Fig. 3. Sample and hold circuit and sampling signal

The amplitude of sinusoidal signal is 2v, and the waveform of figure 3 can be obtained after the pulse sampling.The pulse waveform can be found to be flat top because of the hold holding circuit, i.e., the flat top sampling. If the sampling interval is changed to 0.12 seconds, which is greater than the Nyquist sampling interval, the signal cannot be recovered.

3.2. Compression quantification module

The instantaneous value of the sampling signal is discrete by quantization compression algorithm , i.e. the sampling value is replaced with the most suitable level . Quantization is divided into uniform quantization and non-uniform quantization, and the system adopts non-uniform quantification to improve signal to noise ratio effectively. Quantization interval is 1 in this system.

The right image is the compressed signal after gain module, and the lower right signal is normalized. Obliviously, compressed signal is quantified into corresponding intervals. There are 128 quantified intervals. A-law compressor is used in this system, and A is 87.6.

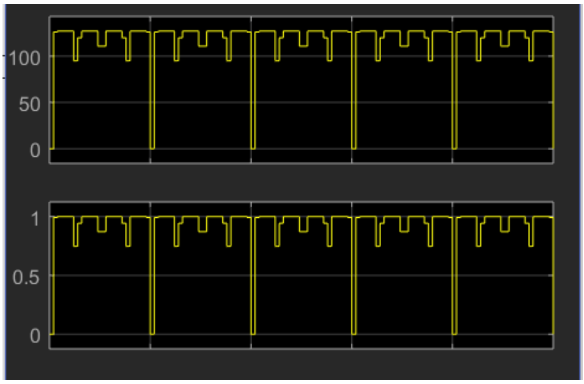

Fig. 4. Compression quantification module and signal waveform of equalized-compressed

3.3. Coding module subsystem

The quantitative value of fixed level is encoded in binary code group. There are 8-bit coding including the first polarity bit. The high level is 1 and the low level is 0.Upper right is a encoded waveform, while the lower right is a waveform formed by "framing" and "buffer" in figure5. This method makes signal strong in the link generation error code or network jitter and sudden transmission.

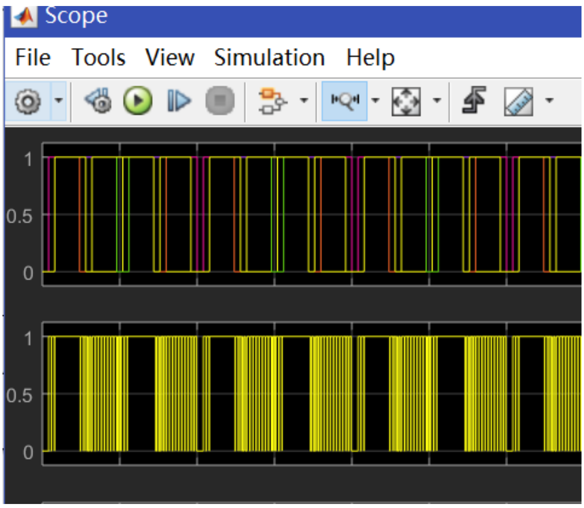

Fig. 5. Coding subsystem and PCM codes

Specification of key modules in the coding module subsystem as follows.

Saturation: Limit input signal to the upper and lower saturation values.Amplitude of output signal is limited into (-1,1) in the system,

Relay Output the specified on or off value by comparing the input to the specified thresholds. The on/off state of the relay is not affected by input between the upper and lower limits. If input signal is positive and it will outputs and otherwise outputs 0.

A-law Compressor Compress the input signal using A-law compression.This block processes each element independently.A is 87.6.

Gain: Element-wise gain (y = K.*u) or matrix gain (y = K*u or y = u*K). Grain is 127 in the system.

Quantizer Discretize input at given interval. Discretize interval is one quantization unit.

Integer to Bit Converter: Map a vector of integer-values inputs to a vector of bits. For fixed-point inputs, the stored integer value is used.Since the maximum value of the quantization value is 127, the module parameter is set 7.

Finally, a multiplexer is used to multiplexing the extreme values pulse and the numerical pulse.

3.4. A-law Expander System

The sub-module of decoding module is basically the inverse transformation of the encoding module. The coded signals output form binary symmetric channel are separated into a polarity pulse and the seven numerical pulse by separator. Then a multiplexer is used to combine the seven numerical pulses into the position converter. Then, through the gain module and A- law expansion in turn.

A-law expansion signal is multiplied by the polarity pulse, and then the decoding filter signal is obtained by low pass filtering. The right signal with noise is A-law expansion signal of non-ideal channel and the left signal is A-law expansion signal of ideal channel as shows in figure 6.When error probability of binary symmetric channel is set to 0.05, the decoding waveform is obviously chaotic. Therefore, the noise of the channel has a great influence on the result of decoding.

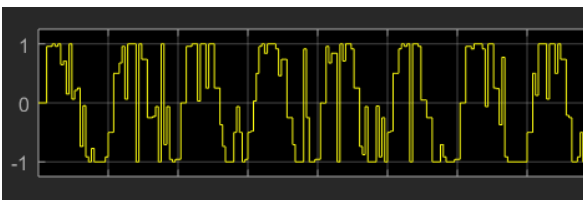

Fig. 6. Decoding module and decoding signal waveform

3.5. Decoding filter signal

Compared to the sending signal, the receiver signal has a delay.The right signal with noise is output of non-ideal channel and left signal is output of ideal channel as shows in figure 7.The receiving signal waveform is distorted when the channel is noisy.

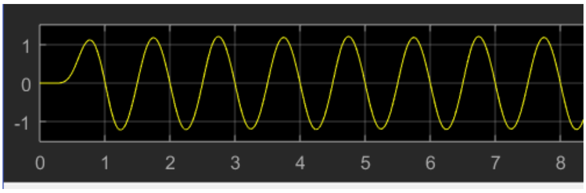

Fig. 7. Filter module and signal waveform with ideal channel and non-ideal channel

3.6. System performance analysis

PCM system involves two kinds of noise: quantization noise and channel additive noise. Quantization noise due to the system using broken line approximation of continuous smooth curve,meanwhile quantification using finite values instead of infinite continues values. Therefore quantization noise can be reduced only by increasing the quantitative series , but can not be eliminated. For the influence of additive noise, the simulation waveform can be observed by adjusting the error rate of the channel. The additive noise can not be eliminated.

4. Summary

System simulates the PCM coding and decoding process. In the simulation process, sampling, quantizing, coding, decoding and filtering signals are observed, and the unification of theory and practice is realized. Students' understanding is deepened and good results will be received in teaching and experiment.

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