Real-Time Application of DPCM and ADM Systems

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Abstract: Digital communication techniques have proved their preference over analog communication techniques due to their higher reliability, flexibility and compatibility. However, the commonly used digital communication techniques such as PCM (pulse coded modulation) and LDM (linear delta modulation) cause quantization error, slope overload distortion, and granular noise which all negatively affect the communication process. In order to solve the aforementioned problems, ADM (adaptive delta modulation) and DPCM (differential pulse coded modulation) are discussed and implemented. DPCM system solves the quantization error problem, and ADM solves the slope overload distortion and the granular noise problems. To implement these two systems, Simulink (The Math Works, INC., Natick, MA, USA) is used on a multithreaded processor computer; moreover, they are tested in real-time and subjected to different kinds of noise.

Key words: Adaptive delta modulation, differential pulse coded modulation, pulse coded modulation, linear delta modulation.

1. Introduction

Communication is based on transmitting a message from a transmitter to a receiver using an appropriate channel. Message should be transformed into a form that is compatible with the channel; thus, they are modulated using one of several modulation techniques [1].

Communication can be analog, where a continuously varying information signal is transmitted, or digital in which discrete message is transmitted after being represented either by mean of line codes, or by limited set of continuously varying form using the digital modulation methods. Channels used by digital communication are either point-to-point or point-to-multipoint, examples of which are copper wires, optical fibers, and wireless communications media [2].

PCM and LDM are the common techniques for digital modulation process to make transmitting data feasible. However, the first technique provides the quantization error problem, and the second provides the granular noise and the slope overload distortion problems.

Thus, the DPCM (differential pulse coded modulation) and the ADM (adaptive delta modulation) are implemented to solve the aforementioned problems; hence, the relevance of this work.

2. Background Information

Digital communication systems use digital sequence, made up of elements from a finite alphabet, as an interface between the source and the channel’s input, and between the channel’s output and the final destination [3].

Digital communication previously relied on two techniques; the Pulse Coded Modulation and the Linear Delta Modulation. Although these two systems provide many advantages over analog systems, they also have problems such as the quantization error resulting from PCM, and the granular noise and slope overload distortion resulting from LDM. These systems are implemented and tested using Simulink of Matlab [4].
The following is a discussion of PCM and LDM systems.

2.1 Pulse Coded Modulation

The two basic operations in the conversion of an analog signal into a digital signal are time discretization and amplitude discretization. In order to accomplish these operations, PCM uses sampling and quantization respectively; moreover, PCM involves the conversion of quantized amplitudes into a sequence of simpler pulse patterns (usually binary) called code words [5].

The signals in PCM are binary; that is, there are only two possible states, represented by logic 1 (high) and logic 0 (low). This is true no matter how complex the analog signal is. The first step in PCM is to sample the analog signal’s amplitude at regular time intervals, on condition that the sampling rate is several times higher than that of the maximum frequency of the analog signal. Second, the sampled signal is quantized, meaning that the instantaneous amplitude of the analog signal at each sampling instant is rounded off to the nearest of several predetermined levels [6].

The number of levels is always a power of 2; thus, the output of pulse code modulator is a series of binary numbers, each represented by some power of 2 bits. At the receiver stage, the pulse coded demodulator converts the binary numbers back into pulses with the same levels as those in the modulator; these pulses are then processed to restore the original analog signal [6].

The performance of a PCM system is influenced by two major sources of noise, namely the channel noise, which is introduced anywhere between the transmitter and the receiver, and the quantization noise, which is introduced in the transmitter and is carried all the way to the receiver output. The noise is a dependent signal in the sense that it disappears when the message signal is switched off [7].

2.2 Linear Delta Modulation

Linear Delta Modulation basically consists of a two-level quantizer and a feedback path containing a single integrator. A sampler can be included in the quantizer or before the subtractor. After sampling, the quantizer produces a pulse of uniform duration and amplitude at each sampling instant. If the quantizer input is positive, the pulse is of positive polarity; otherwise, the pulse is of negative polarity. Consequently, the sequence of binary pulses produced by the quantizer is transmitted through the digital channel to reach the decoder. The decoder consists of an integrator, which is identical to that in the encoder, and a low pass filter with the same bandwidth of the original input signal [8].

Delta modulation is a special type of analog to digital quantizer, which is applicable to smoothly varying analog signals where there is a strong correlation between one sample and the next. DM is the simplest form of differential pulse coded modulation where the difference between successive samples is encoded into n-bits data stream; in Delta Modulation, the transmitted data is reduced to 1-bit data stream [9].

In Delta modulation, the quantization noise lies in two forms: the granular noise and the slope overload distortion. Granular noise is due to the continuous signal being forced to assume discrete values that are multiples of the quantizer step size, so it is similar to PCM quantization noise. It is directly related to the step size; if the step size increases, then the granular noise increases as well. However, the slope overload distortion is caused by the use of a step size delta which is too small to follow portions of the waveform that has a steep slope. Thus, the slope overload distortion is inversely related to the step size: if the step size increases, then the slope overload distortion decreases [8].

To achieve a high SNR (signal to noise ratio), Delta Modulation uses oversampling techniques, that is, the analog signal is sampled at a rate several time higher than that of the Nyquist rate; this sampling is known as the Nyquist Criteria in digital signal processing [9].
3. Proposed Solutions

Digital communication, as mentioned, relied on two techniques, the PCM and the LDM which proved, to a certain extent, their importance and ability to improve the transmission process. Later on, these two techniques faced additional problems which endangered the mentioned transmission and motivated the communication engineers to work hard to find new methods to solve the resulted problems. Consequently, new techniques were developed: the ADM and the DPCM.

3.1 DPCM (differential pulse coded modulation)

Differential Pulse Coded Modulation basically converts an analog signal into a digital signal by first sampling the analog signal and then quantizing the difference between the actual value and its predicted value; finally the quantized difference is encoded to form the digital value. Thus, DPCM code words represent difference between samples unlike PCM where code words represent a sample value [10].

Differential Pulse Coded Modulation reduces the error generated by quantization process (known as "quantization error") at the transmitter of the PCM system; this is achieved by taking a difference relative to the output of a local model of the decoder process instead of taking a difference relative to the previous input sample. In this new technique, the difference can be quantized, securing a good way to incorporate a controlled loss in the encoding. The DPCM transmitter is similar to the PCM transmitter, but it has a prediction filter for prediction of the future values of the signal; consequently, eliminating the quantization error [11].

Concerning SNR, it is much improved in DPCM over PCM; thus, allowing much better noise filtering with less bandwidth. For example, if there were two signals $m(t)$ and $d(t)$ with $M$ and $D$ being their peak amplitudes respectively, and if the same quantization level $L$ is used to sample both signals, the quantization step $\Delta v$ in DPCM is reduced by a factor of $\frac{D}{M}$ and the corresponding quantization noise power is $\frac{(\Delta v)^2}{2}$.

In DPCM, the quantization error is reduced by the factor $\left(\frac{M}{D}\right)^2$; since the SNR is inversely proportional to noise, it increases by the same factor.

In practice, the SNR improvement may be as high as 25 dB in such cases as short-term voiced speech spectra. Alternatively for the same SNR, the bit rate for DPCM could be lower than PCM by 3 to 4 bits per sample; hence, reducing the bandwidth.

3.2 Adaptive Delta Modulation

ADM is a modified DM (delta modulation) in which the step size is not fixed; it differs according to the input signal, using what is known as the "Adaptive Algorithm". When several consecutive bits have the same direction value, the encoder and decoder assume that slope overload is occurring, and the step size becomes progressively larger. Otherwise, the step size would become gradually smaller over time [12].

The ADM system is the system that reduces the granular noise and the slope overload distortion resulting from Linear Delta Modulation; the reduction is achieved due to the presence of the adaptive filter or algorithm in the system [13].

The algorithm uses a gradient descent to estimate a time varying signal. The gradient descent method finds a minimum, if it exists, by taking steps in the direction negative of the gradient. It does so by adjusting the filter's coefficients so as to minimize the error. The gradient is the del operator (partial derivative) and it is applied to find the divergence of a function, which is the error with respect to the nth coefficient in this case. The LMS algorithm approaches the minimum of a function to minimize error by taking the negative gradient of the function [13].

In the following sections, the systems discussed are implemented in Simulink (The Math works, Inc., Natick, MA, USA) to test and observe the results prior to reaching the real time application of the DPCM and
ADM systems.

4. Simulation and Results

In order to prove the importance of the DPCM and ADM systems in solving the aforementioned problems, these systems were implemented and simulated using Simulink.

4.1 Simulink Implementation of Pulse Coded Modulation and Linear Delta Modulation

The implementation of PCM in Simulink, as shown in “Fig. 1”, requires the following: a sine wave source block that provides the test signal; a Bessel low pass filter of the 8th order to limit the signal’s frequency and prevent aliasing error; a Zero-Order-Hold block to allow the sampling process, thus changing the signal from a continuous varying signal to a discrete time signal; a quantizer for the quantization process which approximates the discrete values to levels; a uniform encoder to encode the obtained levels to a bit data stream; and, a decoder followed by a reconstruction filter in order to reconstruct the original signal [3].

The implementation of LDM in Simulink, as shown in “Fig. 2”, requires the following blocks: a sine wave source block to provide the test signal; a Zero-Order-Hold block to act as a sampler for the sampling process, thus changing the continuous varying signal to a discrete time signal; a 1-bit quantizer for the quantization process; a unit delay; and an encoder for the encoding process. On the receiver side, the receiver consists of a decoder followed by Butterworth Filter of the 8th order for the reconstruction of the original signal [3].

The addition of noise to the transmission channel of the previous systems makes the reconstruction of the original signal a difficult task and the reconstructed signal would be completely distorted; these systems were implemented in order to compare their results with the results resulting from ADM and DPCM systems.

4.2 Simulink Implementation of Differential Pulse Coded Modulation

The Simulink implementation of the DPCM, shown in Fig. 3, consists of the following blocks: a sine wave source block to provide the test signal; a quantizer for the quantization process; a differentiator filter that acts as the prediction filter discussed previously; and a uniform encoder for the encoding process; and at the end, a low pass filter of type Bessel for filtering and reconstructing the original signal.

In order to study the effect of noise on the DPCM system, band limited White Gaussian noise was added to the transmission channel between the transmitter and the receiver; the results of the simulation are as shown in Fig. 4.

It’s clear from the previous results that the Differential Pulse Coded Modulation system succeeded, to a certain extent, in reconstructing of the original signal, even after the addition of noise to the transmission channel; this is due to the usage of the prediction filter, which predicts the future values of the signal; hence, preventing the quantization error.

Another approach can be used in order to reduce the quantization error resulting from the PCM system, this
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Fig. 2  Simulink implementation of LDM system.

Fig. 3  Simulink implementation of DPCM.

Fig. 4  Original vs. reconstructed signal of DPCM system.
approach is achieved by feeding back the quantization error to the quantizer through a special noise filter such as the differentiator filter used in the DPCM system.

4.3 Simulink Implementation of Adaptive Delta Modulation

The Simulink implementation of ADM, shown in Fig. 5, consists of the following blocks: a sine wave source to provide the test signal; a quantizer for the quantization process; and a LMS filter with the LMS algorithm chosen to act as the adaptive filter with the adaptive algorithm, driving the input signal to the desired signal. On the receiver side, the same LMS filter with the same algorithm is used in order to get the successful reconstruction of the original signal, and to eliminate the errors resulting from the LDM.

In order to study the effect of the noise on the transmission channel, band limited White Gaussian noise was added to the channel between the transmitter and the receiver; the results of the simulation are as shown in Fig. 6.

The results shown in Figure 6 prove that the reconstructed signal was approximately the same as that of the original signal, which means that the Adaptive Delta Modulation system has successfully reconstructed the original signal without being affected by the noise that was added to the channel. The difference in amplitude is due to the LMS filter which acted as the adaptive algorithm.

Therefore, the ADM system has eliminated the granular noise and slope overload distortion resulting from the LDM.

Upon the completion of the simulation part, two of these systems were implemented: the DPCM and the ADM. These two techniques were chosen since they have produced the best results with the highest accuracy when simulated using Matlab. To implement these given Simulink systems, an important issue is to be taken into consideration: In the beginning, a certain interface or methodology is to be chosen, studied and adapted to implement the previously simulated systems as a real-time application. As the term means, a real-time application must be done using an interface to convert the programming done on a PC to a certain language or algorithm that can be read, sent and received by a given hardware. For example, the observation of the circuits on Simulink and the obtained results must be verified and checked for similar observations when talking real-time application.

Fig. 5 Simulink Implementation of ADM.
Real Time Application and Results

Based on the results of the previous simulations, the Adaptive Delta Modulation System and the Differential Pulse Coded Modulation System succeeded in the reconstruction process of the original signal; therefore, it’s time to implement these two systems in a real-time application in order to prove the importance and the advantages of these two systems when in real-time, and to study the effect of natural and real noise on the transmission channel of the signal coming from the transmitter part of the system to the receiver part.

The idea behind the real-time application shown in Figs. 7 and 8, relies on the following: the simulation of the transmitter part of each of the previously mentioned systems occurs in Simulink; and, the resulting data, which is either the values of an ADM signal or the values of a DPCM signal, are stored in the workspace of Matlab as a structure using the block named “To Workspace” in the Simulink library. After that, a certain code is written in the workspace to establish a certain communication between the Matlab software and the parallel port of the computer; after which, a certain code is executed in the workspace in order to send the data stored as a structure to the parallel port where they are received as an eight-bit binary format [14].

The (8-bit) binary data coming from the parallel port “DB-25” (from pin 2 to pin 9) is fed to a PIC 16C716 microcontroller where they are manipulated and provided to a 433.92 MHz ASK transmitter, which transmits them in a wireless media to the 433.92 MHz receiver. At the receiver’s end, the receiver gets the data and feeds them to another PIC 16C716 microcontroller which prepares the data and provides them as 8-bits that are observed and compared with the sent data from the parallel port on the transmitter side [15].

The microcontrollers at the transmitter side and that used at the receiver side were used in order to ensure a three-way handshaking, and to ensure the transmission of data. The microcontroller at the transmitter sends,
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Fig. 7 Real-Time implementation.

Fig. 8 Process of the real-time implementation.

using the 433.92 MHz ASK transmitter a series of 8-bits of “1” (11111111) to initiate the communication process between the transmitter circuit and the receiver circuit. Afterwards, it sends a bit-stream “10101010” as a start byte; then, it sends the byte to be transmitted (ADM or DPCM data); and, ends by a third byte of a bit-stream “01010101”. In contrast, the microcontroller at the receiver’s side, receives the bytes simultaneously using the 433.92 MHz receiver; so, when receiving the start byte and the end byte, it knows that the data required is the second received byte and provides it as an 8 bits output.

To test the real-time application, certain types of noise were introduced to the transmission channel such as the CDMA and GSM Jamming signals, very high speed wind and electrical interference; not to mention the introduction of certain physical obstacles to the transmission channel. The ADM and DPCM systems proved their strength against such type of noise and interferences.

After the implementation of the real-time application of the ADM and DPCM systems and testing them, the results were successful, and the transmitted signal was successfully received without any error or disturbance. This fact was proved by a comparison done between the transmitted data from the transmitter’s side and the received data at the receiver’s side.

The obtained results prove the importance of the two systems, the ADM and the DPCM, in eliminating the effect of noise on the transmission channel, and in making the reconstruction of the original signal easier than when implementing the commonly used modulation techniques.

6. Conclusions and Future Work

This project proves the importance of certain modulation techniques such as the Adaptive Delta Modulation and the Differential Pulse Coded Modulation in some environments and in solving the errors resulting from commonly used modulation techniques, the Pulse Coded Modulation and Linear Delta Modulation, such as granular noise and slope overload distortion.

Working in real-time is important because the conditions needed for the experiments are better than just testing them using a software such as Simulink; working in real-time enhances the engineer’s knowledge and experience, encouraging him/her to better think; it also helps the engineer to figure out all the problems and try his/her best to solve them not relying on a software in finding the solution.

In this experiment, a synchronization problem occurred when trying to feed the data back to the parallel port of a receiving computer to reanalyze the received data. This error is due to the fact that the transmitter part has sent the data in a rate faster than that of the receiver, leading to a data loss at the receiver. Therefore, all future work will concentrate on solving
this synchronization problem either by applying an addition circuitry which would act as a buffer for the data, or by applying a certain algorithm to act as a “congestion control algorithm”, such as the “Leacky Bucket”.

References