Adaptive Delta Modulation Techniques

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ABSTRACT
Delta modulation is a waveform coding technique which reduces the data rate to a larger extent in data communication; the problem encountered in delta modulation is the slope over load error, which is inherent in the system. In order for the signal to have good fidelity, the slope-overload error needs to be as small as possible. Hence there is need for adaptive techniques to be applied to Delta Modulation to reduce the noise. Adaptive delta modulation reduces the slope over load error to a greater extent. ADM attempts to increase the dynamic range and the tracking capabilities of fixed step-size delta modulation. The adaptive algorithms adjust the step size (from a range of step sizes) to the power level of the signal and thus enhance the dynamic range of the coding system appreciably. This paper discusses several Adaptive Delta Modulation techniques for improving the signal-to-noise ratio (SNR) of Adaptive Delta Modulators (ADM) and also their performance comparison is made.

General Terms
Digital Communication, Signal Processing

Keywords
Delta Modulation, Adaptive Delta Modulation, Analog to Digital Conversion (ADC), Sampling rate, Speech coding techniques

1. INTRODUCTION
All the recent wired and wireless technologies have already shifted to digital. Hence digital storage and digital coding techniques have a paramount consideration in the design of any communication system. Basically in speech communication, the capabilities of the encoders and decoders play an Important role in the quality of the signal. Hence the Analog to Digital Converters and Digital to Analog Converters at encoding and decoding side respectively has to be designed with the required quality of service and the resource availability. For a human speech, sampling rate of 8000 samples per second as the nyquist rate has to be used and for a Pulse Code Modulation (PCM) with 256 quantization levels, minimum of 64 Kbits per seconds of data rate is required. This becomes quite a bit of overhead for the commonly used low cost and low power handheld devices. Thus engineers previously used the human ear sensitivity to reduce the number of bits required for transmission. But the dynamic range of human ear was about 40dB. The coding could not provide good quality of speech where whispering kind of speech are present along with high amplitude signals. [3]

A new kind of quantization called non uniform quantization was introduced which could take care of the dynamic variations in the amplitude. This was introduced in the Public Switched Telephone Network (PSTN) where it is called μ-law Pulse Code Modulation (PCM). Later the engineers recognized the slight correlation between the samples of the speech and found another kind of coding called Differential PCM where the difference of the successive samples is calculated, coded and transmitted. This was helpful to reduce the data rate required to some extent. Further bandwidth was tried to reduce using complex algorithms and Adaptive predictors are used to form Adaptive Differential PCM (ADPCM), this could reduce the data rate requirements at the cost of increased complexity in implementation. Later the techniques that can send one bit per sample are devised and are termed as Delta Modulation (DM) where typically, data rate is equal to the sampling rate. [1][2]

The layout of the paper is as follows. Section 2 describes the basics of Delta Modulation technique. Section 3 describes Adaptive Delta Modulation and four different types of ADM. Section 4 gives performance comparison and section 5 is the conclusion.

2. DELTA MODULATION (DM)
DM as in [1] [2] is a differential quantization scheme that uses two levels of quantization (i.e., one bit quantizer). By using a single bit to represent each sample, the sample rate and the bit rate are equivalent. Consequently, sample rate is directly related to signal quality (SNR). Also, BW of the input signal and band-limiting of the output are significant factors in determining signal quality.

Applying adaptive techniques to a DM quantizer allows for continuous step size adjustment. By adjusting the quantization step size, the coder is able to represent low amplitude signals with greater accuracy (where it is needed) without sacrificing performance on large amplitude signals.

Difference between successive samples is done as $d(n)=x(n)-x(n-1)$. When the $d(n)$ exceeds zero it outputs a logic one, when the input is less than zero, the output is a logic zero. Hence, the output is a single bit indicating the sign of the magnitude of $d(n)$.

Δ is the step size parameter. The value of Δ plays an important role in the performance of DM. If is relatively small, tracking of slowly changing, low amplitude signals is quite good at the expense of poor tracking for fast, abruptly changing signals. When DM is not able to keep up with the input signal a phenomenon called slope overload is exhibited (as shown in Figure 1).[3]

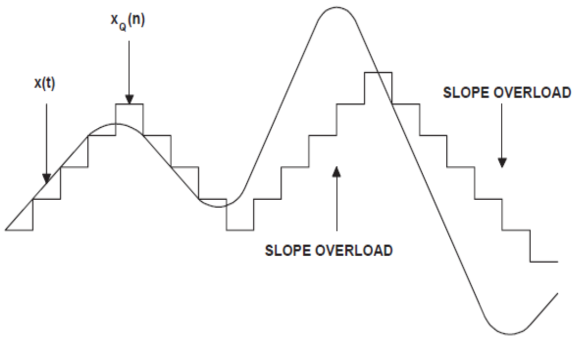

Figure 1: Slope overload in DM.

Increasing the value of Δ can lessen the effects of slope overload but creates a new problem; granular noise. With too large, low amplitude signals will not be quantized at fine enough levels and they appear as idle channel noise (see Figure 2).[3] The idle channel pattern is simply an alternating one-zero sequence indicating the input signal amplitude is not changing. Since an alternating one-zero bit patterns has a mean value of zero, the signal out of the decoder will integrate to zero.

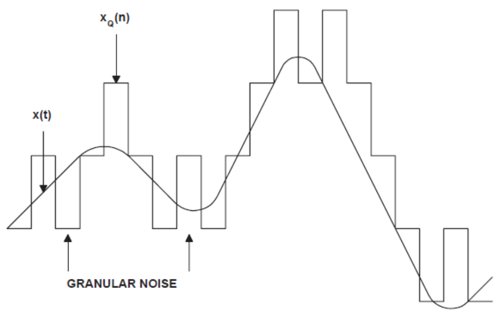

Figure 2: Granular noise in DM.

3. ADAPTIVE DELTA MODULATION (ADM)

Adaptive delta modulation (ADM) reduces the slope-overload and granular distortions encountered in delta modulators (DM) to a greater extent. In each of the step-size algorithms of ADM, the processor detects the pattern to see if the delta modulator is operating in the granular noise region, in which case it produces an alternating ……1010….. Pattern, or in the slope over load region in which case it produces an all-1 or all-0 pattern.

- If the ADM senses a …1010…. pattern, it decreases the step-size, and
- If it senses …..1111…. or ….0000…. pattern, it increases the step-size.

Different step-size adaptation algorithms change the rate of change of step-size in different ways.

This work was first done by Jayant [4]. A good amount of work has been done in this area using SONG and modified ABATE step-size algorithms. [5] [6]

In Jayant ADM, an increase in step size is achieved by multiplication with a constant and decreases it by dividing by the same constant. SONG algorithm will add or subtract a constant to step size with 11… or 00… pattern respectively. Modified ABATE works similarly with limiting step size to a value. In other new algorithm, both multiplicative and addition constants are made use so that slope-overload noise is reduced.

3.1. Adaptive Delta Modulation with one bit memory:

In this method, a delta modulator which, at every sampling instant, adapts its step size Δ(for staircase approximation to the input signal) on the basis of a comparison between the two last channel symbols, Cr and Cr-1. Specifically, the ratio of the modified step size Δr to previous step size Δr-1 is either +P or –Q depending on whether Cr and Cr-1 are equal or not. A simulation of the delta modulator with a band-limited speech input has revealed that PQ=1 and PQ~1.5 represent optimal adaptation characteristics.

An important disadvantage of this technique is that the dynamic range of modulated signal increases.

3.2. SONG Algorithm

Let m(t) be the input signal and be its staircase approximation. Let error, at the kth sampling instant. k = 0, 1, 2, 3 . . . e (k) can be of positive or negative value. The kth transmitted symbol is '1' if e(k) > 0, otherwise it is '0' if e(k) < 0.

If e(k) = 0 , either '1' or '0' can be transmitted.

The SONG algorithm used by NASA produces the step-size S(k+1) which minimizes the mean-square error between m(t) and m^(t). In the implementation of the SONG system ±5 V was the maximum signal level and the minimum step-size was So = 10 mV. Here we see that as long as e(k) is of the same sign as e(k - 1) the magnitude of the new step-size S(k+1) will exceed the magnitude of the old step-size S(k) by So, the minimum step-size. However, if e(k) and e(k-1) differ in sign, the magnitude of S(k+1) will be less than the magnitude of S(k) by the amount So. The algorithm can also be written in terms of the following equation (1).

$$|s(K+1)| = \begin{Bmatrix} |s(k)| + s_o \quad if\ e(k) = e(k-1) \\ |s(k)| - s_o \quad if\ e(k) != e(k-1) \end{Bmatrix} \quad (1)$$

3.3. Modified ABATE algorithm:

The modified ABATE algorithm is another step-size adaptation algorithm and is more susceptible to slope overload than the SONG algorithm. The unique feature of this algorithm is that it is designed to adaptively follow the received signal even in a channel with high error rate. The equation describing the modified ABATE algorithm is (2).

$$|s(K+1)| = \begin{cases} (|s(k)| + s_o) + s_0 \cdot e(k) & if\ e(k) = e(k-1) \\ & and\ s(k) < 8s_0 \\ s(k) \cdot e(k) & if\ e(k) = e(k-1)\ and \\ & s(k) = 8s_0 \\ s_0 \cdot e(k) & otherwise \end{cases} \quad (2)$$

When an error occurs in the received data stream the step size processor will produce erroneous step sizes until a correctly received data transition is detected. The average number of erroneous step sizes following a received error in the modified ABATE algorithm is less than other ADM algorithms apart from SONG algorithm.

3.4. Modified step size Song algorithm:

The modified SONG technique for the step-size adaptation is described as

$$|s(K+1)| = \begin{cases} (\alpha|s(k)|+s_o)e(k) & if\ e(k)=e(k-1) \\ (\beta|s(k)|-s_o)e(k) & if\ e(k)\neq e(k-1) \\ & and\ \beta\cdot s(k)s_0 \\ s_0\cdot e(k) & if\ e(k)\neq e(k-1)\ and\ \beta\cdot s(k)<s_0 \end{cases}$$

(3)

α is the adaptation parameter nearly equal to 1 but, greater than 1.β=1/α.

In this algorithm the rate of change of step-size in the slope-overload region can be s_0 or $\alpha.s_0$ or $\alpha^2.s_0$ etc., by proper choice of α>1, the rate of change of step-size can be made greater than So. It is seen that choice of gives a better performance to slope overload and the parameter β takes care of the granular noise as a result of which a better performance is obtained as compared to SONG and modified ABATE algorithms. This can be observed in the performance comparison described later.

4. PERFORMANCE COMPARISON

In this section, figure 3 is the computer simulation results for comparing the waveforms of all the algorithms are presented with sine wave of 1 kHz, amplitude 3V peak.

Following observations are made from the figure 3.

- ADM with 1-bit memory gives more quantization noise compare to others.
- Modified ABATE has less noise compared to ADM but gives error with decreasing curve of the sinusoid.
- SONG algorithm takes time to get adapted to the sinusoid. But gives better performance later.
- Modified SONG algorithm adapts soon and has equally better performance like SONG algorithm

Applying the 4 different ADM algorithms to a speech signal with sampling rate 44.1 kHz (CD quality speech) gives the following inference.

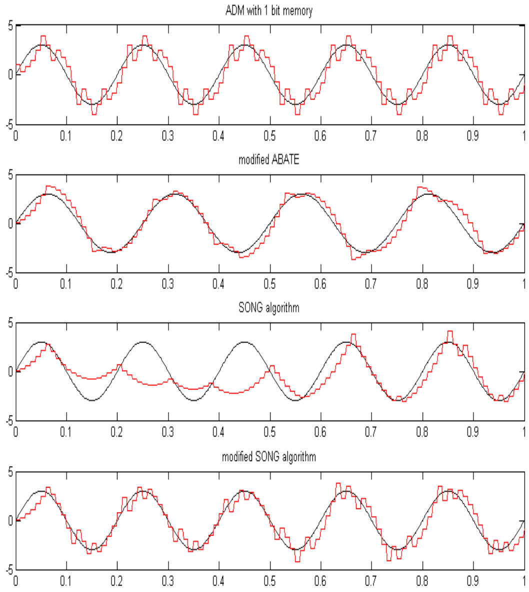

Figure 3: Time comparison of different ADM techniques with sine waveform

For further comparison of the algorithms for the time it would take for faster transitions as well as the response for the invariable amplitude, the algorithms are fed to the square waveform and figure 4 gives the plot.

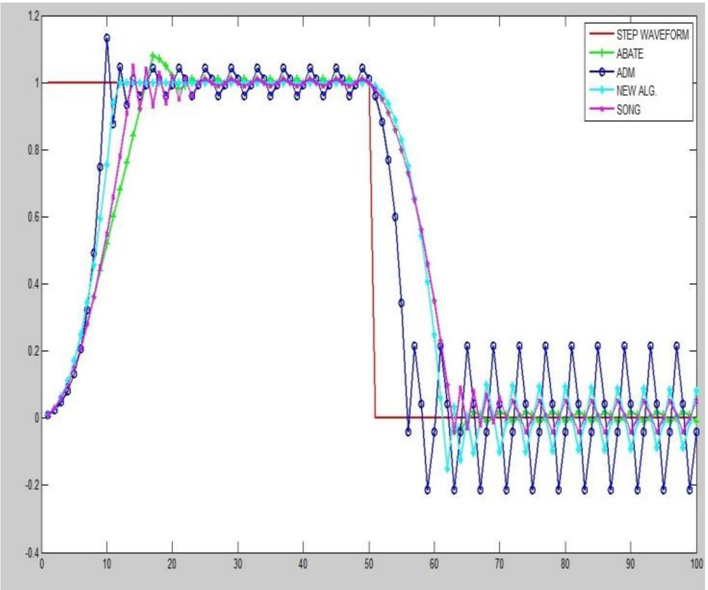

Figure 4: Time comparison of different ADM techniques with square waveform

- As can be observed the 1 Bit ADM algorithm takes a quite less time to get transitioned into the lower level so that the slope over load noise is less. At the same time the granular noise is very bad in this due to the large oscillations.
- ABATE and the SONG algorithms behave same way with fast transition as well as the constant amplitude. Still it produces considerable amount of granular and slope overload noise.

- The new SONG algorithm takes a bit less time to make the transition compared to SONG and ABATE where as the granular noise is observed to be predominant comparatively.

In speech processing, the performance is analyzed by plotting Input signal strength against output SNR.

In figure 5, the performance comparison of 1bit ADM, the modified ABATE, SONG, modified SONG algorithms are done assuming that the channel is noiseless.

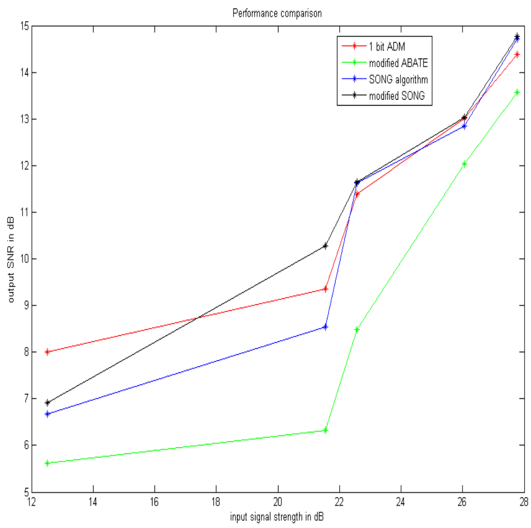

Figure 5:SNR Performance analysis for different ADM techniques

Following observations are made.

- It is noticed that 1 bit ADM has good performance at low input signal strength and is equally well compared to SONG and modified SONG algorithms at high signal strengths.

- Modified ABATE algorithm has very less output SNR, 2.5dB less compared to 1-bit ADM and 1 dB less compared to SONG algorithms. At high input signal strengths the algorithm margin of SNR remains same as seen in low strengths.
- Performance of SONG and modified SONG algorithms are almost same for higher strength input speech. But significant increase in SNR of modified SONG over SONG algorithm is observed as the input signal strength increases at lower signal strength levels.

5. CONCLUSION

Adaptive Delta Modulation which comes under waveform coding techniques plays an important role for low cost hand held devices and applications. Better in terms of less complexity and memory requirement compared to the Linear Predictive Coding (LPC) techniques used in mobile phones. Simple c program can be written to implement it in any processor without involving complex signal processing techniques. Good SNR can be expected. Preprocessing speech before coding would not enhance the performance to a bigger extent.

Main parameter that has to be taken care is the sampling frequency. Sampling at Nyquist rates might not give expected results. Performance betters with good sampling frequencies due to highly correlated sample

5. ACKNOWLEDGMENTS

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