CHOICE OF ANALOG-TO-DIGITAL CONVERTERS FOR AUDIO MEASUREMENTS USING MLS ALGORITHM

Daniel Król

Department of Technology, Higher State Vocational School in Tarnów
Ul. Mickiewicza 8, 33-100 Tarnów, Poland
email: danielkrol@poczta.onet.pl

ABSTRACT

This paper describes research results of two types analog to digital converters (ADC) for audio measurement system using Maximum Length Sequence (MLS) algorithm. Comparison was subjected to ADC with one bit Sigma Delta (SD) modulator and Multibit ADC with Successive Approximation Register (SAR). Superior performance of the SAR analog to digital converters over Sigma Delta ones has been shown.

1. INTRODUCTION

A Digital Signal Processing (DSP) technology needs an analog-to-digital (AD) and digital-to-analog (DA) converters for communication with analog world. The AD and DA converters are bridge among digital processors and measured objects (devices). This paper presents influence of the ADC types on frequency and impulse response measuring quality. A MLS (Maximum Length Sequence) algorithm [1 ÷ 6] is the basis for several audio measurement systems. A maximum length sequence system effectively measures the impulse response of loudspeakers, amplifiers, rooms etc. Impulse response could be measured by transmitting an impulse, and recording the response. The problem is that the magnitude of the impulse has to be relatively high to obtain a decent signal-to-noise ratio (SNR). The MLS technique is a clever method for obtaining a very large processing gain, therefore accurate measurements can be made with moderate sound levels. A MLS is a pseudorandom binary sequence of 1 and -1. The autocorrelation of a MLS for length = M, is M for zero-lag, and -1 for all other lags (Fig. 1). In other words the autocorrelation of the MLS approaches to an impulse as M goes to infinity.
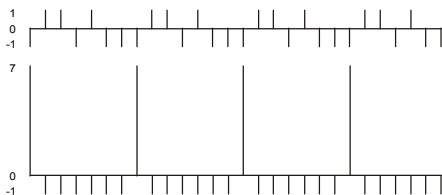

Figure 1 – autocorrelation of MLS with M = 7

A signal from output device under test (DUT) is correlated with original MLS sequence to obtain impulse response of measured element (Fig. 2). In order to reduce calculations of the cross-correlation algorithm a Fast Hadamard Transform (FHT) is often applied [7, 8]. A benefit of the FHT is that it request, like the more familiar Fast Fourier Transform, only n log₂(n) operations. Since the MLS is represented by +1 and -1, the FHT consist of additions and subtractions only. The impulse response is transformed by Fast Fourier Transform (FFT) from time domain to frequency domain so as to generate characteristics of DUT.
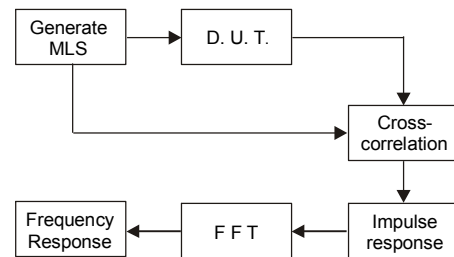

Figure 2 – Block diagram of audio measurement system using MLS

2. PROBLEM STATEMENT

A sigma delta technology is used in most audio applications. The one bit converters are used in music and audio measurement systems for conversion of sine wave signals. However a MLS pseudorandom binary sequence, generated on output measurement system is similar to square wave. To measure device under test accurately analog-to-digital converter used in measurement system, must capture this signal precisely.

2.1 Sigma Delta analog to digital converter

Sigma Delta conversion technology is based on oversampling, noise shaping, and decimation filtering [9 ÷ 11]. There are many advantages of ADC with sigma delta modulators. The major advantages are high resolution (up to 24bits), high signal-to-noise ratio (SNR) in a working pass-band and low cost of implementation.
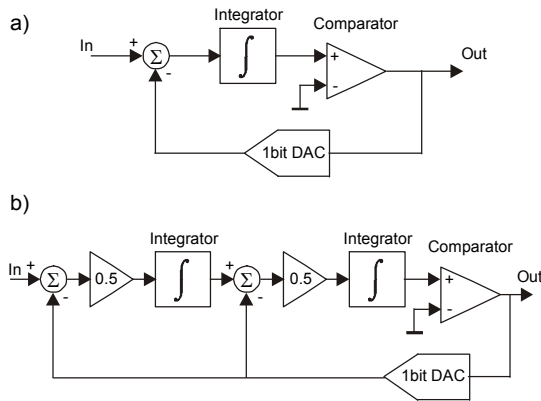
a)

b)

Figure 3 – Block diagram of the first (a) and second (b) order one bit Sigma Delta modulator.

The one bit digital stream from modulator is transformed to multi bit samples in decimation filter. The decimation filter is often realized as Cascaded Integrator Comb (CIC). Block diagram of exemplary third order CIC is shown on figure 4.
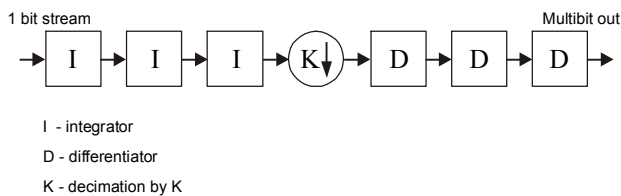

I - integrator
D - differentiator
K - decimation by K

Figure 4 – Block diagram of a third order CIC filter without external decimation stage.

$$I = \frac{1}{1 - z^{-1}} \qquad (1)$$

$$D = 1 - z^{-1} \qquad (2)$$

Disadvantage of SD technology is generation of oscillations for Direct Current (DC) and for the steep signals.

2.2   SAR analog to digital converter

The SAR is a successive-approximation ADC (Fig. 4) based on a charge redistribution digital-to-analog converter (DAC). The charge scaling DAC simply consists of an array of individually switched binary-weighted capacitors. The amount of charge upon each capacitor in the array is used to perform the binary search in conjunction with a comparator interal to the DAC and the successive approximation register [12].
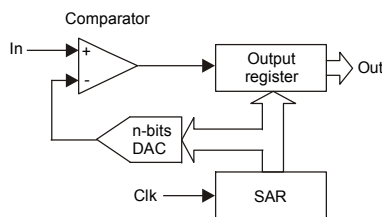

Figure 5 – Block diagram of ADC with successive approximation register.

A SAR ADC offers up to 18 bits resolution. The basic problem in SAR ADC is nonlinearity error of internal multibit DAC and low signal-to-noise ratio (SNR) in comparison to sigma delta ADC. The major disadvantage of high resolution SAR ADC is high cost of implementation.

3.   EXPERIMENTS AND RESULTS

Experiments were divided into the two stages:
- analysis and comparison of square wave signal conversion results;
- analysis and comparison of MLS signal conversion results.

A 24bit/48kHz Sigma Delta ADC and 10bit/48kHz SAR ADC was compared in experiments. Measured data from both converters are normalized to range $-1 \div 1$.
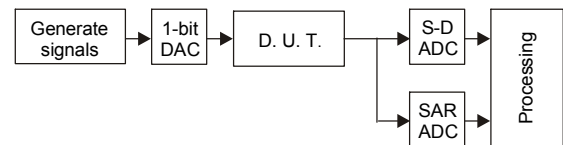

Figure 6 – Block diagram of system for comparing ADC.

3.1 Stage I - Square wave conversion

This stage presents comparison of the conversion a square wave signal. A SAR converter processes this signal exactly with noise at one bit level, however a sigma delta ADC generates oscillations (Fig. 7). This stage shows that these oscillations will be negative for the correlation product.
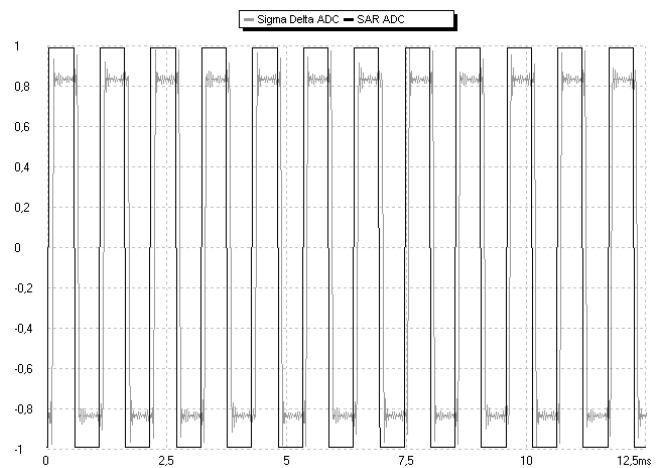

Figure 7 – Conversion of square wave

3.2 Stage II - MLS signal conversion

A MLS signal is similar to square wave and problems of conversion are the same as for square wave. This stage presents influence of the oscillations on correlation product. The MLS output signal is cross-correlated with original MLS sequence (Fig. 2). A length of sequence in experiment was equal to M = 16383 (2^{14}-1).

3.2.1 Analysis of system measurement

Cross-correlation product from SAR analog-to-digital converter (Fig. 8a) is M for zero-lag and −1 for all other lags with noise amplitude value approximately 1/M. The result is almost ideal. Oscillations generated by sigma delta ADC induce errors of MLS correlation (Fig. 8c). Figure 8b shows impulse response of sigma delta modulator with decimation filter. The response of SD converter will be added to impulse response of device under test (DUT).
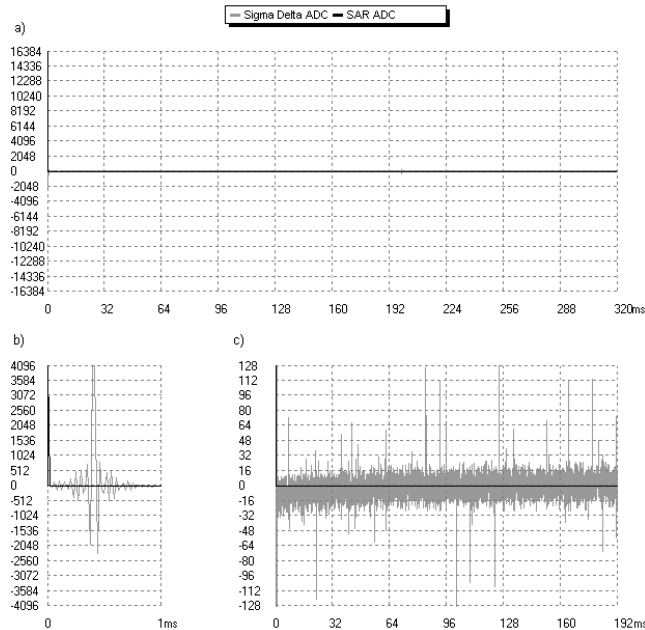

Figure 8 – Impulse response of measurement system.

Frequency response of measurement system is shown on figure 9. The frequency response for SAR ADC is almost ideal line, however plot of sigma delta ADC is strongly noised. A plot of sigma delta ADC requires smoothing process, however even smoothed will be not flat in frequency domain.
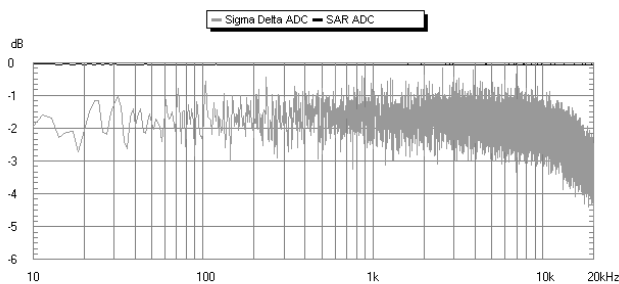

Figure 9 – Frequency response of measurement system.

3.2.2 Measuring of devices

Two types of first order, analog, lowpass and highpass filters were measured in experiment. The response of DUT was convoluted with response of measurement system. When impulse response is a pseudo Dirac's delta function, then plot will be shown on the original impulse response of DUT. Because response of sigma delta ADC is distorted by modulator

and decimation filter, results are falsified. Fig. 10 shows an exemplary impulse response of the first order high-pass filter measured by two ADCs.
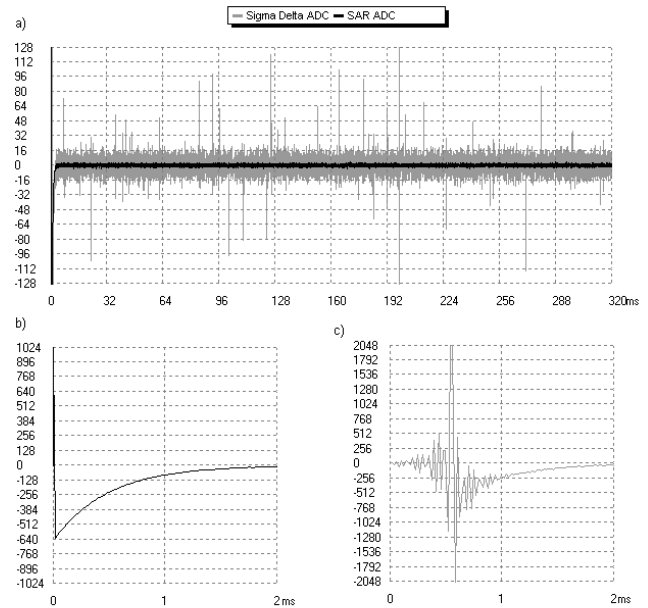

Figure 10 – Impulse response of measured high-pass filter.

It should be noted that impulse response measured by SAR ADC (Fig. 10b) is very precise while the impulse response measured by sigma delta ADC is distorted by oscillations (Fig. 10c). Moreover oscillations generated by sigma delta ADC induce errors of cross-correlation (Fig. 10a). Frequency response of measured filter is shown on figure 11. Plot of the sigma delta ADC is strongly noised and falls over 8kHz while plot of SAR ADC is clear and precise.
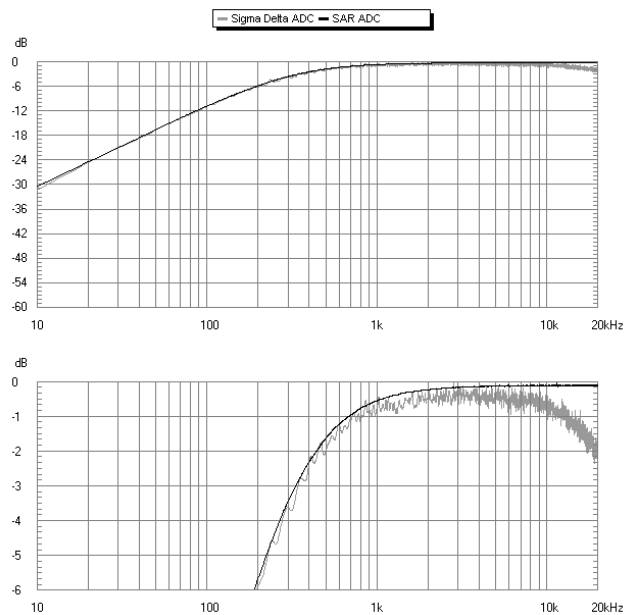

Figure 11 – Frequency response of measured high-pass filter.

Exemplary impulse response of the first order low-pass filter measured by two ADCs has been shown on Fig. 12. Similarly like in the case of high-pass filter, oscillations generated by sigma delta ADC induce errors of cross-correlation (Fig. 12a) and are added to impulse response of measured filter (Fig. 12c).
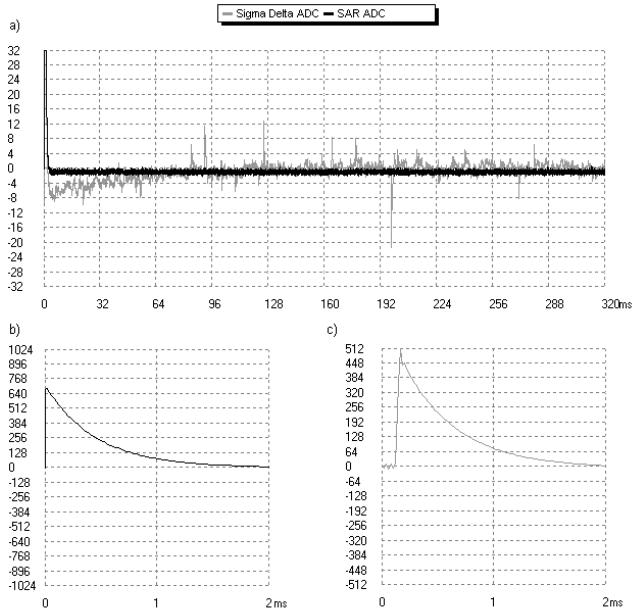

Figure 12 – Impulse response of measured low-pass filter.

Frequency response of low-pass filter is shown on Fig 13. The plot of sigma delta ADC is noised and falls over 8kHz likewise in previous case.
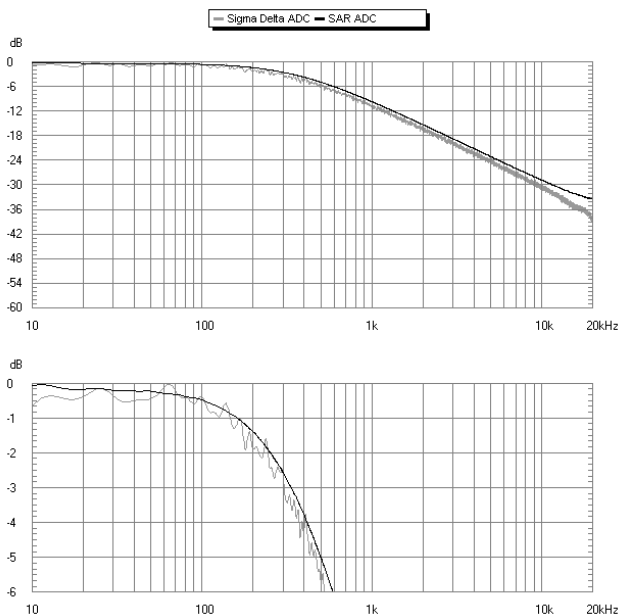

Figure 13 – Frequency response of measured low-pass filter.

4. CONCLUSION

SAR analog-to-digital converters are highly recommended for the system measurement based on MLS algorithm. The frequency characteristics from classical audio measurements system based on sigma delta ADC and using MLS algorithm are falsified and must be corrected. These systems can not measure exactly impulse response of DUT. Many CAD programs import measured impulse response for simulation speakers, filter etc. Exactitude of simulations depends on exactitude measure. This paper describes comparison results of analog to digital converter with SAR and 1-bit sigma delta modulator. In the future comparison of 16 bits SAR ADC and sigma delta with multibit modulator ADC is planned.

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