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On superiority of Successive Approximation Register over Sigma Delta AD converter in standard audio measurements using Maximum Length Sequences

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***Abstract*—The paper describes some new research results from designing analog-to-digital conversion for audio measurement system using MLS (Maximum Length Sequence) algorithm. Evaluation version of USB audio measurement interface with two types of high quality analog-to-digital converters (ADC) has been presented in the paper. Comparison was subjected to 16 bit ADC with one bit Sigma-Delta (SD), fourth-order modulator and 16 bit SAR (Successive Approximation Register) ADC. Superior performance of the SAR analog-to-digital converters over Sigma-Delta ones has been shown.**

***Keywords*—Audio, Measure, MLS, SAR, Sigma-Delta**

I. Introduction

Preliminary research results from comparison of 10 bit Successive Approximation Register ADC and 24 bit sigma-delta ADC have been presented in [1] where superior performance of SAR analog-to-digital converters over the sigma-delta ones, has been shown. This paper presents some new research results for two types of 16-bits analog-to-digital converters applied in audio measurement system using MLS (Maximum Length Sequence) algorithm [2 ÷ 4]. ADC with 1 bit, 4th order sigma-delta modulator versus ADC with Successive Approximation Register has been compared. In order to accomplish described research an USB audio measurement interface with two types of ADCs was built. The MLS algorithm is the basis for several audio measurement systems used for effective measuring the impulse response of loudspeakers, amplifiers, rooms etc. [5, 6]. Impulse response could be measured by transmitting an impulse, and recording the response. An MLS is a pseudo-random binary sequence of 1 and -1 (Fig. 1a). Its autocorrelation for length M, is equal M for the 0-th lag, and -1 for all other lags (Fig. 1b). In other words the autocorrelation of the MLS approaches to an impulse as M goes to infinity. Output signal from device under test (DUT) is correlated with original MLS sequence to obtain impulse response of measured

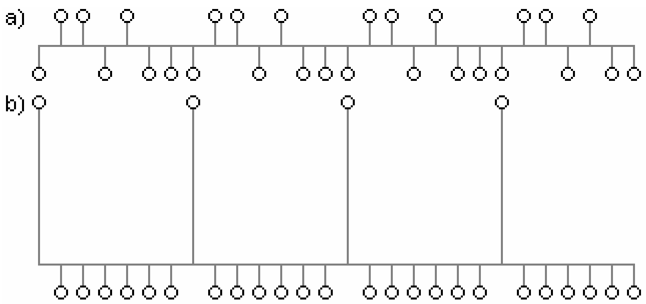

Fig. 1. Example of MLS (a) and its autocorrelation (b) for M = 7

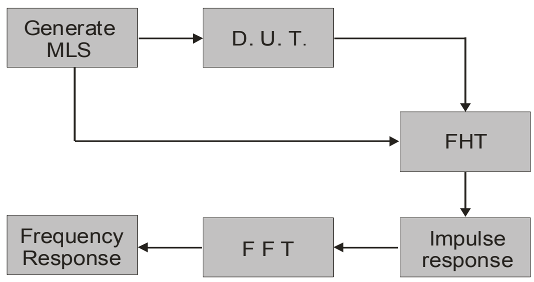

Fig. 2. Block diagram of MLS algorithm

element (Fig. 2). In order to reduce calculations of the cross-correlation algorithm a Fast Hadamard Transform (FHT) is often applied [7, 8]. The impulse response is transformed by Fast Fourier Transform (FFT) into the frequency domain in order to generate frequency characteristics of DUT. It should be however emphasised that implementation of MLS algorithm using universal audio measurement equipment gives inaccurate results. Significant improvement can be achieved by specially designed hardware.

II. Problem Statement

Most audio measurement systems is based on sigma-delta ADCs and DACs. A sigma-delta technology is a good solution for music conversions and signal measurement based on sinusoidal waveforms. Its main disadvantage is a pre- and post-ringing effect. The effect is clearly observed in conversion of signals with fast rising and falling edges like square-wave or MLS [9 ÷ 11]. The sigma-delta ADC gives an average of oversampled bitstream [12 ÷ 14] and in consequence resolution for a fast rising signal is smaller than for sinusoidal waveform. In case of 64- fold oversampling ADC there are 64 adds or subtracts.

III. Hardware Realization

In order to compare results of analog-to-digital conversion of two types of ADCs, a special evaluation USB audio measurement interface has been built. The hardware is optimized for MLS with sampling rate 48kHz and resolution 16 bit in both ADCs. For precise MLS signal generation DAC was given up. Output buffer amplifiers are driven directly from microcontroller port pins by simple voltage level converter. Input and output circuits were designed for a wide frequency range including DC (Direct Current). Both analog-to-digital converters have a linear-phase, first order antialiasing filters according to datasheet [15, 16].

A. Analyzer Board

Block diagram of USB measurement interface is shown in figure 2. Binary signal from microcontroller pin is sent to output buffer amplifiers by level converter 0V – 5V to -1V – 1V.

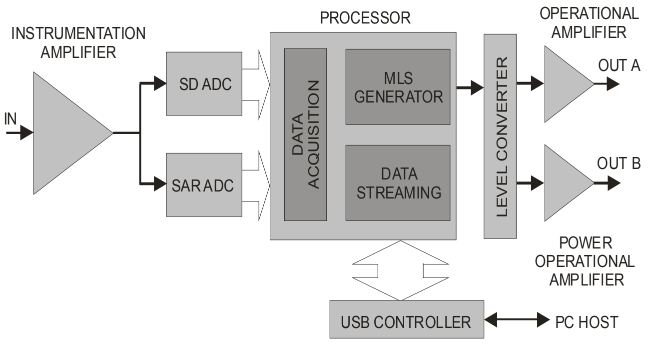

Fig. 3. Block diagram of dual ADC board

The audio measurement interface has two outputs. The first one (Out A) is high impedance output that is used for driving active devices, like amplifiers, active filters etc. The second output (Out B) is meant for driving low impedance loads like speakers, and is equipped with power operational amplifier. This solution gives good quality MLS signal. High precision instrumentation amplifier with programmable gain from 0 to 50dB has been applied on the interface input. Measurement data from two types (sigma-delta and SAR) of 16 bits ADCs are acquired in microcontroller and sent by USB to the application on host PC.

B. Sigma-Delta ADC

The AD1870 [15] is a stereo, 16-bit oversampling ADC based on sigma-delta (Σ-Δ) technology designed primarily for digital audio bandwidth applications requiring a single 5V power supply. Each single-ended channel consists of a fourth-order one-bit noise shaping modulator and a digital decimation filter. Input signals are sampled at 64 × fs onto internally buffered switched-capacitors, eliminating external sample-and-hold amplifiers and minimizing the requirements for antialias filtering at the input.

C. Successive Approximation ADC

The AD7652 [16] is a 16-bit, 500 kSPS, charge redistribution SAR analog-to-digital converter that operates from a single 5V power supply. The part contains a high speed 16-bit sampling ADC, an internal conversion clock, internal reference, error correction circuits, and both serial and parallel system interface ports.

IV. Experiments and Results

A. Time and frequency response of measurement interface

Figure 4 shows results from conversion of MLS signal performed by two tested analog-to-digital converters. The MLS signal on the output of measurement interface is two state, for example 1V or -1V. The SAR type ADC processes this signal exactly whereas a sigma-delta ADC generates oscillations. Figure 5 shows impulse response of measurement interface calculated from cross-correlation of MLS. The result of cross-correlation product from SAR type ADC is almost ideal.

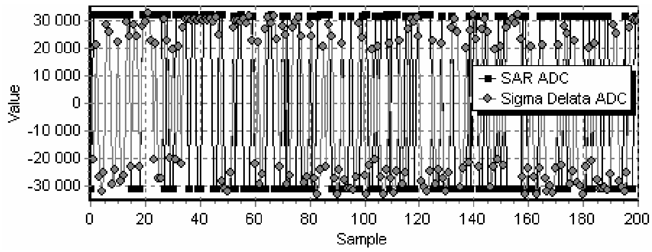

Fig. 4. Acquisition MLS by SAR and Sigma-Delta ADC

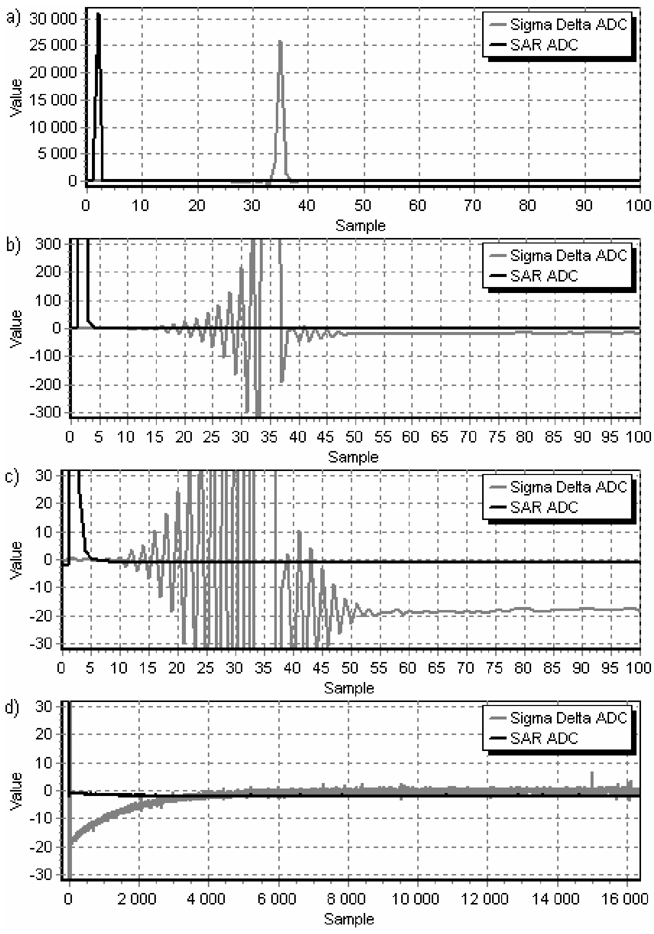

Fig. 5. Calculated impulse response of two types of ADCs

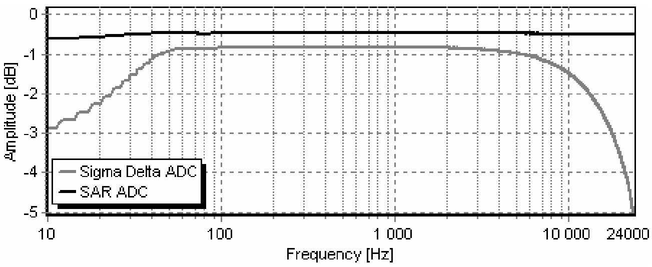

Fig. 6. Calculated frequency responses of two ADC types

In turn oscillations generated by the sigma-delta ADC cause errors in MLS correlation (Fig. 5d). The response of decimation filter are added to the impulse response of device under test (DUT). Figure 6 shows frequency responses of two ADC types. Frequency response of SAR analog-to-digital converter is an almost ideal flat line in whole frequency band. The frequency response of sigma-delta ADC is limited by lower cutoff frequency resulted from offset cancellation filter response and by upper cutoff frequency resulted from decimation filter.

B. Load influence on power amplifier work

Typically Impulse and step responses are used for analyzing dynamic parameters of amplifiers. Load influence on amplifier with closed feedback loop (See Fig. 7) should be very well visible in impulse responses. Impulse response calculated from SAR ADC data is very clear, while sigma-delta ADC generates strong oscillations and noisy impulse response. See figures 8 and 9.

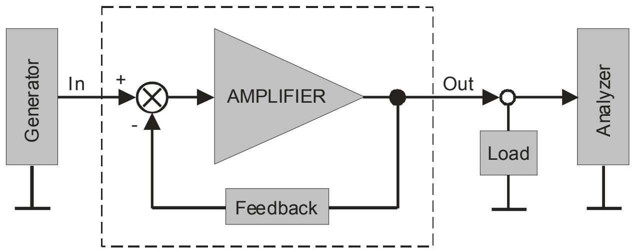

Fig. 7. Circuit for measurement of amplifier parameters

The type of impulse response can be very well interpreted from its first 20 samples calculated from the data measured by Successive Approximation Register ADC. Influence of load connected to amplifier is clear and easy to interpret. In case of sigma delta ADC, the load influence is masked by oscillations generated by decimation filter.

The influence of resistance-capacitance (RC), resistance-inductance (RL) and resistance-inductance-capacitance (RLC) loads is shown in figures 9 a) - c). Similarly to figure 8, the oscillations generated by decimator are masking essential part of impulse response information. The frequency response shows influence of different loads on amplitude characteristic. Attention should be paid on nonlinearity in amplitude characteristic of sigma delta ADC. In the frequency response of amplifier with RLC load measured by SAR ADC (Fig. 9c) a resonance near the frequency 10kHz is visible.

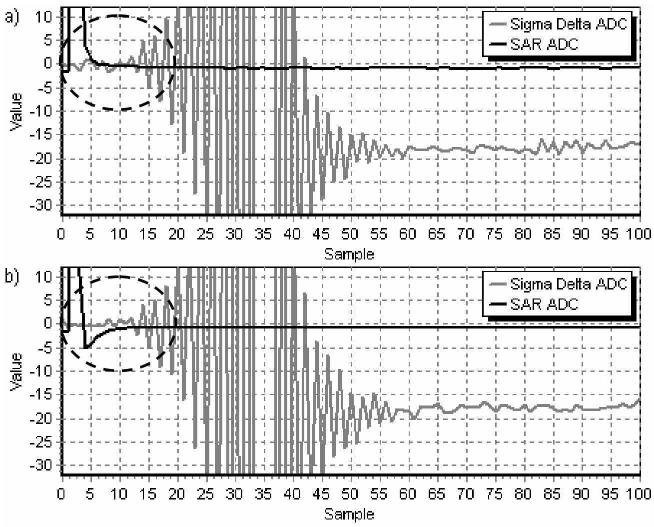

Fig. 8. Load influence on power amplifier work, a) without load, b) with resistant load

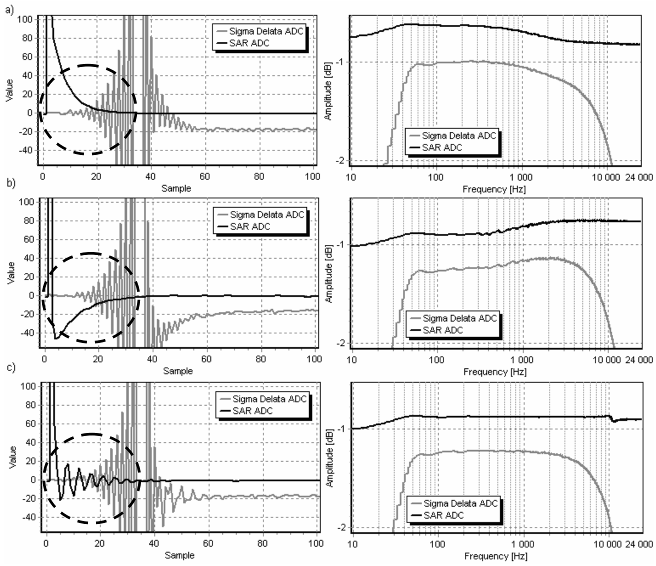

Fig. 9. Load influence on power amplifier work, a) resistance-capacitance load, b) resistance-inductance load, b) resistance-inductance-capacitance load

C. Real resolution of ADC in near field headphone measurement

The SPL (Sound Pressure Level) measurement method of headphones in near field is shown on figure 10. The sigma delta ADC conversion operation consists of averaging the one-bit stream from modulator in decimation filter. As a result the resonances are masked. In the experiment a measurement microphone has been shifted 0.25 mm per measurement point. A sigma-delta ADC averages SPL plot masking resonances in medium and high frequency band (See Fig. 11).

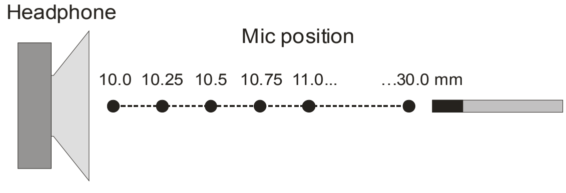

Fig. 10. Method of headphones SPL measuring

Measured SPL as a function of the distance from the sound source is shown an figures 12 a) - c) as a 3D plot. Figure 12 a) shows calculated near field SPL for sigma delta ADC in 80 different point whereas conversion results for Successive Approximation Register ADC are shown in figure 12 b). Difference between both analog-to-digital converters is depicted in figure 12 c). We see the resonances averaging effect that is present in sigma delta ADCs. In it precision audio measurements it may seriously change measurement results and lead to incorrect simulations based on inaccurate models.

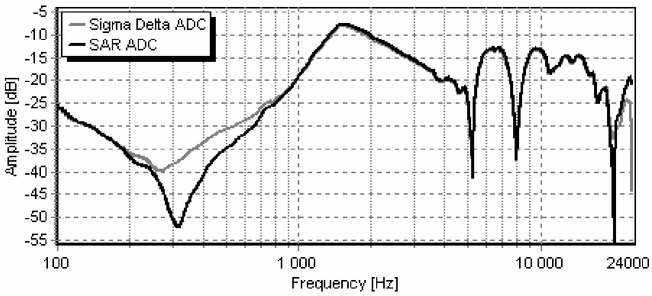

Fig. 11. Different SPLs measured by two types of ADCs

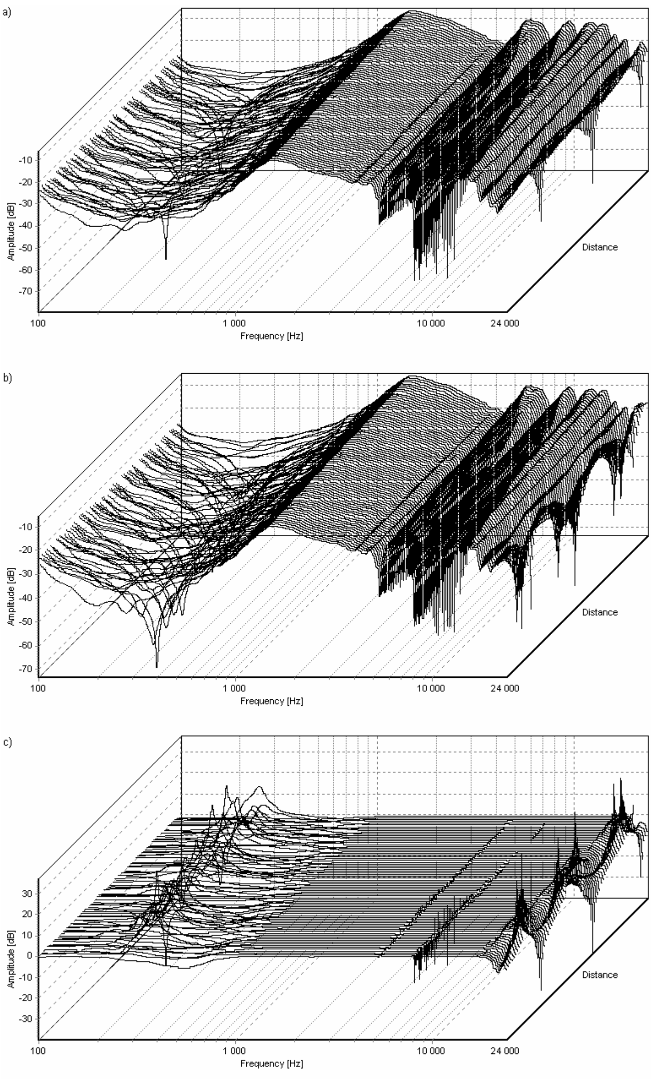

Fig. 12. Headphones frequency response in different distance, a) for sigma-delta ADC, b) for SAR ADC, c) the difference

V. CONCLUSIONS

In the paper two 16 bit analog-to-digital converters, realized in different technologies, have been compared. The SAR analog-to-digital converters are highly recommended for impulse and frequency response measurements based on the MLS algorithm. In contrary frequency responses from classical audio measurements system based on sigma-delta ADC and using MLS algorithm are falsified, these systems can not exactly measure also impulse response of DUT. The influence of load connected to amplifier, measured by Successive Approximation Register ADC, is clear and easy to interpretate. In case of sigma delta ADC, influence of load is masked by oscillations generated by the decimation filter. Many CAD programs import measured impulse response for simulation of speakers, amplifiers, filter etc. Circuit resonances are averaged and masked in sigma delta ADCs. In precision audio measurements these effects may seriously change measurement results and lead to incorrect simulations based on inaccurate data.

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