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# THE APPLICATION OF SIGMA DELTA D/A CONVERTER IN THE SIMPLE TESTING DUAL CHANNEL DDS GENERATOR

***J. Fischer***
Faculty of Electrical Engineering
Czech Technical University, Prague, Czech Republic

*Abstract*: *This article deals with the application of sigma delta DAC in the low cost testing dual channel generator with direct digital synthesis based on digital signal processor. The results of measurement performed by means of ADSP2181 EASY KIT are presented.*

*Keywords: DDS generator, sigma delta D/A converter*

## 1 INTRODUCTION

Sigma delta D/A converters are originally designed and most often used for the purpose of audio signals generation in the frequency range approximately from several tens Hz up to 15 kHz. As an example could serve D/A converter included in the codec AD1847.
The codec Analog Devices AD1847 integrates dual channel 16 bit sigma delta A/D converter (ADC) and dual channel 16 bit sigma-delta D/A converter (DAC). The DAC contains digital low pass interpolation filter. The anti-imaging interpolation filter oversamples and digitally filters the high frequency images [1]. The DAC outputs are then filtered in the analogue domain by a combination of switched- capacitor and continuos-time filter.

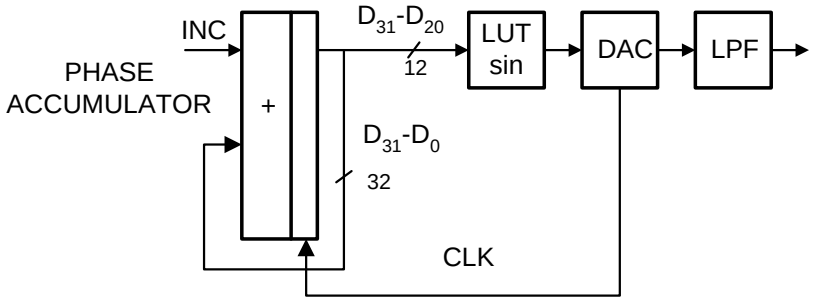


**Figure.1** Basic diagram of a DDS generator

## 2 SIGMA DELTA DAC IN THE DSP BASED SIGNAL GENERATOR

This DAC AD1847 is coupled to the digital signal processor (DSP) ADSP2181 in the Easy Kit ADSP2181. The block functional diagram of designed Direct Digital Synthesis (DDS) generator is on the Fig.1. The heart of the system is the 32 bit phase accumulator contents of which is updated once per each clock cycle. The phase increment (INC) is summed in the phase accumulator at every period of clock signal CLK. The frequency of output signal depends on the value of INC. The 13 most significant bits of phase accumulator actually contribute to the address of the current sample of sin wave stored in the waveform memory - look up table (LUT). This sample is transferred from the waveform memory to the DAC. The internal low pass filter (LPF) eliminates high harmonics above one half of sampling frequency.

This block diagram has been realised in the DSP - ADSP2181 by software. The sampling signal *CLK* with frequency $f_s$ =44100Hz is generated by the codec, which requests every 1/44100 second a new data. The DSP realises two 32 bit phase accumulators $P.ACC_1$ and $P.ACC_2$. They accumulate independently phase increments $INC_1$ and $INC_2$ respectively. If the phase increments $INC_1$ and $INC_2$ are the same, the difference of the initial values $SET_1$ and $SET_2$ stored in the $P.ACC_1$ and $P.ACC_2$ sets phase shift of signals $OUT_1$ and $OUT_2$.

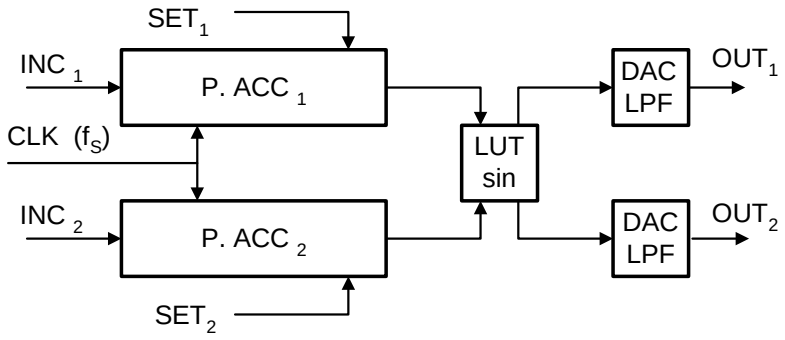


**Figure. 2** The realisation of DDS generator in the ADSP2181

The frequency resolution of this generator is theoretically determined by the equation (1)

$$f_{res} = \frac{44100}{2^{32}} = 1.027 * 10^{-5} \; [Hz] \tag{1}$$

and phase shift resolution by (2).

$$PH_{res} = \frac{2p}{2^{32}} = 1{,}46 * 10^{-9} \; [rad] \tag{2}$$

If the phase increments *INC1* and *INC2* are constant, but not equal, the output signals with different frequencies are generated. The mean value of generated frequency is given by (3),

$$f = f_S \frac{INC}{2^{32}} \tag{3}$$

and mean value of phase shift by (4).

$$PH = \frac{2p}{2^{32}} (SET_2 - SET_1) \tag{4}$$

The DSP may vary independently the output frequency (in sweep mode) of both channels by the continuously modifying of the phase increments $INC_1$ and $INC_2$.

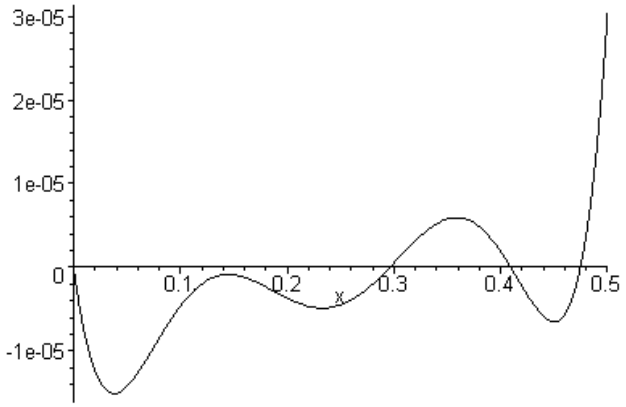


**Figure 3.** The approximation error

The experimental DSP based DDS generator uses internal RAM of the ADSP2181 as LUT. The sin waves with 256, 1024, 2048, 4096 and 8192 samples of 16- bit word stored in the LUT have been designed.

Due to small capacity of internal RAM at some ADSP21XX family members, instead of using LUT calculation of value of samples in the real time by using polynomial approximation of sin function could be performed - [2] and [3].

The DSP calculates the values of samples according the formula (5).

$$sin(x) = 3.140625\,x + 0,02026367\,x^2 - 5,325196\,x^3 + 0,544677\,x^4 + 1,800293\,x^5 \qquad (5)$$

where $x$ lies in the interval from 0 to 0.5 and $\alpha$ is the phase angle measured in radians.

$$x = \frac{a}{p} \qquad (6)$$

The coefficients in polynomial function are suitably chosen so that they can be represented by 16-bits numbers with an error less than 1E-7.

The maximum deviation of polynomial approximating function from exact sin x function is less than 3E-5. The calculation of value by polynomial approximation takes less than 2.5 μs. The graphical representation of the approximation error is on Fig.3. The resolution of approximation of sin x function by calculation corresponds to resolution which could be reached by LUT designed for 65536 samples.

## 3 THE RESULTS OF MEASUREMENT

The measurement was performed for both cases i.e. when function sin x was generated using LUT as well as when values were calculated by the above mentioned algorithm. The spectral purity of generated signal has been measured by the FFT Signal analyser HP35670A and Spectrum analyser HP3580A. The content of higher harmonics does not represent a problem; the measured value of THD is better than 0,01%.

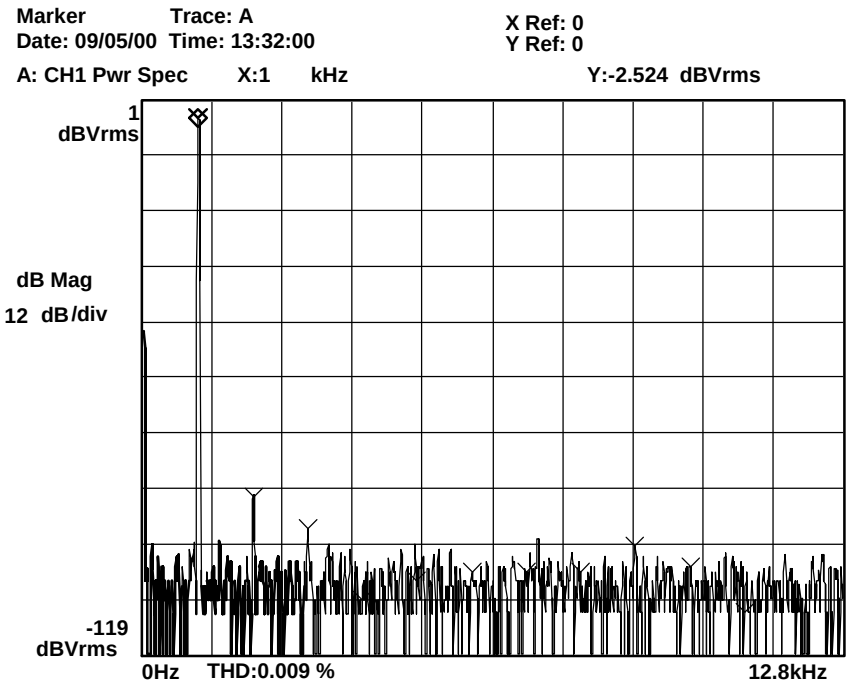


**Figure 4.** Spectrum 1, measured without difference amplifier.

The Fig.4 shows spectrum of signal with f=1kHz generated using calculation according Equ.5. The existence of second harmonic at 2kHz is clearly visible. The improvement has been reached by implementation of a difference amplifier (type -TL084).

The signals from channel *OUT1* and *OUT2* have been brought on the input of difference amplifier. The same signal with frequency of 1 kHz and phase shift of 180 degrees has been generated. The degree of improvement can be seen from the Fig.5 and Fig.6.

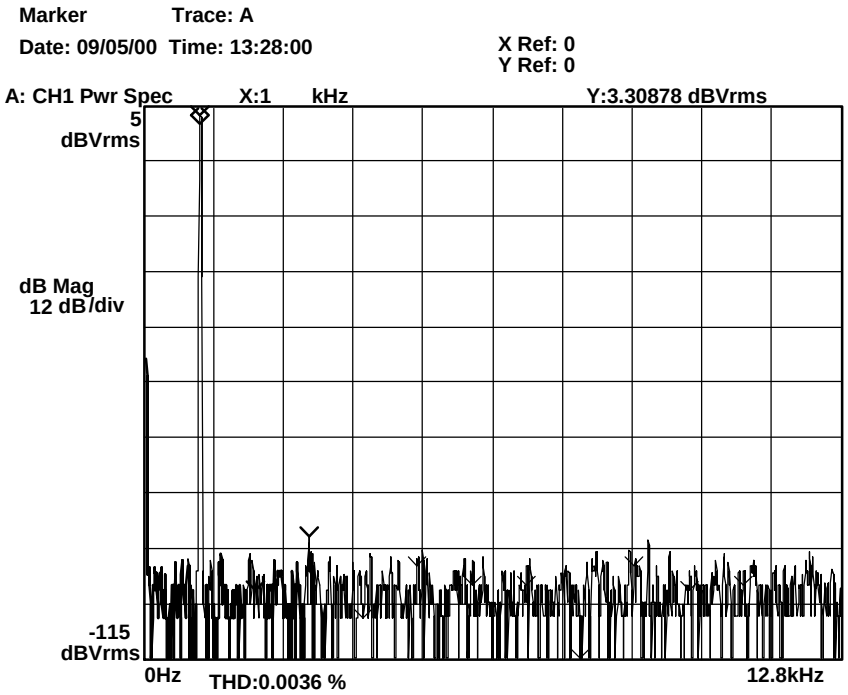


**Figure 5.** Spectrum 2, measured with the difference amplifier

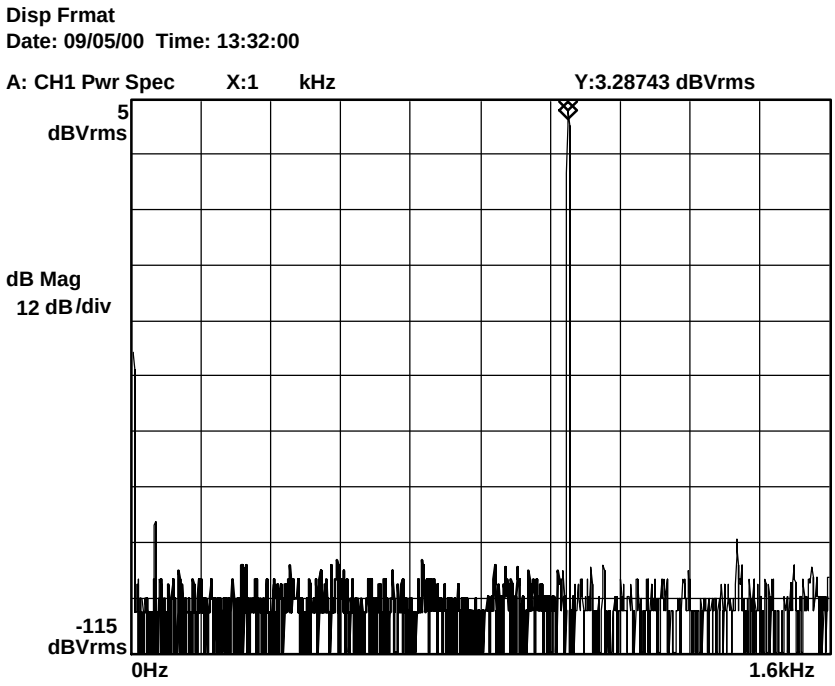


**Figure 6.** Spectrum 2,measured with the difference amplifier, detail

## 4 CONCLUSION

The all experiments and measurements have been performed with the simple low cost (approx. 100 USD) Analog Devices Easy Kit Lite ADSP2181. The RS232 interface has been used for

communication with PC. In order to avoid the influence of interferring signal coming from PC it was necessary to disconnect the PC after the parameters setting. The better solution is to use an optical isolation. For testing purposes, it is possible to load the internal RAM - LUT with a user-defined waveform. The waveform data (for. example (sinx)/x, sin(x) * $e^{-x}$, saw signal) were computed by Microsoft Excel and then linked to the ADSP program. The experiments showed that performance of internal LPF of codec is not optimal for arbitrary signal generation. This is especially true for generation of signals with saw waveform.

The ADSP2181 may to realise two different look up tables for channel 1 and channel 2 in its internal RAM, so that the different waveforms could be generated on the output 1 and output 2.

The described system realises the known method of DDS generation by a simple low cost sigma delta DAC in the co-operation with the digital signal processor.

Sinusoidal signals with frequency higher than 100 kHz can be generated if high speed D/A converters and external LP filters are used. The generator here described can be used as low cost source for testing of A/D converters included in one chip microcomputers.

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**AUTHOR:** Ing. Jan FISCHER, CSc. Department of Measurement, Faculty of Electrical Engineering, Czech Technical University, Technicka 2, 166 27 Prague-6, Czech Republic, Phone Int. ++42 2 2435 2179, Fax Int ++42 2 311 9929, E-mail: Fischer@feld.cvut.cz