XVIII IMEKO WORLD CONGRESS
11th WORKSHOP ON ADC MODELLING AND TESTING
September, 17 – 22, 2006, Rio de Janeiro, Brazil

DYNAMIC CHARACTERIZATION OF A/D CONVERTERS BY RAMP TESTING SIGNALS

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Abstract: In this paper a method for evaluation the dynamic performances of an analog-to-digital converter (ADC) by ramp testing signals is proposed. This method permits the estimation with high accuracy of ones of the most important dynamic parameters of an ADC – signal-to-noise and distortion ratio (SINAD) and effective number of bits (ENOB). Carried out simulations confirm that the proposed method leads to accurate results.

Keywords: ramp signals, modulo time-plot, estimation of the dynamic parameters SINAD and ENOB of an ADC.

1. INTRODUCTION

Dynamic parameters of an analog-to digital converter (ADC) are obtained by time-domain methods or frequency-domain methods. The most common used time-domain methods are the sine-fit test methods which performs the estimation of three or four unknown sinewave parameters [1]-[4]. In the frequency-domain methods a sinewave is applied at the input at the ADC under test. Then, a certain algorithm, based on the discrete Fourier transform (DFT) of the corresponding ADC output codes is used to estimate the ADC dynamic parameters [1], [2,], [5], [6]. So, the most employed methods for estimating the ADC dynamic parameters use the sinewaves as test signals. The sinewaves are used because they are easy to generate in practice at the frequency of interest with adequate fidelity.

In this paper a time-domain method for dynamic characterization of ADCs by ramp testing signals is proposed. This method is an alternative to the existent time-domain methods which use the sinewaves as test signals. The modern signal generators provide also, high accurate ramp signals which can be employed to test the ADCs.

By the proposed method ones of the most important dynamic parameters of an ADC – signal-to-noise and distortion ratio (SINAD) and effective number of bits (ENOB) can be estimated with high accuracy.

The performances of the proposed method are analyzed by simulation.

2. PRESENTATION OF THE PROPOSED METHOD

We consider an n-bit ADC. The test signal is a ramp signal with amplitude A, frequency f_{in} and offset d. For the k-line the ramp signal is given by

$$x(t) = \frac{2A}{T}(t + t_0) - A - 2A(k-1) + d, \quad k = 1, 2, 3, \ldots \quad (1)$$

where T is the ramp signal period ($T = 1/f_{in}$) and $0 \le t_0 < T$.

To test all the ADC output codes it is necessarily to set the peak-to-peak value of the ramp signal to ADC full-scale range, *FSR*.

The signal obtained at the ADC output for the k-line is given by

$$y(mT_s) = \frac{2A}{T}(mT_s + m_0 T_s) - A - 2A(k-1) + d + e(m), \quad k = 1, 2, 3, \ldots \quad (2)$$

where: T_s is the ADC sampling period ($T_s = 1/f_s$),
$t_0 = m_0 T_s$, in which m_0 is integer, $0 \le m_0 < T/T_s$,
$e(m)$ is the ADC noise.

When M samples of the ADC output are acquired, the relationship between the frequencies f_{in} and f_s is

$$\frac{f_{in}}{f_s} = \frac{J + \delta}{M} \quad (3)$$

where J is the number of recorded ramp cycles (J is an integer) and δ is the fractional part of the recorded cycles, $0 \le \delta < 1$. When $\delta = 0$ the sampling process is coherent with the ramp input signal. J must be relative prime with M to test different ADC codes.

The rapport between the frequencies f_{in} and f_s represents the normalized frequency, $f = f_{in}/f_s$.

Moreover, to test all the ADC codes it is necessarily to have a number of samples which satisfy the following condition [7]

$$M \geqslant 2^n - 1. \quad (4)$$

To estimate with high accuracy the dynamic parameters SINAD and ENOB of an n-bit ADC by ramp testing signals a method is proposed. This method consists in the following steps:

1) A ramp signal with amplitude A, frequency f_{in} and offset d (like the one given by the expression (1)) is applied at the input of the ADC under test. The peak-to-peak value of the ramp signal is set to ADC full-scale range, *FSR*.

2) M samples of the ADC output $y(0)$, $y(T_s)$,…, $y((M\text{-}1)T_s)$ are acquired. M must satisfy the condition (4). Also, J must be relative prime with M.

3) The modulo T of the sample times $t_m = mT_s$ are computed

$$\tilde{t}_m = t_m \bmod T = (mT_s) \bmod T, \quad m = 0,1,2,\ldots,M-1. \tag{5}$$

Thus, the following vector $\tilde{\boldsymbol{t}}$ is obtained

$$\tilde{\boldsymbol{t}} = [\tilde{t}_0\ \tilde{t}_1\ \tilde{t}_2 \ldots \tilde{t}_{M-1}]. \tag{6}$$

Then the elements of the vector $\tilde{\boldsymbol{t}}$ are sort in ascending order. Thus, the vector $\tilde{\boldsymbol{t}}_{\boldsymbol{a}}$ is obtained

$$\tilde{\boldsymbol{t}}_{\boldsymbol{a}} = sort(\tilde{\boldsymbol{t}}) = [\tilde{t}_{a0}\ \tilde{t}_{a1}\ \tilde{t}_{a2} \ldots \tilde{t}_{a(M-1)}] \tag{7}$$

where $sort(\boldsymbol{z})$ sorts the elements of vector $\boldsymbol{z}$ in ascending order.

The sample values $y(mT_s)$, m= 0, 1, 2,…, M-1, are reordering to correspond to the values $\tilde{t}_{am}$, m= 0, 1, 2,…, M-1. Thus, the following vector $\tilde{\boldsymbol{y}}_{\boldsymbol{a}}$ is obtained

$$\begin{aligned}\tilde{\boldsymbol{y}}_{\boldsymbol{a}} &= [y(\tilde{t}_{a0})\ y(\tilde{t}_{a1})\ y(\tilde{t}_{a2}) \ldots y(\tilde{t}_{a(M-1)})] \\ &= [\tilde{y}_{a0}\ \tilde{y}_{a1}\ \tilde{y}_{a2} \ldots \tilde{y}_{a(M-1)}].\end{aligned} \tag{8}$$

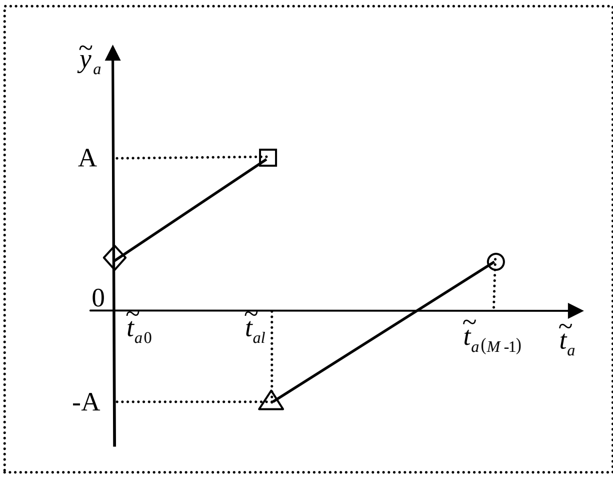
The elements of vector $\tilde{\boldsymbol{y}}_{\boldsymbol{a}}$ are plotted by points as a function of the elements of vector $\tilde{\boldsymbol{t}}_{\boldsymbol{a}}$. This plot represents also, the modulo-time plot of the acquired ramp signal [8].

4) The modulo-time plot of the acquired ramp signal obtained at the previous step is modified to achieve a plot in which the points are represented in one period, following a line, beginning to the minimum (maximum) value of $\tilde{\boldsymbol{y}}_{\boldsymbol{a}}$ and up to the maximum (minimum) value of $\tilde{\boldsymbol{y}}_{\boldsymbol{a}}$ in the case of an increasing (decreasing) ramp signal. For this purpose the elements of the vectors $\tilde{\boldsymbol{t}}_{\boldsymbol{a}}$ and $\tilde{\boldsymbol{y}}_{\boldsymbol{a}}$ are reordering. Thus, it follows the vectors $\tilde{\boldsymbol{t}}_{\boldsymbol{r}}$ and $\tilde{\boldsymbol{y}}_{\boldsymbol{r}}$

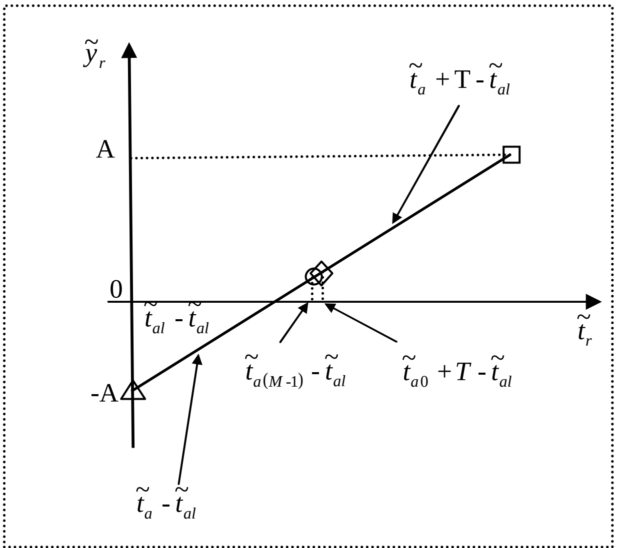
$$\tilde{\boldsymbol{t}}_{\boldsymbol{r}} = [\tilde{t}_{r0}\ \tilde{t}_{r1}\ \tilde{t}_{r2} \ldots \tilde{t}_{r(M-1)}] \tag{9}$$

$$\begin{aligned}\tilde{\boldsymbol{y}}_{\boldsymbol{r}} &= [y(\tilde{t}_{r0})\ y(\tilde{t}_{r1})\ y(\tilde{t}_{r2}) \ldots y(\tilde{t}_{r(M-1)})] \\ &= [\tilde{y}_{r0}\ \tilde{y}_{r1}\ \tilde{y}_{r2} \ldots \tilde{y}_{r(M-1)}].\end{aligned} \tag{10}$$

To understand much better how this modification has been made in Fig. 1 is shown, in the case of a bipolar increasing ramp signal, how the modulo-time plot of the ramp signal, $\tilde{\boldsymbol{y}}_{\boldsymbol{a}}(\tilde{\boldsymbol{t}}_a)$ is modified to obtain the plot $\tilde{\boldsymbol{y}}_r(\tilde{\boldsymbol{t}}_r)$.



(a)



(b)

Fig. 1. (a) Modulo-time plot of a bipolar increasing ramp signal; (b) Modified modulo-time plot of a bipolar increasing ramp signal.

5) The best straight-line corresponding to the modified modulo-time plot is determined by means of the least squared fitting algorithm

$$\hat{y} = p\tilde{\boldsymbol{t}}_{\boldsymbol{r}} + q \tag{11}$$

where

$$p = \frac{\sum_{m=0}^{M-1} \tilde{y}_{rm}\tilde{t}_{rm} - \frac{1}{M}\left(\sum_{m=0}^{M-1} \tilde{y}_{rm}\right)\left(\sum_{m=0}^{M-1} \tilde{t}_{rm}\right)}{\sum_{m=0}^{M-1} \tilde{t}_{rm}^2 - \frac{1}{M}\left(\sum_{m=0}^{M-1} \tilde{t}_{rm}\right)^2}$$

and

$$q = \frac{1}{M}\sum_{m=0}^{M-1} \tilde{y}_{rm} - \frac{p}{M}\left(\sum_{m=0}^{M-1} \tilde{t}_{rm}\right).$$

6) The residual error ***err*** between $\hat{\boldsymbol{y}}$ and $\tilde{\boldsymbol{y}}_r$ is computed

$$\boldsymbol{err} = \hat{\boldsymbol{y}} - \tilde{\boldsymbol{y}}_r . \qquad (12)$$

7) Finally, the dynamic parameters SINAD and ENOB are estimated by

$$SINAD_{est} = 20\log\frac{[\hat{y}(M) - \hat{y}(1)]}{\sqrt{12}\ rms(\boldsymbol{err})} \qquad [dB] \qquad (13)$$

and

$$ENOB_{est} = n - \log_2\left(\frac{rms(\boldsymbol{err})}{q/\sqrt{12}}\right) \qquad (14)$$

in which rms(***err***) is the rms value of the error ***err*** and q is the ideal ADC code width.

3. A CONSTRAINT IMPOSED TO THE MODULUS OF THE RELATIVE ERROR OF THE NORMALIZED FREQUENCY

The performances of the proposed method depend upon the accuracy of the vector $\tilde{\boldsymbol{t}}_r$ values. From this reason the input frequency and the sampling frequency must be known with very high accuracy.

We assume that the errors which affect the modulo-time plot of the acquired ramp signal are caused only by the inaccuracy of f (the normalized frequency is $f+\Delta f$). Based upon the determination of the modulo-time plot the difference between the values of the vector $\tilde{\boldsymbol{y}}_a$ and the ideal ones is equal to the difference between the ADC output signal samples and the ideal ones. The last difference is equal to

$$err_y(m) = 2A(m+m_0)\Delta f, \quad m = 0,1,2,\ldots,M-1. \qquad (15)$$

The squared rms value of the error err_y is given by

$$err_{yrms}^2 = \frac{1}{M}\sum_{m=0}^{M-1} err_y^2(m) - \left(\frac{1}{M}\sum_{m=0}^{M-1} err_y(m)\right)^2. \qquad (16)$$

After some calculus we obtain

$$err_{yrms}^2 = \frac{A^2(\Delta f)^2 M}{3}. \qquad (17)$$

To have very small errors caused by the inaccuracy of f, err_{yrms} must be μ times smaller than the standard deviation of the ADC quantization error

$$err_{yrms} = \frac{q}{\mu\sqrt{12}} \qquad (18)$$

where $q = FSR/2^n = 2A/2^n$.

From (17) and (18) it can be established

$$|\Delta f| = \frac{1}{\mu 2^n M}. \qquad (19)$$

Based upon (3) and considering $J + \delta \cong J$, the modulus of the relative error of the normalized frequency is given by

$$\frac{|\Delta f|}{f} = \frac{1}{\mu 2^n J}. \qquad (20)$$

To obtain small errors the value μ must be higher than 2, i.e. $\mu \geq 2$. This leads to the following constraint for the modulus of the relative error of the normalized frequency

$$\frac{|\Delta f|}{f} \leq \frac{1}{2^{n+1} J}. \qquad (21)$$

Due to the definition of the normalized frequency, at the limit situation we have

$$\frac{|\Delta f|}{f} = \frac{|\Delta f_{in}|}{f_{in}} + \frac{|\Delta f_s|}{f_s} \qquad (22)$$

where $|\Delta f_{in}|/f_{in}$ and $|\Delta f_s|/f_s$ are the modulus of the relative errors of the frequencies f_{in} and f_s.

We must point out that the constraint (21) is a necessary condition for the $|\Delta f|/f$ because the performances of the proposed method depend, as it follows from the step (4), upon the accuracy of the period T.

4. COMPUTER SIMULATION

The performances of the proposed method are analyzed by simulation. The ADC output signal used in simulation is

$$y(mT_s) = x(mT_s) + h(mT_s) + e_q(mT_s), \quad m = 0,1,2,\ldots,M-1 \qquad (23)$$

where: $x(mT_s)$ is simulated ramp input signal; for k_1 –line, $x(mT_s)$ is given by

$$x(mT_s) = 2A_1\frac{(J+\delta)}{M}(m+m_1) - A_1 - 2A_1(k_1-1), \quad k_1 = 1,2,3,\ldots$$

$h(mT_s) = h_2(mT_s) + h_3(mT_s)$, in which $h_2(mT_s)$ and $h_3(mT_s)$ are simulated second and third harmonics of $x(mT_s)$; for k_2-line, $h_2(mT_s)$ is given by

$$h_2(mT_s) = 4A_2\frac{(J+\delta)}{M}(m+m_2) - A_2 - 2A_2(k_2-1), \quad k_2 = 1,2,3,\ldots$$

and for k_3-line, $h_3(mT_s)$ is given by

$$h_3(mT_s) = 3A_2\frac{(J+\delta)}{M}(m+m_3) - 0.5A_2 - A_2(k_3-1), \quad k_3 = 1,2,3,\ldots$$

$e_q(mT_s)$ is simulated quantization noise.

The amplitude of the ramp input signal is $A_1 = FSR/2 = 2.5$. The number of the recorded ramp cycles is $J = 21$. The ADC tested has 8 bits. The number of the recorded samples is $M = 512$. The ramp input signal and the harmonics are characterized by $m_1 = 10$, $m_2 = 7$ and $m_3 = 3$.

The amplitude A_2 is established as a function of the parameter ENOB of the ADC under test.

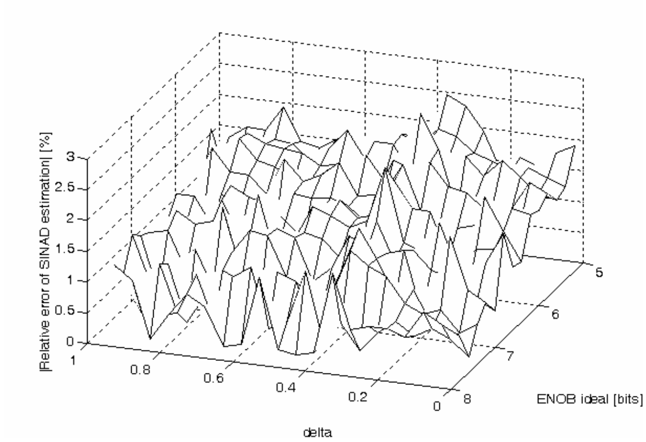
The ADC sampling period is $T_s = 3\mu s$.

It is assumed that the quantization noise is uniformly distributed and quantization errors from sample to sample are statistically independent.

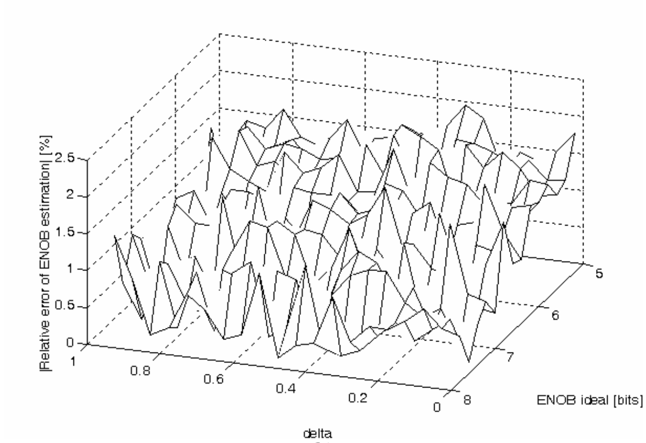
δ varies in the range [0, 1) with an increment of 0.05. ENOB varies in the range [5, 8) bits with an increment of 0.2 bits. A given ENOB value implies the amplitude value A_2.

The parameters SINAD and ENOB are estimated by the proposed method.

Fig. 2a shows the modulus of the relative error of the SINAD (ε_{SINAD} [%] = 100·|SINAD$_{est}$-SINAD|/SINAD) as a function of δ and ENOB. Fig. 2b shows the modulus of the relative error of the ENOB (ε_{ENOB} [%] = 100·|ENOB$_{est}$-ENOB|/ENOB) as a function of δ and ENOB.



(a)



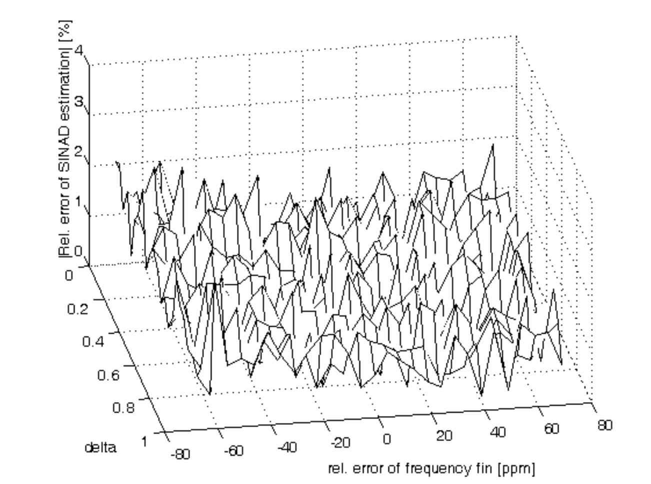
(b)

Fig. 2. (a) The error ε_{SINAD} as a function of δ and ENOB; (b) The error ε_{ENOB} as a function of δ and ENOB.

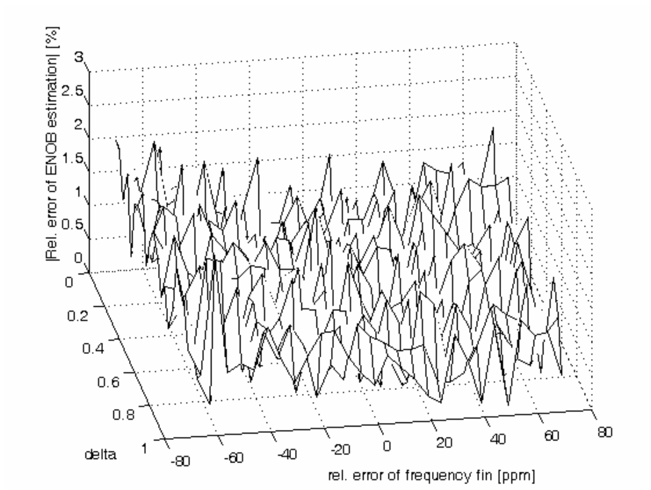
From Fig. 2 it follows that the parameters SINAD and ENOB are accurately estimated by the proposed method.

As precised above the performances of the proposed method depends upon the accuracy of f_{in} and f_s frequencies.

Fig. 3 shows the errors ε_{SINAD} and ε_{ENOB} as a function of δ and relative error of frequency f_{in} for the same ADC output signal (given by (23)), when ENOB = 7.4 bits and the relative error of frequency f_s is 20 ppm. The relative error of frequency f_{in} varies in the range [-70, 70] ppm with an increment of 5 ppm. This range was choosing because from (21) and (22) it follows that the $|\Delta f_{in}|/f_{in}$ must be smaller than 73 ppm to obtain high accurate results with the proposed method.



(a)



(b)

Fig. 3. (a) The error ε_{SINAD} as a function of δ and relative error of frequency f_{in}; (b) The error ε_{ENOB} as a function of δ and relative error of frequency f_{in}.

Fig. 3 shows that the method leads to accurate SINAD and ENOB estimates when $|\Delta f_{in}|/f_{in}$ is smaller than 70 ppm.

Based upon the results obtained in Fig. 3 it follows that the condition (21) is also a sufficient condition.

5. CONCLUSIONS

In this paper a time-domain method for estimating ones of the most important dynamic parameters of an ADC – SINAD and ENOB – by ramp testing signals is proposed. As it is follows from simulation this method leads to accurate SINAD and ENOB estimates when the ramp input signal frequency and the ADC sampling frequency are known with very high accuracy. A constraint imposed to modulus of the relative error of the normalized frequency is also derived to ensure that the dynamic parameters of an ADC are accurately estimated by the proposed method.

The proposed method does not require complex calculations. Moreover, the number of samples necessary to test all ADC codes is smaller in the case of a ramp test signal than in the case of a sinewave test signal. From these reasons the proposed method can be an alternative to the existent time-domain methods which use the sinewaves as test signals.

The main drawback of this method is that this requires a very high accuracy ramp generator.

ACKNOWLEDGMENTS

The work presented in this paper is performed under the type A project "Dynamic characterization and modeling of the A/D converters used in the high-speed data communications", theme 2, code 340, supported by CNCSIS România. The project involves collaboration between the Department Measurements and Optical Electronics of the Faculty of Electronics and Telecommunications, Timişoara and the IXL Laboratory - ENSEIRB - Bordeaux 1 University.

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