# Analyzing the linearity of ramp waveforms generated by means of AWG

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*Abstract:* - Highly linear ramp waveforms are important in a variety of industrial processes, control systems, and test and measurement applications. The generation of ramp waveforms can be accomplished in several ways by means of both dedicated circuits and arbitrary waveform generators; the latter ones also grant the widest configurability of the output waveform both in terms of time and amplitude parameters. But, the waveforms generated by means of arbitrary waveform generators are characterized by piecewise time domain evolution. The underneath piecewise frame always implies a reduced conformity of the generated waveform to the intended one, even if the waveform is smoothed by means of a low-pass filter integrated in the generation chain. Conformity analyses can be carried out both in time and frequency domain, and their results expressed either by means of an error waveform or by pointing out the undesired spectral contributions. Typically, the insights allowed by the spectral analysis greatly help in both understanding the causes of poor conformity, and documenting the precision of the output waveform. Hereinafter, first a general explanation of the mechanisms that originate the distortion contributions and reduce the conformity of a piecewise waveform to the intended one is given. Then, the attention is focused on ramp waveforms generated by means of arbitrary waveform generators and a thorough analysis of their conformity to an ideal ramp is carried out.

*Key-Words:* - Arbitrary waveform generator; Digital-to-analog converter; Direct digital synthesis; Waveform modeling; Linearity; Distortion analysis.

## **1** Introduction

Arbitrary waveform generators (AWGs) that operate by means of an internal digital-to-analog converter (DAC) represent the most versatile source for control and measurement applications in industry and research laboratories [1],[2]. They allow the user to define the desired waveform and reproduce it either in single shot mode, in order to have a transient stimulus, or in continuous mode, in order to have a periodic stimulus; they also allow to control the amplitude and time parameters that characterize the output waveform within a wide range of values [3],[4].

The operating principle of AWGs is nowadays largely exploited in the most recent hardware-in-theloop test-benches, adopted to test complex real-time embedded systems. These test-benches require to emulate the electrical output of sensors and actuators, which are acquired and processed by the system under test. Also, in electronic warfare applications based on digital radio frequency memory (DRFM) techniques, AWG-like architectures play a crucial role to create false targets to radars by re-transmitting radiofrequency signals that have been previously captured by means of a digitizer. Nonetheless, AWGs are largely employed as function generators to produce canonical waveforms [5]-[9].

The great versatility of AWGs is paid in terms of the distortion that could appear in the output waveform. The distortion usually consists in attenuated replicas of the double sideband desired spectrum, centered at integer multiples of the generation frequency. The use of a low pass filter, often integrated in the generation system and connected in series to the DAC, allows to attenuate the distortion whose spectral contributions are far outside the frequency band occupied by the useful waveform. As a general rule, it is always convenient to use a generation frequency sufficiently greater than the useful bandwidth, in order to move the distortion contributions at higher frequencies and improve the effectiveness of the smoothing operation. In the presence of constraints that limit the selection of the generation frequency, as well as

in critical applications that require a substantial reduction of the overall distortion, the user has to adopt dedicated approaches, such as the use of external filters, to prevent artifacts [10].

There are some special conditions in which additional distortion contributions can emerge. These contributions are centered at frequencies that are sub-multiples of the generation frequency, and can even fall within the bandwidth of the useful signal. A comprehension of the mechanism that originates them is definitely needed in order to design proper solutions to reduce their magnitude, and to thoroughly document the precision of the output waveform. At the best of the author's knowledge, the mechanism that originates the distortion contributions at sub-harmonic frequencies of the generation frequency has not been sufficiently taken into consideration in the literature. In particular, no functional relationships that links the magnitude and frequency of the distortion contributions to the length of the digital signal, given in terms of number of samples, and the number of digital codes utilized by the DAC can be found in the literature [11]-[13].

After overviewing the aforementioned concepts, the case related to the employment of AWGs as ramp generators is dealt with. Ramp signals are adopted in test and measurement widely applications. In particular, they are used to calibrate the response of sensors, transducers, and several other devices; to drive and/or conditioning the behavior of actuators by means of servo control systems; to activate large magnets adopted in MRI machines, NMR spectrometers, mass spectrometers, and particle accelerators. In several of these applications the conformity of the actual ramp signal to the ideal one is a critical issue. In particular, the generated ramp has to satisfy tough requirements in terms of linearity and should avoid to excite resonances at high frequencies [14],[15].

The paper is structured in order to initially provide in Section II a brief overview of the most common approaches to arbitrary waveform generation. Then a general framework for analyzing the distortion to be expected in the output analog waveform is presented in Section III. Starting from this general framework, the case of digitally synthesized ramp signals is discussed in Section IV. Finally, Section V provides concluding remarks.

## 2 Waveform Generation

Arbitrary waveform generators can be of three different types, namely: true-arbitrary generators, direct-digital-synthesis (DDS) generators, and

interpolating DAC generators. Actually, all the types share the same basic architecture that includes: a waveform memory to store a digital representation of the desired signal, one or more auxiliary registers to address memory cells and manage samples, a digital-to-analog converter, a master clock, and an output front-end device that consists in a smoothing filter and an amplifier circuit.

### 2.1 True arbitrary generator

The true arbitrary generator operates according to the most straightforward approach: the samples stored in the waveform memory are read one by one and converted into analog levels by means of a DAC operating at the master clock rate. A block diagram describing the architecture of a true arbitrary waveform generator is shown in Fig. 1. The user can control the period or the single-shot duration,  $T_0$ , of the output waveform by regulating the period  $T_{ck}$  of the master clock; in fact, for a waveform defined by means of M samples, one has the identity:

$$T_0 = M T_{ck} \tag{1}$$

The samples stored into the waveform memory are addressed by means of a pointer (PTR), and hence fetched and given as digital input to the DAC. PTR is programmed to increase by one at each clock period in order to perform a sequential scan of the waveform memory. PTR can be managed according to an *M*-modulus arithmetic, *M* being the number of samples stored in the waveform memory.



Fig. 1. Block diagram of the architecture of a truearbitrary waveform generator.

This is implemented when the user chooses a continuous generation mode: thanks to the M-modulus arithmetic, the increments by one automatically produce repetitive scans of the whole waveform memory. Several true arbitrary generators allow the user to employ only a portion of the whole waveform memory by choosing M from a limited set of values, typically including only power-of-two numbers. In this case, the waveform memory is managed by adjusting the length, in terms of

number of bits, of the PTR register, that according to the utilized portion has to be equal to  $lg_2M$ .

#### 2.2 DDS-based generator

The generators that rely on the direct digital synthesis (DDS) technique have similar architectures, as shown in Fig. 2, but work at a fixed clock period,  $T_{ck}$ , and exploit two auxiliary registers, namely accumulator (ACC) and phase increment register (PIR), to grant the desired period or single shot duration to the output waveform, otherwise fixed at the duration given by equation (1).

The DDS technique can be explained by considering the value stored in ACC and PIR as fixed point real numbers, consisting of an integer and a fractional part. The value of PIR is a function of the period or single shot duration  $T_0$  selected by the user and the fixed period  $T_{ck}$  of the master clock:

$$PIR = \frac{MT_{ck}}{T_0}$$
(2)

The PTR is mapped into ACC, and contains the integer part of the value stored into ACC. PTR addresses the memory location from which the digital input for the DAC is fetched. At each clock period the value stored in PIR is added to ACC, so that PTR is incremented according to the increase of the integer part resulting for ACC.



Fig. 2. Block diagram of the architecture of a DDS generator.

It is worth noting that the increments of PTR, caused adding PIR to ACC can produce sometimes a carrier digit that makes the increment greater than the previous ones. For instance, if the PIR value is greater than 1, and q is the integer such that q < PIR < q+1, the PTR value changes at each clock period, incrementing by q or q+1. Note that in this case not all the samples hosted in the waveform memory are addressed and fetched in a single scan, but only one every q or q+1. The DDS technique provides for the accurate balance between the coexisting decimation factors, q and q+1, during the waveform reproduction, to attain the required period or single shot duration.

If the PIR value is less than 1, named p the integer such that  $p < PIR^{-1} < p+1$ , a unitary increment is observed in PTR only after that p or p+1 clock periods have been elapsed. In other terms, adding PIR to ACC changes the fractional part of ACC, causing an increment of PTR only when a carrier digit is produced. The DDS technique provides for the adequate balance between the waiting clock periods, respectively p and p+1, to attain the required period or single shot duration for the output waveform.

#### 2.3 Interpolating DAC generator

The most recent and advanced solution for arbitrary waveform generation is represented by the interpolating DAC generator. A block diagram useful to describe the architecture of this generator is shown in Fig. 3. Here, the output waveform is built up by a DAC system that is clocked at a higher sample rate with respect to the waveform memory access rate. A digital signal processing (DSP) block interfaces the waveform memory to the DAC and makes nontrivial the up-sampling implied by the different operating clock rates. Specifically, the DSP block, by means of a stable and computationally efficient finite impulse response (FIR) filter, performs a real time padding and interpolation between adjacent samples fetched by the waveform memory.



Fig. 3. Block diagram of the architecture of an interpolating DAC generator.

The interpolating DAC generator exploits at best the vertical resolution of the internal DAC, that can be even finer than the quantization characterizing the desired waveform. This typically happens when the specified waveform consists in an acquired record related to a real waveform of an observed source that the user wishes emulating. Besides, the user has to accept that the desired waveform, downloaded in the local memory of the generator, is somehow refined through real time processing operations performed at the reproduction stage by the firmware of the generator.

## **3** Piecewise Waveforms Modelling

In this Section an analytical representation of an analog piecewise waveform produced by a DAC system for a given digital waveform s(n), aimed at representing a smoothed waveform s(t), is considered. It is always assumed that the sequence s(n) is causal, i.e. s(n) = 0 for n < 0. Also, the digital waveform is considered unlimited every time it represents the cyclical reproduction of the finite sequence stored in the local memory of the DAC system. Equally, s(n) is considered unlimited if synthesized in real time according to a given rule by a DSP block and given in streaming to the DAC system.

The digital waveform s(n) is instead considered finite if the generator is utilized in single shot mode operation.

The analog waveform produced by the DAC and observed in a limited time interval can be described in terms of a sequence of rectangular pulses, according to:

$$s_{DAC}(t) = rect\left(\frac{f_{ck}t}{L}\right)\sum_{n} s(n) rect(f_{ck}t - n)$$
(3)

in which  $f_{ck}$  is the generation frequency, rect(t) is a rectangular pulse equal to 1 in the time interval (0,1) and zero elsewhere, and the waveform is observed in a time interval characterized by duration equal to  $LT_{ck}$  seconds.

To analyze the distortion characterizing the waveform  $s_{DAC}(t)$  it is useful to evaluate its amplitude spectrum. By Fourier transforming equation (3) it is obtained:

$$|S_{DAC}(f)| = \left| \operatorname{sinc}\left(\frac{f}{Lf_{ck}}\right) * \left(\frac{1}{f_{ck}}\operatorname{sinc}\left(\frac{f}{f_{ck}}\right)\sum_{k}S(f-kf_{ck})\right) \right|$$

$$(4)$$

in which S(f) is the Fourier transform of s(t). Equation (4) reveals the presence of distortion terms consisting in attenuated replicas of the desired spectrum, S(f), centered at frequencies  $kf_{ck}$ . As well known the observation time impacts on the frequency resolution of the spectral analysis. If the observation time is sufficiently long, the effects of the limited frequency resolution can be considered negligible, and the spectrum of the piecewise waveform accurately approximated by:

$$|S_{DAC}(f)| = \left|\frac{1}{f_{ck}} sinc\left(\frac{f}{f_{ck}}\right)\right| \left|\sum_{k} S(f - kf_{ck})\right|$$
(5)

since the *sinc*(.) core function, in the convolution product in equation (4), converges to an ideal Dirac distribution  $\delta$ (.) as  $LT_{ck}$  approaches plus infinity.

Actually, equations (3) and (5) although formally right, are not suitable to highlight the presence of some additional distortion contributions. These are produced every time that the same value persists at the input of the DAC system for two or more clock periods. This occurs when either the digital sequence s(n) itself presents equal values in next cells in the waveform memory, or when the AWG generator operation keeps constant the value given in input to the DAC for one or more clock periods, as can happen in DDS-based generators. Whichever the cause, an exact analytical description should take into account the number of clock periods for which the same value persists at the input of the DAC system. To this end, equation (3) is reformulated using time-stamped rectangular pulses characterized by different durations. Also, the positive discrete function, a(m), defined for a subset of all the discrete time instants n, named  $A = \{m\}$ , is introduced. The time instants identified by the values of *m* timestamp the change of the DAC input; the function a(m) returns the number of clock periods to wait before observing the next change. The time instants m collected in A also timestamp the occurrences of the rectangular pulses that allow to represent the output of the DAC system as:

$$s_{DAC}(t) = \sum_{m \in A} s(m) \ rect\left(\frac{f_{ck}}{a(m)}t - m\right)$$
(6)

In equation (6) the time duration of the *m*-th rectangular pulse is  $a(m)T_{ck}$ , and an unlimited observation window has been assumed. Note that equation (6) includes equations (3) when the set *A* coincides with the set of integers, which implies a(m) = a(n) = 1.

## 4 Ramp Waveform

The use of arbitrary generators to produce ramp signals represents a relevant case study.

If the adopted generator is a true arbitrary type integrating a DAC with *N* bits, and if allows an arbitrary segmentation of the waveform memory, the user can conveniently choose to describe the ramp signal with a sequence made up of exactly  $C = 2^N$  samples, i.e. with a sequence containing just once all the admissible input values for the DAC. A portion of the analog waveform produced by the DAC in this hypothesis is shown in Fig. 4. The

greatest displacement from linearity is observed immediately before and after the DAC output level switching, and it is equal to half time the vertical resolution of the DAC. The error signal is described by a saw-tooth waveform with peak-to-peak value equal to the DAC vertical resolution.



Fig. 4. Portion of the analog waveform produced by the DAC (bold line) compared to an ideal ramp (dotted line).

According to equations (4) and (5), the amplitude spectrum of the piecewise ramp, shown in Fig. 5, is characterized by several replicas of the ideal amplitude spectrum of a ramp with finite duration. The replicas are centered at all the multiples of the generation frequency,  $f_{ck}$ , selected by the user; the higher the center frequency of the replica the more attenuated its magnitude. It is interesting to observe the effects due to the replicas, which consist in some reduction of the energy spectral density at low frequencies with respect to the ideal case and in the notches at the frequencies  $f_{ck}$  in Fig.5.



Fig. 5. Amplitude spectrum of a piecewise ramp.

To accurately describe the output of the generator one has also to consider the effects of the smoothing filter and that of the master clock stability. The first can largely contribute to reduce the distortion caused by the undesired replicas; it smooths and attenuates the saw-tooth error signal and gets the piecewise waveform approach the desired linear trend. The effectiveness of the filter is related to the attenuation that can be granted out of the band of interest, and principally at all the frequencies that are multiples of the generation frequency  $f_{ck}$ . It would be convenient to have a smoothing filter with stop-band characteristics tuned to  $f_{ck}$ . But, due to the general purpose character of the AWG, the user can only rely on the effects of one or a very limited set of selectable filters with fixed configuration and no tuning options. The stability of the master clock represents an effect of secondary importance although not negligible. The variable clock has to grant a wide choice of the timing signal, and cannot compete in performance with fixed frequency clocks. A timing error  $\tau$  due to a lead or delay of the clock pulse in the worst case can increase the displacement from linearity by  $\tau$ times the ramp slope.

It is worth noting that if in place of a true arbitrary generator a DDS-based AWG is adopted, the linearity of the output ramp can be analyzed taking into account, equivalently, that systematic timing errors in the master clock could be intentionally and repeatedly introduced according to a regular or quasi-regular pattern. The DDS-based AWG has however the advantage of using a fixed stable frequency.

The linearity of the ramp waveform can be substantially degraded when the adopted generator does not allow an arbitrary segmentation of the waveform memory. In this case the user is compelled at defining the digital signal using a given number M of digital samples. Typically M is larger enough to be greater than the number of DAC input codes, C, and allows avoiding decimation (M > C). Actually, if the ratio M/C is equal to an integer value Q, each digital code is consecutively found Q times in the sequence that defines the digital waveform. The output waveform can be described using equation (6) with a(m) = Q, and considering m as a subsequence extracted from nbeing  $A = \{m = Qn\}$ :

$$s_{DAC}(t) = \sum_{m} s(m) \ rect\left(\frac{f_{ck}}{Q}t - m\right)$$
 (7)

Equation (7) accounts for a piecewise waveform with all pieces characterized by duration equal to  $QT_{ck}$ ,  $T_{ck}$  being the constant period of the master clock of the generator. The amplitude spectrum of the generated ramp can be expressed by:

$$|S_{DAC}(f)| = \left|\frac{Q}{f_{ck}} sinc\left(\frac{Q}{f_{ck}}f\right)\right| \left|\sum_{k} S\left(f - k\frac{f_{ck}}{Q}\right)\right|$$
(8)

Equation (8) highlights attenuated replicas of the desired amplitude spectrum, which are centered at

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multiples of a sub-harmonic of the master clock frequency, namely  $f_{ck}/Q$ . Note that equation (8) has been derived from equation (5) and therefore does not take into account the narrow-band leakage effect

produced by the limited time duration of the generated ramp, which is in general highlighted by equation (4).

If the ratio M/C is not integer but R < M/C < R+1, any digital code in the sequence can be consecutively found either R or R + 1 times, being R implicitly defined by the aforementioned inequalities as the integer part of the ratio M/C. If D is the remainder after the division of M by C, the following identities are true:

$$M = CR + D = (C - D)R + D(R + 1)$$
(9)

The input codes can be therefore distinguished into two groups: the first one includes C - D codes that are repeated R times, the other group D codes that are repeated R + 1 times. The codes of the two groups are uniformly intermingled between each other, according to a uniform or quasi-uniform pattern. Specifically, the pattern is periodic if:

$$E = \min\{C - D, D\}$$
(10)

divides *M*, and its period is given by:

$$P = \frac{M}{E} \tag{11}$$

In particular, the pattern includes a set counting *L* codes of one group followed by a code of the other group. Two cases can be distinguished: if E = D, then one can identify in the digital ramp *L* codes repeated *R* times before a single instance of a code repeated R + 1 times. The period of the pattern satisfies equation:

$$P = LR + (R + 1)$$
(12)

according to which

$$L = \frac{P-1}{R} - 1 = \frac{\frac{M}{D} - 1}{R} - 1 \tag{13}$$

In the other case, E = C - D, the period is:

$$P = R + L(R + 1)$$
 (14)

according to which:

$$L = \frac{P - R}{R + 1} = \frac{\frac{M}{C - D} - R}{R + 1}$$
(15)

In the first case the output waveform can be analytically described in the time domain by:

$$s_{DAC}(t) = \sum_{i} \left( \sum_{l=0}^{L-1} s(i+lR) rect \left( \frac{f_{ck}}{R} t - i - lR \right) + s(i+LR) rect \left( \frac{f_{ck}}{R+1} t - i - LR \right) \right)$$
(16)

where the index of summation *i* timestamps the occurrences of each periodic pattern; *i* is a subsequence extracted from *n*, namely i = Pn. Similarly an analytical description can be given also for the second case.

The amplitude spectrum of the waveform is given by the Fourier transform, which, in both cases, cannot be expressed in closed form without approximations. Nonetheless, the two cases analyzed are related to very particular conditions, whereas in the general case there is no period P, and the lacks of regularity in the code repetition patterns have to be regarded as local non-stationarity features of the waveform. The Fourier transform cannot inform on non-stationary characters, but offers anyway an insight on the average energy spectral distribution. Taking into account these premises, the amplitude spectrum of the generated ramp can be approximated by:

$$\begin{aligned} |S_{DAC}(f)| &= \\ \frac{|(C-D)R|}{Cf_{ck}} sinc\left(\frac{R}{f_{ck}}f\right) + \\ \frac{D(R+1)}{Cf_{ck}} sinc\left(\frac{R+1}{f_{ck}}f\right) e^{-i\pi\frac{f}{f_{ck}}} \left| \left| \sum_{k} S\left(f - k\frac{C}{M}f_{ck}\right) \right| \end{aligned}$$
(17)

Equations (17) accounts for the undesired presence in the amplitude spectrum of the generated ramp of attenuated replicas of the desired spectrum centered at multiples of a fraction of the master clock frequency, namely  $Mf_{ck}/C$ . The weighting function that describes the attenuation is a linear combination of the spectral images of the rectangular pulses of time durations equal to  $RT_{ck}$  and  $(R+1)T_{ck}$ ; the linear combination uses as weighting coefficients the relative number of occurrences of the corresponding pulses in the time domain representation of the piecewise ramp. It is worth noting that if the ratio M/C is integer then D = 0 and equation (17) coincides with (7).

As an example the amplitude spectrum of a ramp waveform described by means of a sequence counting 1000 samples and generated by means of an AWG that uses an 8-bit DAC is shown in Fig. 6. It is worth noting the notches localized at frequencies  $Mf_{ck}/C$  that are multiple of a fraction of the generation frequency.

## **5** Conclusions

A study aimed at investigating the conformity of waveforms played by means of arbitrary waveform generators to the intended one has been carried out. It has been shown how the underneath piecewise character of AWG output always implies a reduced conformity, even if the waveform is smoothed by means of a low-pass filter. Conformity has been analyzed both in time and frequency domain. Special attention has been payed to ramp waveforms, widely used in control and measurement applications. Approaches to describe ramp waveform generated by means of AWG have been presented and discussed.



Fig. 6. Amplitude spectrum of a ramp waveform described by 1000 samples quantized wit 8 bits.

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