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A perspective on advanced signal generation techniques

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Abstract – **The functionalities of signal sources have kept the pace with the mounting complexity of the available systems. It could appear, that the types of signal sources nowadays available on the market allow the user to satisfy every need in test and measurement applications. On the contrary, despite the wide range of functionalities provided by the very last generation sources, there is still room to imagine challenges and propose hypothetical architectures of generators suitable to face them. The speculative case of architectures made up of multiple direct digital synthesizers (DDS) circuits that operate synchronously to produce non periodic signals and emulate interesting scenarios is discussed.**

I. INTRODUCTION

Signal sources are necessary equipment in measurement applications that allow to highlight the behavior of passive devices, assess their reliability and performance, and troubleshoot and diagnose their shortcomings. In the last century, a number of applications in several different fields have boosted up research and development actions that have led to the realization of very different signal sources [1],[2].

At the beginnings, the concern was to convert DC energy into alternate signals by means of oscillators, and consistently control frequency and amplitude parameters. The requirement related to frequency and amplitude precision soon spread into new claims for sources characterized by high amplitude dynamics, and sources that allowed the user to select the frequency value in very large ranges. Oscillators with modulation capabilities led to the rapid development and installation of communication systems [3],[4],[23],[24]; at the same time, measurement applications demanded oscillators with frequency sweeping facilities for impedance and spectrum analysis. Continuous waveforms generators started to be replaced in some applications, at first by function and pulse generators, then by arbitrary waveforms generators. Also, the more and more extensive use of electronic in military operations, and the huge investments to define approaches and realize systems to produce electronic attacks and/or countermeasures, significantly contributed to improve signal generation know-how: sources were required to generate noise or other forms of jamming to disturb enemies operations and communications [5],[6].

Today, a variety of applications enlisting digital communication systems, remote control, radar surveillance, satellite communications, and navigational assistance have set up tougher requirements concerning: capability of managing complex modulation schemes, flexibility to emulate sophisticated waveforms, programmability to reproduce manifold scenarios by sequencing different bursts [7]. The same applications are driving the attention on very specific performance features of signal sources, namely: phase noise, harmonics, spurious free dynamic range, tuning speed, ultra-wide output bandwidth, etc [8], [19]-[21].

As the complexity of systems and facilities has grown further and further, the functionalities of signal sources have supplemented in time. It could appear, therefore, that the types of signal sources available on the market allow the user to satisfy whichever need allowing the generation of arbitrary signals.

It can be argued, however, that despite the high degree of complexity reached by the very last generation sources, there is still room to imagine future challenges and propose hypothetical architectures of generators that overcome the remaining limits [9]. In this paper, the speculative case of architectures made up of multiple direct digital synthesizers (DDS) circuits that operate synchronously to produce non periodic signals and emulate seemingly chaotic scenarios is discussed [10].

II. ESTABLISHED ARCHITECTURES OVERVIEW

In this section an overview of the consolidated architectures utilized to generate signal stimuli is given.

A. Oscillating circuits

The generation of sinusoidal waveforms is ordinarily accomplished by means of a simple circuit consisting of an amplifier with a filtered feedback. This circuit has been largely exploited since the beginning of 1900 to attain sinusoidal signals both in the audio and radio frequency range. In order to let the circuit work as a free running oscillator, it is necessary that the adopted filter provides positive feedback at a resonant frequency, whereas the amplitude limitation for the oscillating signal

is obtained by exploiting the inherent saturation of the amplifier: as the input to the amplifier approaches the saturation region, the differential gain exhibited by the amplifier diminishes, thus limiting the amplitude of the oscillations [1],[2]. The requirements for low harmonic contents and high stability of amplitude and frequency parameters are matched by equipping the basic oscillating circuit with automatic gain control circuits and/or by accurately selecting the circuit topology, components and materials [11].

B. Function generators

Function generators are required to satisfy the interest/need of testing devices with non-sinusoidal input signals [1],[2]. Like the large majority of signal sources, function generators are oscillator-based circuits that offer the possibility of selecting different output waveforms, such as triangle, square, saw-tooth, exponential, etc. Different waveforms are obtained by means of nonlinear circuits that transform the periodic output provided by the local oscillator.

C. Frequency synthesizers

More sophisticated architectures chase frequency synthesis by combining signals all derived by a very stable reference oscillator [1]. They are exploited in frequency synthesizers that can generate frequencies which are multiple or fraction of the frequency of the reference oscillator. Frequency synthesis can be gained by means of either direct or indirect approaches. The direct synthesis is realized by assembling systems with frequency dividers, multipliers, mixers, and filters by means of which it is possible to combine signals characterized by different frequencies to obtain several frequency values. The indirect synthesis uses instead a voltage controlled oscillator, which is phase locked to the reference oscillator, to produce an output that is characterized by a frequency equal to m/n times the value of the reference. Frequency synthesizers that use the indirect synthesis offer the possibility of selecting the frequency value within ranges starting from fractions of hertz up to gigahertz, and also allow very fast frequency changes; differently, synthesizers that use the indirect approach show low frequency switching speed, but are characterized by a relatively simple architecture, and are consequently much less expensive that the former.

D. Sampled signals generators

Frequency synthesis can be also performed exploiting approaches that execute sampled signals generation. In detail, the values of one cycle of sinusoid are stored into a waveform memory, which is essentially a look-up table, from which they are cyclically withdrawn one by one at an assigned rate, established by means of a reference clock, and given in input to a digital to analog converter. A single register is utilized as pointer to address the memory cell from which the current sample has to be withdrawn and transferred to the converter. The pointer is programmed to increase by one at each clock cycle and, thus, to perform a sequential scan of the waveform memory. The length of the pointer, in terms of number of bits B, is linked to the size M of the waveform memory; commonly equation $2^B = M$ is satisfied. Due to the inherent M-modulus arithmetic for the pointer itself, when the user choses a continuous generation mode, the increments by one of the pointer automatically produce repetitive scans of the whole waveform memory. The output of the converter, which is a piecewise signal, is then smoothed in order to attenuate alias contributions produced by the continuous switching of the converter output.

E. Arbitrary waveform generators

Sampled signal generation has open the way to modern digital communications that largely employ in-phase and quadrature (I/Q) and orthogonal frequency division multiplexing (OFDM) modulation schemes; its inherent versatility is fully exploited by the arbitrary waveform generators utilized in measurement applications [12],[16]. The basic architecture of an arbitrary waveform generator includes:

- a waveform memory, which contains a digital representation of the desired signal;
- a reference oscillator or master clock;
- some auxiliary registers to address single memory cells and manage samples;
- a digital-to-analog converter (DAC);
- and an output front-end device that consists in a filter and amplifier circuit.

The digital blocks of the generator architecture, namely the waveform memory, the complementary registers, and the DAC, except for the latest technological solutions, operate synchronously, waiting for the master clock signal to update their status.

The arbitrary waveform generators can be distinguished according to the control logic exploited to play the sampled signal stored in the waveform memory.

In fact, in the true arbitrary generators the samples are addressed by a pointer and withdrawn one by one to be converted into analog levels at a sampling rate set by the user. A direct control on the frequency of the master clock allows a fine matching of the desired time domain specifications in terms of period or single-shot duration of the output waveform.

In alternative, the generator can work at a fixed sample rate and exploit the direct digital synthesis (DDS) technique. This technique allows the user to control the period or single-shot duration of the output waveform by means of two auxiliary registers, named accumulator and phase increment register, that control throughout the pointer the waveform memory addressing. Specifically, the phase increment register determines the pointer

increment, that does not occur for each clock signal and can also be different from one. In particular, to prolong the time duration the DDS technique makes the pointer increase only after that a number of clock pulses, equal either to k or at times $k+1$ has been elapsed. The same DDS technique provides for the adequate balance in the exchanging of the basic waiting times, respectively k and $k+1$ clock cycle periods, to attain the desired precision for the period or single-shot duration of the output waveform. To shorten the time duration, instead, the pointer is increased by an integer number equal either to l or at times $l+1$ so that not all the samples hosted in the waveform memory and representing the desired waveform are considered in a single scan, but only one of l or at times $l+1$; again the same DDS technique provides for the accurate balance between the decimation factors l and $l+1$ during the waveform reproduction to attain the desired precision for the time parameters of the output waveform. The use of a phase increment register with a relevant number of bits grants utmost resolution in the definition of the frequency and period duration value.

For the most recent and advanced solution, the output waveform is built up by a DAC 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 cares for processing and up-sampling implied by the different operating clock rates. In particular, the DSP block, by means of a stable and computationally efficient finite impulse response filter, performs a real time interpolation between the adjacent samples specified by the user. The interpolating approach grants fully exploitation of the vertical resolution of the internal DAC, that is finer than the quantization caused at the reproduction stage by the particular choice of parameters and/or the use of the DDS technique, and often even finer than that performed by the user at the synthesis stage. On the counterpart, 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 by the generator itself.

F. Software tools for signals creation

Software tools that accompany arbitrary waveform generators to support signals creation and customization, have become almost indispensable to keep the pace with the rapidly changing standards that define test and measurement tasks needed to verify modern systems. In fact, the very last generation systems claim for application-specific stimuli, whose generation requires a preliminary meticulous design phase, in which the user must configure several parameters, define specific test patterns, and properly organize the data. Finally, the user has to compile the program enlisting all the configuration actions. This allows to format the data in a suitable form to be downloaded into the waveform memory of the generator, that, finally, plays the desired analog signal.

III.   FUTURE ARCHITECTURES

A. Limits of arbitrary waveform generators

The arbitrary waveform generators available on the market operate as multifunction generators, support many modulation formats, and let the user customize the output waveform. Therefore they provide flexible solutions to accomplish the generation of the complex waveforms, which are necessary to comply with the more and more stringent requirements of the modern standards concerned with electronic systems test and measurement applications. Anyway, some classes of waveforms can still be outside the capability of the most versatile AWGs. One of these classes is identified by signals made up of different sinusoidal components characterized by arbitrary frequency values [9], [17]. Actually, the arbitrary waveform generators agilely generate periodic signals, which, as well known, can include several frequency contributions, all multiple of a fundamental frequency. Similarly, the transmitting sources of modern digital communication systems that implement OFDM modulation schemes, can generate signals including frequency contributions that are uniformly spaced in a frequency interval and belong to a predefined grid of values. Yet, if the frequency values of the contributions are arbitrary chosen, there is no way at present to generate such a compound signal other than: either (i) having an internal processor that, by evaluating a mathematical expression, computes in real-time the current value of the signal and provides the data stream to the converter as the samples are step by step calculated, or (ii) summing, by means of adequate combiners, the outputs of so many sinusoidal generators as the frequency contributions. For the first approach a substantial limit is represented by the sample rate characterizing the data stream, which could be much lower than the maximum sample rate of modern DACs; the second approach is hardware intensive and therefore expensive.

To clearly comprehend the aforementioned issue, the case of a periodic waveform with extra inter-harmonics can be considered [17],[25]. In this case a single complete cycle of the basic periodic waveform does not include an integer number of cycles of inter-harmonics. If this is neglected, and a sampled version of a single cycle of the periodic part of the signal is downloaded into the local memory of the arbitrary waveform generator, when the finite-length time record is replicated to attain a periodic version, the inter-harmonics originate a discontinuity at the end point, thus introducing energy leakage in the frequency domain. It can be underlined that, if the frequency are expressed at a finite resolution, there is a common period that is multiple both of the cycle duration of the periodic signal and of the cycle duration of all extra inter-harmonics. Anyway, in practical situations, this common period can be too long, thus, the digital

sequence that should represent the waveform will likely overflow the waveform memory. Using a very large size waveform memory, and putting a hard limit on the frequency resolution to coerce the compound signal to have a common period, could represent a pragmatic solution to avoid wrap around artifacts but could involve unacceptable approximations of the signal features.

B. Multiple access DDS and multi-DDS architectures

To bypass coarse approximations, new architectures can be considered. One of these requires the use of so many couples of phase increment and accumulators registers as the number of frequency contributions that must be included in the compound signal; a similar conceptual architecture has been also presented in [22] as a viable solution for digital beam forming systems. An operating cycle can be defined such a way that each couple of registers is activated in sequence and takes the control of the pointer to withdraw a single sample at time from the waveform memory. The samples are buffered up to the completion of the operating cycle, then they are linearly combined in order to attain the value of the multi-tone signal to be given in input to the DAC. Proper countermeasures have to be undertaken to avoid calculus overflow without excessively penalizing the vertical resolution. Although this architecture cannot fully exploit the maximum sample rate of the DAC, it represents a good compromise between the cheap and slow architectures that work in streaming, and the expensive and hardware-intensive ones. Alternatively, to improve the speed of the data stream in input to the DAC, multiple-bus solutions capable of granting concurrent accesses to the waveform memory can be designed, or even more supplemental synchronous DDS synthesizers can be deployed in the same system. Hybrid solutions that merge sequential and parallel approaches should be also experimented.

C. Software solutions for parallel DDS architectures

An architecture that uses multiple coordinated DDS synthesizers should likewise be complemented with software tools to assist the user in the correct and optimal exploitation of the hardware resources. In this case the software is in charge of highlighting signals features that appear in different domains with respect to that in which the signal itself is created. As an example, for a multi-tone signal, which is straightforwardly defined in the frequency domain, the software can help in highlighting the time domain features in terms of crest factor or cumulative distribution function of the amplitude values; the latter feature cannot be immediately grasped if the frequency of the contributions forming the signal are arbitrary chosen by the user. A very interesting facility of software solutions could be related to the calculation of the zero-crossings instants, which for multi-tone signals containing arbitrary frequency are, in theory, deterministic but not analytically computable. Such a signals could support, in conjunction with zero-crossing detectors, controlled non uniform high-rate sampling schemes.

As an example, the case of a dual-tone signal is briefly discussed. The zero-crossings characterizing dual-tone signals can be analytically solved when the two contributions are characterized by the same amplitude. In this case, let $s(t)$ be defined as:

$$s(t) = A\sin(2\pi f_1 t) + A\sin(2\pi f_2 t) \tag{1}$$

in which $f_2 > f_1$, and let us also consider the values:

$$f = \frac{f_1 + f_2}{2} \tag{2}$$

$$\Delta f = \frac{f_2 - f_1}{2} \tag{3}$$

The signal can be thus represented in terms of f and Δf as:

$$s(t) = A\sin(2\pi(f - \Delta f)t) + A\sin(2\pi(f + \Delta f)t) = \\ = 2A\sin(2\pi f t)\cos(2\pi \Delta f t) \tag{4}$$

Its zeroes occur at the time instants:

$$t = \left\{ \frac{k}{2f}, \frac{(2k+1)}{4\Delta f} \right\} \quad k \in \mathbb{Z} \tag{5}$$

where Z is the set of integers.

A digitizer triggered to sample the input signal on the occurrence of the zeroes of the dual tone signal, would accomplish a non-uniform sampling, as if it would be driven by a time base that is made up of two interleaved pulse trains: one characterized by a high pulse rate, equal to $2f$, the other by a low pulse rate equal to $2\Delta f$.

For the general case of multi-tone signals, the pattern characterizing the zeroes can be determined by means of dedicated software tools throughout simulations or approximate calculus formulas.
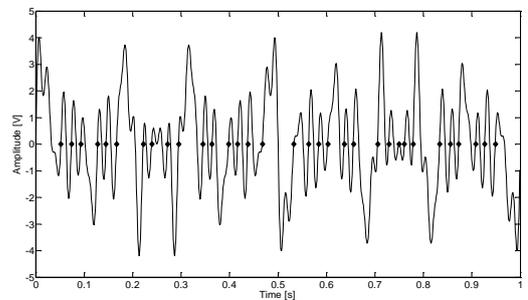

Figure 1. Multi-tone signal made up of 6 components characterized by the frequency values (7, 13, 17, 23, 41, 55) Hz; the black diamond markers highlight the positive-slope zero crossings.

IV. APPLICATIONS

Multi-tone signals are really claimed for to efficiently test devices and components that operate in wide bandwidths. In these measurement applications the residual nonlinear effects introduced by the equipment under test are addressed and quantified in terms of second and third order intermodulation contributions.

Other noteworthy applications, which could require the signals produced by the conceptual architectures presented above, involve the assessment of the performance, in terms of frequency resolution and robustness with respect to leakage phenomena, of advanced spectral analysis instruments. The performance of the majority of approaches discussed in the literature are, in fact, exhaustively characterized by means of simulations, but rarely experimentally confirmed, and whenever they are, the experiments are carried out by means of sources which allow poor control of test conditions and settings.

In general, playing signals consisting of frequency contributions that are very close to each other allows the emulation of electric, mechanical, and nuclear systems admitting multiple hyperfine oscillating modes. Nonetheless, the possibility of almost arbitrarily select the frequency of each tone allows the generation of signals that can be considered aperiodic for any practical time interval duration. Aperiodic signals can reproduce uncommon challenging scenarios to test equipment or measurement approaches.

Beside some applications for encrypted communications, the generation of aperiodic quasi-chaotic signals is useful to derive pseudorandom number sequences by uniformly sampling the aperiodic signal.

Last but not least, modern compressive sampling approaches often require sampling time-bases to perform non-uniform samplings. It can be shown that a signal including two or more components characterized by non-harmonic frequencies, in conjunction with a zero crossing detector, can be useful to the purpose. In particular, the use of an analog zero detector permits to overcome both the limitations in terms of finite time resolution that characterize the non-uniform time-bases realized by introducing quantized delays upon a regular clock, and the constraints that characterize phase and frequency modulation approaches to attain a controlled non uniform clock signal.

V. CONCLUSIONS

After an overview of the consolidated technologies and related architectures, currently implemented to realize signal sources, the work has presented a perspective on likely future advanced signal generation techniques. The attention has mainly been paid to parallel architectures including more DDS synthesizers, whose coordinated use can be exploited to produce signals that are difficult to generate by means of the sources available on the market.

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