Non-linear Components for Mixed Circuits Analog Front-End

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Abstract

This paper presents the development of some front-end analog circuits for mixed signals systems. The paper proposes the use of externally linear, internally non-linear analog circuits. Using this approach, analog area is greatly reduced, and circuits can be built on top of completely digital technologies. Experimental results in the analog and digital domain support the proposed approach to mixed circuits design.

1. Introduction and motivation

The analog fronted of mixed signal systems is among the most important applications of analog circuits one can find. Even with the increasing use of digital signal processing, there are some functions that can not be done in the digital world, like low noise amplification, anti-aliasing filtering and data conversion.

Figure 1 shows a typical front-end topology for the type of mixed signals applications we wish to cover. From a real continuous process one obtains a signal that generally has to be amplified, since sensors rarely give the required voltage or current levels. As most of the signal processing is done in the digital domain, an anti-alias filter is absolutely required, to assure that only the signals of interest are entering the system. Finally, depending on the speed and resolution of the required converter, a sampler might be needed to maintain the resolution while working with fast signals.

At the same time, mixed signal systems should be designed using a digital technology, since complete system integration commonly requires a microprocessor, memory and eventually a specialized digital function developed with an ASIC. Digital technology is also more often advanced than the analog one, allowing smallest transistors dimensions and requiring fewer masks.

Using digital technology, however, one looses important components. Resistors and linear capacitors require extra processing steps, and the area occupied is meaningful. Instead of using linear devices, a mixed signal circuit could be built with non-linear components, although with an external linear behavior. There are available different examples of the use of externally linear, internally non-linear systems, like the Mosfet-C filter, using the MOS transistor as a linear resistor ([1-3]). With the same non-linear VxI characteristic of the MOS transistor a linear current division technique can be obtained [4]. Biasing of non-linear MOS gate capacitors can result on a linear capacitor, as shown in [5]. A revision of non-linear techniques applied to linear circuit design is presented in [6]. Every non-linear behavior compensation scheme is based on a certain cost. This can come in the form of area, speed or design time, since a complex circuit requires more design time.

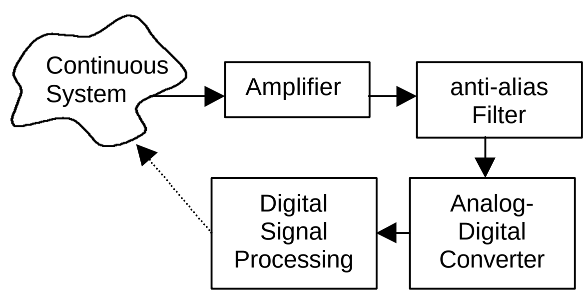

Figure 1 – front-end analog subsystem for embedded applications

This work shows the realization of analog circuits using non-linear components, together with digital post-processing. In this approach, there is no restriction regarding the origin of the non-linearity. Moreover, the non-linearity can be present in different components (resistors, capacitor, sensors, and operational amplifiers) with various VxI curves. The design technique can be either sampled or continuous, using pre-defined cells or using a sea-of-transistors approach. Meaningful area gains are obtained, when this kind of circuit is compared to its linear counterpart.

With the techniques presented here, one intends to save not only area on the analog part, but also to speed design times, by the use of simple circuits, done on top of a digital technology. Moreover, the design of the digital part can be readily achieved by the use of synthesis packages or by programming a microprocessor.

This paper is organized as follows: section 2 presents our target system, section 3 discusses the theory behind the influence of MOS non-linearities on the design of analog continuous time circuits like integrators. Section 4 presents the proposed design methodology, based on non-linear adaptive filters, followed by experimental results presented in section 5. Finally, section 6 presents our conclusions and future work.

2. The target application

A mixed signal system could be modeled as shown in figure 1. Some physical energy is transformed into electrical energy, and an amplifier and signal onditioning system must precede the A/D converter. Most part of the signal processing is done digitally, using a microprocessor. For control applications, there might be a feedback loop using a D/A and other analog conditioning circuits. The front-end circuit of figure 1 could model a large class of mixed systems, mainly those directed at embedded applications.

However, most sensors produce a signal that is not completely linear. The linearization of sensors can be developed by different techniques ([7-9]). Still, the sensor is not the only problem. Most sensors produce a weak electrical signal, which must be amplified. Since an analog to digital conversion will take place, an anti-alias filter must also come into play, before the converter itself. Any sensor non-linearity will be masked by amplifier or filter non-linearity before it can be compensated in the microprocessor. In the proposed approach, one can compensate any possible non-linearity present in the signal path, composed of sensor, amplifier and anti-alias filter.

This work proposes a mixed system based on a set of non-linear devices, capacitors and resistor plus switches, together with operational amplifiers of limited gain and bandwidth. There should also be available at least a couple of A/D and D/A converters, and a programmable hardware part to implement a digital filter. With this methodology, the arrangement of non-linear devices is not important, neither the design technique, since continuous or sampled time analog sub-system are supported.

3. Theoretical background

3.1. Limitations of digital technology

The amplifier and filter front-end of the mixed signal system are problem dependent. Sometimes the amplifier can also be used as a filter, like in the case of accelerometers, due to their noisy nature [10]. In this case, the anti-alias filter is mixed with the amplifier, since the effect is to limit the pass band of the sensor response. With strain gauges, the excitation is composed of few frequencies, and generally the acquisition system excites the sensor with a pure sinus [11]. The design of the anti-alias filter is dependent also on the used converter. The closest to the Nyquist rate the converter works, the higher the complexity of the anti-alias filter.

For amplifiers and anti-alias circuits, discrete implementations are common and readily available in the literature [12]. However, the silicon realization of these circuits is not as simple. Although a high gain amplifier can be achieved, its use to build an amplifier with linear and controlled gain is a hard task. Linear resistors and capacitors take to much area, and the other option is to use non-linear components like the MOS channel (as a resistor) or gate (as a capacitor). Besides that, there is also some non-ideal behavior in the operational amplifiers themselves, like limited gain and pass band.

3.2. Effects of non-linearities on the signal

Let us take as an example a simple integrator, implemented using non-linear devices. One can imagine a non-linear capacitor, with its capacitance depending on the voltage across it in a linear manner, obeying the law $C=f(v_C)=Co+K1*\ v_C$. In this case, assuming that the operational amplifier gain is large enough to put the negative input in virtual ground or near that, the capacitance C is a function of the output signal. If one considers a linear resistor, by solving the differential equation one obtains equation (1), showing that the output has a cross product between the input and the output. Alternatively, one can consider the resistor as $R=g(v_R)=Ro+K2*v_R$, and the capacitor linear, and the result will show a term depending on the square of the output, like in equation (2).

$$vo = -\frac{1}{Ro \cdot C}\left[\int vi \cdot dt + C \cdot K1 \cdot vi \cdot vo\right] \qquad (1).$$

$$vo = -\frac{1}{Ro \cdot C}\left[\int vi \cdot dt + R \cdot K2 \cdot \frac{vo^2}{2}\right] \qquad (2).$$

In case both components are non-linear, the output will have a large harmonic distortion, even for such simple non-linear characteristics. Although the above analysis assumes imaginary components (MOS transistors have more complexes RxV and CxV curves), the overall effect of a non-linearity is to generate harmonics at the output. To these harmonics problem one can also add the effect of limited opamp gain and pass band, as well as static characteristics like offset. The goal of the digital compensation is to restore the frequency behavior of the output signal without harmonic distortions.

4. Externally linear, internally non-linear circuits

4.1 Characteristics

The simple mathematical study developed in section 3 showed that any approach of using MOS devices as resistors will introduce non-linearities in the signal path, and a correction circuit will always be needed. However, this would introduce an area overhead. However, even when using linear components one pays the area price either as capacitor area or as the area increase required by an older process supporting 2 polysilicon layers. Any other compensation scheme developed at the analog domain also requires compensation with area overheads. For example, in [1] the area of the compensation circuit for a linear MOSFET-C filter is roughly 23% of the analog part, and the linear capacitor area adds another 23% of the total circuit area. Hence, huge area gains can be expected by the use of non-linear circuits in the analog part of mixed-signal systems.

In order to cope with these problems of larger area and parasitic non-linearity, a new class of circuits should be used, with externally linear, internally non-linear behavior. Non-linear analog circuits, although still not as characterized as linear ones, do have some advantages, like:

- one can develop the circuits in less area, since MOS gate capacitor is well controlled and with a high capacitance per area;
- one can use MOS channels as resistors, also saving area;
- one can use switched capacitor circuits using non-linear capacitance as well;
- one can use a state-of-the-art digital process to develop a mixed signal system.

4.2 The compensation methodology

Digital compensation proposed in this work is based on adaptive digital filters. The technique is very simple and easy to implement, either as a digital circuit or as some software code in a microprocessor [13-15]. Adaptive filters are generally designed to work with linear systems. In most practical applications, however, some non-linear behavior is quite common to be found. Some designers choose not to consider the non-linearity, or limit the dynamic range of the signal so that they are not meaningful. There is, however, some work on the design of adaptive non-linear filters, like [16-18]. Although mathematically complex and harder to design, there are available interesting results of their use in control and communication systems.

In this work one has used a non-linear adaptive LMS filter, modified from [18], and reproduced in figure 2. The main modification concerns the filter topology, where the non-linear behavior comes from the powers of the input signal in each filter tap. The presence of the third and fourth harmonics in the filter itself eases the compensation task. An important point regarding adaptive filters is the need for training before its operation in regime. This scheme is proposed in figure 3(a). While working in regime (figure 3(b)), all circuit characteristics have been trained in the filter, and the filter compensates for the non-linear behavior of the analog circuit.

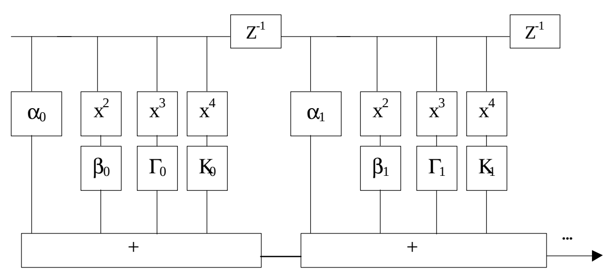

Figure 2 - Adaptive non-linear filter

It is important to notice that one must have a DA converter to excite the filter and the analog circuit (so that the excitation is known in the training phase). This converter must also have enough pass band to guarantee that the set filter-circuit is excited with all frequencies of interest. However, if in the mixed signal system some actuation is required, a DA will be naturally present, and its use during the training phase could be a minor cost. In the case of programmable mixed signal systems, the overhead would be tolerable, thanks to the extra programmability feature. The training signal must be rich in frequencies to excite all poles and zeros of the analog circuit, and this can be achieved with white noise.

5. Practical results

5.1 Digital compensation with a DSP processor

This section presents some practical results. A strongly non-linear circuit was assembled and compensated with a TMSC25 locally developed board. The circuit is presented in figure 4, and thanks to the diode, a strong non-linearity is present, ranging from the second to higher harmonics. As a first step, training and operation are performed to a single sinus, like in a strain gauge operation, for example. The measured Fast Fourier Transform of the output signal is presented in figure 5, together with the signal after linearization. As it can be seen in the frequency domain, the non-linear filter cancels out all harmonics inserted by the non-linear amplifier.

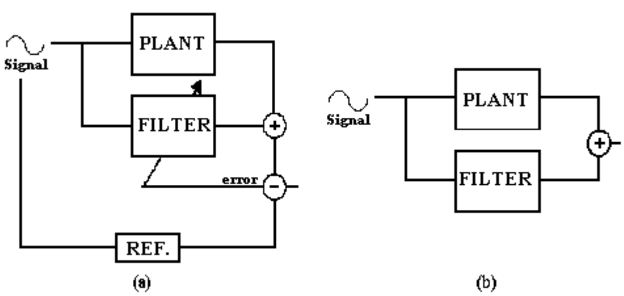

Figure 3 - Training of the filter (a) and operation in steady state (b)

Figure 6 presents the measurement of the same filter response now in the time domain. Notice the correct compensation of amplitude and phase of the signal. One can notice the high non-linearity of the output, mixed with a DC offset. There is an excellent match between the filter output and the desired signal. Even when trained with a large spectrum signal, like white noise, the filter maintains its correction capabilities, as presented in figure 7.

The adaptive filter was developed with few words of code of a TMSC25 processor running at 40MHz. It is interesting to notice that the number of non-linear coefficients is two-thirds of the total number of coefficients. Figure 8 presents the used integrator and acquisition boards (with AD and DA converters) and the C25 based processing board.

5.2. A dedicated digital hardware

A digital filter able to compensate non-linearities of the proposed example was built using Altera 10k20 FPGAs [19]. This was done to evaluate the amount of area needed for embedded applications, for example. A total number of 8 taps was implemented, and each coefficient was a full non-linear tap (x, x^2, x^3 and x^4). The total area of the filter was of 518 Logic Cells, or roughly 8k gates, with a maximum sampling frequency of 70 KHz for a 10 MHz clock. It should be noticed that, presently, even a common FPGA could be used to prototype such a filter.

5.3 Analog area gain

Using the proposed approach, one could reach 50% savings in area when comparing the analog area in a process supporting linear devices. When the comparison is made with a purely digital technology, the area in the analog part could be reduced by a factor of 40 (capacitance metal1-Poly) or 60 (Metal2-metal1). If a microprocessor is not available, the area of the digital filter must be taken into account for the global area computation. However, the more digital is the signal processing, the more easily automated is the design of the mixed integrated system.

6. Limitations, future work and conclusions

Although still without an integrated VLSI system, this work has shown some promising results. The compensated test cases can configure the set amplifier and filter as a sensor system. There are however still some open points. For example, the digital filter dissipates more power in comparison with a complete analog solution; on the other hand, the microprocessor is generally available in integrated mixed signal systems, and could be used for this task. In [20] some digital filters are compared with their analog counterpart in terms of power dissipation, and the result is that the power of an analog filter grows with the number of taps required, as well as the digital power dissipation decreases with technology scaling.

Another issue is the area occupied by the digital adaptive filter. The area trade-off must be developed at system level. This way, although the inclusion of a signal processor increases the used area, the analog part is reduced by the use of non-linear components. A simple non-linear capacitor is more then one order of magnitude smaller than its linear equivalent in a digital process. Moreover, the microprocessor already present on the intelligent sensor can help on the task of non-linear compensation. One must also consider that, even when using linear capacitors and MOS arrays as non-linear resistors, some internal circuit compensation is always present. For example, in [1] the area of the compensation circuit for a linear MOSFET-C filter is roughly 23% of the analog part, and the linear capacitor area adds another 23% of the total circuit area. Hence, huge area gains can be expected by the use of non-linear circuits in the analog part of mixed-signal systems.

The use of non-linear adaptive filters is limited, in the sense that, during the training phase, the plant and the filter must be excited with white noise. Although this is easily achieved when electric energy is involved (amplifiers, filters, transformers, strain gauges), the generation of a white noise excitation might be troublesome in other situations, like, for example, temperature sensors. Our future work involves the expansion of this methodology to cope with some non-linear characteristics of the AD converter as well.

The techniques presented in this paper are a clear step in the direction of mixed signals system-on-chip devices. Although the digital part and microprocessor can nowadays be easily synthesized with commercial tools, analog automation is still difficult. The presented compensation methodology can ease the task of the automation process by the use of simple textbook circuits. Moreover, the use of non-linear devices allows area savings, and allows the use of the latest state-of-the-art digital process.

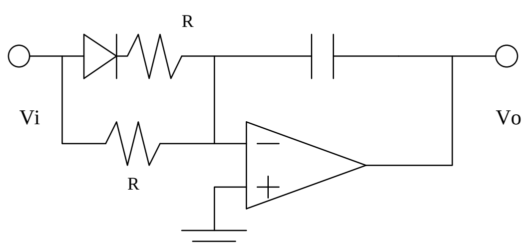

Figure 4. Non-linear integrator used for validation

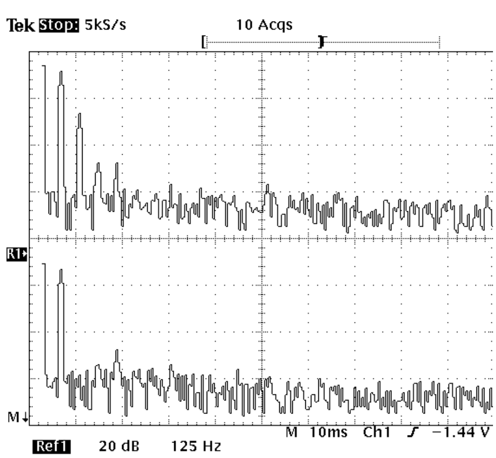

Figure 5 - FFT of the non-linear integrator (R1) and the compensated output (M). Notice the cancellation of second and third harmonics

7. References

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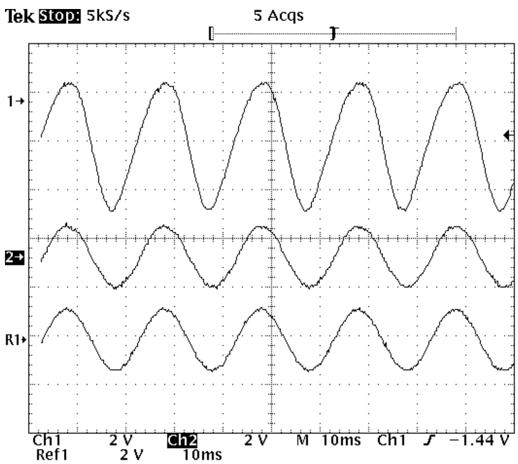

Figure 6 - Time domain results of the plant trained with a sinus. Waveform 1 is the non-linear output, notice distortion at peaks and the DC component. The bottom curve is the reference signal, while in channel 2 the compensated signal is presented

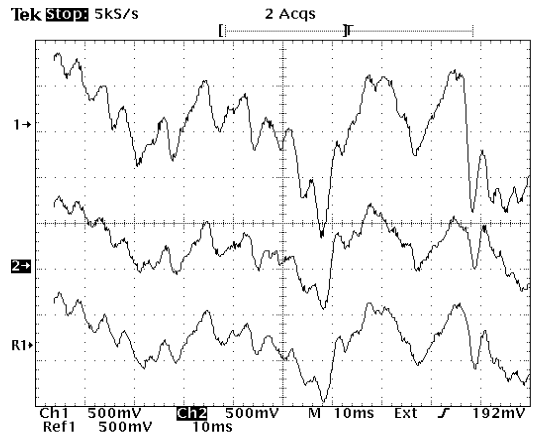

Figure 7 - Results of the plant trained with white noise.

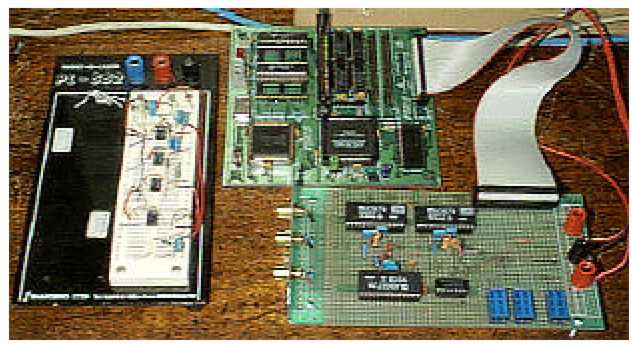

Figure 8 – From left to right, clockwise, the non-linear integrator, the C25 processing board and the acquisition board.