

The Rebirth of Computer Music by Analog Signal Processing

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ABSTRACT

Digital computation continues to offer spectacular solutions to signal processing problems at a rapidly decreasing cost. However, advances in VLSI design and manufacturing technology are not specific to digital processing systems and great cost/performance advances are also available from analog VLSI circuits. Indeed, many problems are solved most cost-effectively by using "mixed signal" technologies that allow analog and digital components on one chip. This paper explores the possibility that musical sound synthesis, processing and control systems would be better served by significantly more analog processing than the contemporary, mainly digital approach. An analog signal processing architecture is proposed consisting of a memoryless, multidimensional nonlinear function generator, a linear time-invariant system simulator, a non-volatile store, and interface, calibration, and control modules. This architecture is designed for real-time simulation of nonlinear dynamical systems, especially those that characterize acoustic musical instruments.

INTRODUCTION

The goal of this paper is to encourage a re-examination of analog signal processing techniques for computer music applications in the context of mixed signal VLSI technology and new results in nonlinear models of musical instruments. The motivation for this is not a nostalgic return to the sounds and machinery of the past. Digital computers and music synthesizers displaced their analog counterparts for a good and practical reason: It was cheaper to exploit developments in component and integration technologies with digital signal processing rather than analog techniques.

In the twenty-five years since the transition from analog to digital signal processing, there have been considerable advances in analog VLSI technology, design techniques and circuits. Many of these developments are simply a beneficial side effect of advances in digital technology. Other developments stem from the increasing importance of specialized applications where analog signal processing has been the dominant method, e.g. R.F. (radio frequency), video and neural networks [Mead 1989]. These technological developments make it possible to realize analog musical sound synthesizers that are potentially cheaper to manufacture and easier to use than digital sound synthesizers.

ANALOG AND DIGITAL SIGNAL PROCESSING COMPARISON

Introduction

Digital computers offer very accurate and reproducible results by time-multiplexing the use of a small number of high resolution arithmetic units using a very accurate quartz crystal timing reference. High component densities are possible because small transistors can be made to switch reliably between two states. However, as device sizes shrink and their switching rate goes up, the overhead associated with switching and moving data increases; this impacts system design significantly. In current computer architectures, the major design difficulty at all system levels is associated with moving data. On the other hand, basic processing elements represent a small part of design effort and system cost. With analog systems, conversely the problems and systems cost associated with connecting computing nodes are minor compared to those associated with designing the basic computing elements to meet performance requirements in the areas of component parameter spread, signal/noise ratio, distortion and crosstalk. Therefore if analog circuits can be designed to meet applications requirements, a mainly analog system will have lower system costs than a digital one.

Cost of Time-multiplexing

Current digital computer systems are built with a memory hierarchy of as many as 6 levels: pipeline registers, multiport register array, primary cache, secondary external SRAM cache, DRAM store and non-volatile rotating media. The first three levels often consume more than half the die area of a processor chip. It is instructive to estimate how much of this area is used to store parameters that are strictly required by an application compared to the amount required to support the digital signal processing paradigm.

A simple application example is an array of second order filters. Such arrays have proved useful for a variety of audio and music synthesis applications, e.g. [Potard et al. 1986] and [Lyon and Mead 1988].

A parallel bank of 2-pole digital filters is a standard module in the HTM DSP design environment [Freed 1992]. This module has been used to implement resonator banks and formant filters. Each filter in the bank requires the storage of two output state variables and 3 coefficients. A bank of 100 filters therefore requires storage of 500 values. Musical applications require that the 3 filter coefficients be computed in real-time from higher level triplets: (frequency, bandwidth and energy). A digital implementation of this filter bank on a modern RISC processor, such as the R4000 [Heinrich 1993], is straightforward but requires a surprisingly large amount of memory. The following inventory details the memory required beyond the 500 values fundamental to the computation itself:

Because of finite bandwidth constraints, the computation of filter coefficients from high level control triplets requires computation of sine, cosine, and exponential functions [Tang 1991]. Large tables stored in relatively expensive secondary cache memory are required to meet precision and real-time performance constraints.

A program needs to be stored to describe how to move the data between memories and basic arithmetic computational elements. To amortize loop variable overhead and keep pipelines full, loop code is unwound (replicated) 4 times [Kastens 1990].

To keep data in the fastest locations in the processor, namely the pipelines and registers, each filter operation is vectorized, thus necessitating extra storage for the input and output vectors.

To amortize bus access overhead, input and output data are stored in buffers to be moved (by DMA) from A/D and to D/A converters, respectively.

Packaging Cost

The cost of the 100,000 memory locations required by this example application extends beyond the die area used to store the values. Moving data between packages requires high-speed line drivers, a high pin count package, and multilayer circuit boards. One or two hundred pins is typical for current processors. Many of these are additional power pins for the line drivers.

Power Consumption

Commonly, more than half of the power consumed by a processor chip is for communication to external memory and peripherals [Barrière et al. 1990]. Of the rest, most is used to move data between the elements of the memory hierarchy and the arithmetic units.

Noise and Interference

The high speed and power levels required for communication between chips create other problems, e.g. EMI (electro-magnetic interference) which requires filtering and shielding, and ground bounce which requires decoupling and anti-latchup circuitry.

This example of a digital solution to a signal processing problem shows that most of the engineering effort is expended in supporting the flow of data to a very small part of the system. Admittedly, more streamlined and specialized digital processor designs are possible for this application, e.g. [Wawrzyniec 1989]. However, they still rely on switching and data flow and have little provision for efficiently computing trigonometric functions.

Analog Processing

In contrast to the digital solution, consider the aforementioned memory and interface requirements in the context of analog arrays of second order filters. If the universal biquad scheme for OTA-C (operational transconductance amplifier-capacitor) filters of [Khan et al. 1991] is used, no trigonometric functions are required to map the control triplets since frequency and bandwidth of these OTA-C filters are independently and linearly related to circuit transconductances. With digital filters there appears to be no way to avoid the trigonometric computations--although ingenious tabling (at the cost of memory) is possible [Nikolaidis 1993]. Independent control of frequency, gain and amplitude in digital filters requires a normalized ladder filter with 4 times the computational cost of the corresponding direct form filter [Massie 1993].

Since continuous-time analog OTA-C filters are multiplexed in space rather than time, no program memory, instruction cache and decode, or input/output vectors are required.

No D/A and A/D converters or associated FIFO buffers are required in an entirely analog system. Special conditioning is required on the input and output pins but fewer pins are used than on a digital processor since each signal requires only a single wire instead of a bus. There are no EMI problems because bandwidth and energy of audio analog signals are low.

Also, there is no need to store data in external memory chips since they may be readily stored on capacitors in the analog processor chip.

As a result of these economies and use of transistors in the weak inversion regime the power consumption of analog filter arrays is easily 1/100th that of digital solutions [Lazzaro et al. 1994].

Of course, analog signal processing circuits have problems not found in digital signal processors. These problems caused analog methods to be mostly abandoned for musical applications in the 1980's. These problems will be considered in the following sections.

Cost-effective Design

Cost-effective design tools are required to realize VLSI analog circuits. Such tools exist for OTA-C filters [Daasch et al. 1992].

Linearity and Signal/Noise Ratio

The acuity of human auditory perception necessitates the use of analog circuits with good signal/noise ratio and high linearity. [Durham & Redman-White 1993] report a very high linearity OTA in 5V CMOS confirming that these quality requirements can be met in modern analog processes.

Parameter Spread

It is the accuracy of pitch perception that poses one of the greatest challenges. Parameter spread in analog circuits simply does not provide sufficient accuracy. Ratios of device parameters can be achieved to 0.1% accuracy, but absolute accuracy is only 20%. Therefore, additional circuitry is required to tune time constants in analog circuits [Sakurai et al 1991, Kozma et al 1991, Loh et al. 1992].

Temperature

The effects of temperature can be compensated for by careful circuit design, and if necessary by controlling the temperature of the die itself, a technique afforded by the low power and low mass of analog VLSI chips.

Programmability

The programmability of the stored program computer would at first glance appear to offer many advantages over a permanently routed analog signal processor. However, commercial music synthesizers do not use programmability to allow customers to change signal processing algorithms dynamically. Most digital synthesizers are based on optimized implementations of a small number of special purpose algorithms, such as sample rate conversion [Rossum 1986] [Smith and Gossett 1984], frequency modulation [Chowning 1973] or waveguides [Smith 1992].

Programmability is exploited in commercial synthesizers primarily to allow synthesizer manufacturers to benefit from economies of scale by using processor chips and signal processing arithmetic elements that are used in high volume for other applications. Analog circuits offer another way to obtain the cost advantages associated with higher production volumes: products can be built with small (10-20) arrays of identical analog signal processing chips. Comparable cost savings have not been demonstrated with digital multiprocessor architectures because of the high cost of connecting large numbers of memory chips and high pin count processors [Barrière et al. 1989]. This has frustrated the development of commercial products based on powerful but computationally expensive algorithms such as additive synthesis [Freed 1993].

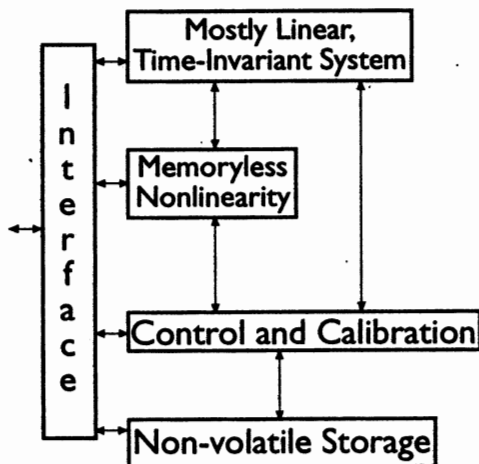
The analog signal processing architecture for musical applications introduced in the next section is not programmable in the familiar stored program sense. It is based on general and flexible computing paradigms that allow for synthesis of a wide range of interesting and intimately controllable sounds.

ANALOG SYSTEM ARCHITECTURE PROPOSAL

The idea of placing a nonlinear circuit element in a feedback loop around a linear system has been exploited musically for a long time; in electronic music it dates to Lee de Forest in 1915 [Darter and Armbruster 1984]. In traditional sound synthesizers such nonlinear oscillators are optimized to generate simple signals such as pulse trains, square waves, sawtooth waves, sine waves and noise [Tilton 1986]. These signals are then summed (additive synthesis) and/or filtered (subtractive synthesis). This architecture of summed and filtered tone generators, has changed little since it was borrowed from electronic organ designs of the 1930's [Kock 1978].

It is now possible to take advantage of a deeper understanding of the nonlinear behavior of musical instruments [McIntyre et al. 1983, Gilbert et al. 1989, Legge and Fletcher 1989, Brod 1990, Fletcher 1990, Hall 1992, Idogawa et al. 1993] and the human voice [Lucero 1992]. Especially exciting is the possibility that a small number of nonlinear oscillator structures may suffice to simulate a broad range of natural and artificial instrument models [Gibiat 1992, Mayer-Kress et al. 1993, Rodet 1992, 1993a, 1993b, 1994].

Traditional analog music synthesizer designs lack two critical elements required to implement these new nonlinear dynamical systems models: a general memoryless nonlinear function unit and an accurately controllable, high-order linear systems simulator. These two elements are therefore the point of departure for the proposed systems architecture:



Memoryless Nonlinearity

Many compact analog circuits for nonlinear function synthesis are known, e.g. trigonometric functions [Gilbert 1982] and square roots [van der Gevel and Kuenen 1994]. Techniques using OTA's are relevant to the proposed architecture [Sánchez-Sinencio et al. 1989].

How can these analog nonlinear circuits be exploited to implement a flexible, configurable nonlinear function synthesizer? One viable method is the artificial neural network (ANN), a multidimensional function approximator which has already been used in musical applications [Lee et al. 1992, Lee & Wessel 1992, Lee & Wessel 1993, Thorson et al. 1993]. Analog ANN realizations are being actively researched [Zurada 1992] with many circuits reported, e.g. [Lansner & Lehmann 1989, Soelberg et al. 1994].

Linear Element

A popular way to implement the mostly linear, time-invariant system is to combine a low order filter with a delay line [Smith 1993]. Although the digital implementation of such a structure is efficient when signal flows are considered, commutation of the desired properties of the linear system into a filter creates challenging control problems when system properties must be changed. Deviations from linear time-invariance are usually slight, e.g. vibrato of strings, but they are sometimes large and musically important, e.g. fingering of woodwinds. These considerations and the difficulty implementing lossless delay lines with analog circuits necessitate the adoption of a different approach for analog linear system simulation.

One strong candidate is the state-variable approach, a well-understood technique used with analog computers [Hausner 1971]. State-variable model parameters may be derived from structural descriptions of acoustical networks [Rodet 1993]. [Rodríguez-Vázquez & Delgado-Restituto 1993] report a circuit using OTA-C integrators for state-variable operations.

Non-volatile Parameter Storage

There are two popular approaches for non-volatile parameter storage. Digital non-volatile memory may be used and the data converted to analog form when required. Alternatively, an analog non-volatile store may be used [Lazzaro et al. 1994], thereby avoiding the conversion step required with a digital store.

Calibration and Control

Calibration methods involve referencing to an accurate high speed, quartz crystal clock and measuring time intervals by counting this clock. This implies the use of digital circuit techniques. This is possible with many modern analog processes. The power consumption of this part of the system can be readily contained and the clock can be shut off after calibration.

Interface

The basic functions of the interface circuitry are buffering, level shifting and device protection. In addition, the use of analog technology allows for custom signal conditioning for direct connections to external gesture sensors such as potentiometers, switches, proximity detectors, microphones, and pickups.

FUTURE DIRECTIONS

The fundamental obstacle preventing the realization of an analog VLSI chip using the ideas presented here is a lack of experience with real-time simulations of nonlinear dynamical models for musical sound synthesis. At this point it is difficult to decide what percentage of chip area to devote to each element in the architecture. To explore this issue further without investing in a VLSI realization, a small analog signal processor consisting of an array of monolithic OTA's has been built. Each OTA can be configured as an integrator or a nonlinearity. Preliminary results with this processor are encouraging.

CONCLUSION

This paper does not announce a commercially viable working example of an analog VLSI music synthesizer. It is possible that the ideas proposed here might never be realized commercially – the unfortunate fate of many VLSI architectures proposed for music synthesizers, e.g. [Kahrs 1981, Wawrzynek & von Eicken 1989]. The following factors suggest that successful realization of an analog VLSI synthesizer may soon be announced:

Resurgence of interest in vintage analog synthesizers and the commercial success of the recently designed Oberheim OB-Mx digitally controlled analog synthesizer [Oberheim 1994].

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Lower cost of ASIC (Application Specific IC) design tools and fabrication.

Large demand for sound synthesis resources fueled by the multimedia and game markets.

Low power requirements of portable computing devices such as PDA's.

Active solicitation by semiconductor companies for new applications designs for their latest analog processes.

Successful implementation of VLSI chips incorporating most of the basic analog signal processing elements required by the proposed architecture [Rodríguez-Vázquez & Delgado-Restituto 1993, Loh et al. 1992].

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