RECENT CCRMA RESEARCH IN DIGITAL AUDIO SYNTHESIS, PROCESSING, AND EFFECTS

*Julius O. Smith**

Center for Computer Research in Music and Acoustics (CCRMA)
Music Dept., Stanford University, Stanford, CA 94305, USA
jos@ccrma.stanford.edu

ABSTRACT

This extended abstract summarizes DAFx-related developments at CCRMA over the past year or so.

1. INTRODUCTION

DAFx-related research has been very active at CCRMA over the past year and beyond. This extended abstract (for one of the keynote talks) provides pointers to research threads in which the author participated as adviser, collaborator, or both.[1]

2. SPRING REVERB SIMULATION

In previous work [5], a spring reverberator emulation was developed based on dispersive allpass filters in a waveguide structure. This work has now been extended to include additional modes of wave propagation, and a full paper is in review [6].

3. REAL-TIME DISCRETE-TIME CIRCUIT MODELS

This past June, David Yeh completed his electrical-engineering Ph.D. dissertation entitled "Digital Implementation of Musical Distortion Circuits by Analysis and Simulation" [7]. The most comprehensive journal paper on this work to date has been submitted [8], and much of the work has been presented at prior DAFx meetings [9, 10, 11, 12]. The basic topic is real-time simulation of analog circuits (both linear and nonlinear) in the digital domain.

* Bio: For the past three decades Julius O. Smith III has been with the Center for Computer Research in Music and Acoustics (CCRMA), and is formally a professor of music and associate professor (by courtesy) of electrical engineering at Stanford University. He received the BS/EE degree from Rice University (Control, Circuits, and Communication) in 1975, and the PhD. degree in EE from Stanford University (Information Systems Lab) in 1983. He worked in the Signal Processing Department at ESL, Inc., on systems for digital communications, in the Adaptive Systems Department at Systems Control Technology, Inc., on problems in adaptive filtering and spectral estimation, and at NeXT Computer, Inc., where he was responsible for sound, music, and signal processing software for the NeXT computer workstation. Smith's teaching and research pertain to music and audio applications of signal processing. He is a Fellow of the Acoustical Society of America and the Audio Engineering Society, and a member of the IEEE since 1976. He is the author of the four online books [1, 2, 3, 4], and numerous research publications devoted to music and audio applications of signal processing. For further details, see http://ccrma.stanford.edu/~jos/.

[1] CCRMA DAFx-related threads not summarized here include those supervised by Consulting Professor Jonathan Abel and his collaborators/advisees. Most of this work has been submitted to the upcoming New York meeting of the Audio Engineering Society in San Francisco.

4. VIRTUAL ACOUSTIC GUITAR MODELS

At the time of this writing (July 2009), Nelson Lee's computer-science Ph.D. dissertation exists in draft form. Recent papers based on this work include (in this proceedings) "Low-Order Allpass Interpolated Delay Loops" [13], "Pitch-Glide Analysis and Synthesis from Recorded Tones" [14] (also in this proceedings), "Excitation Extraction for Guitar Tones" [15], and "Measuring and Understanding the Gypsy Guitar" [16]. In addition to working on finishing his thesis, Nelson is working on a book chapter for the upcoming Springer handbook on musical acoustics of stringed musical instruments, edited by Tom Rossing.

5. HAPTIC FEEDBACK CONTROL FOR VIRTUAL INSTRUMENTS

Edgar Berdahl's electrical-engineering Ph.D. dissertation is also nearing completion. Berdahl has been studying how to use feedback control to design tangible electronic musical instruments.

First, he has introduced several frameworks for changing the acoustical properties of traditional, acoustic musical instruments. The positive real framework provides insight into designing linear feedback controllers [17], while a nonlinear framework enables the design of feedback controllers that can induce especially unusual dynamic behaviors [18].

Second, Berdahl has studied how to use haptic devices to mechanically couple a musician to a virtual musical instrument. Besides introducing a general method for designing haptic musical instruments using digital waveguide technology [19], Berdahl has shown that feedback control can further be employed to advantageously affect a musician's gestures. The Haptic Drumstick demonstrates that feedback control can enable a musician to make gestures that would otherwise be difficult or impossible [20], and a haptic Theremin-like instrument demonstrates that feedback control can aid a musician in playing more accurately [21].

Coincidentally, David, Nelson, and Ed were all EE undergraduate students together at UC Berkeley. It is remarkable to see them all graduating from Stanford at about the same time. The four of us formed a rock band late in the evening on commencement day and had a fitting musical celebration at Nelson's parents' house.

6. NEW DIGITAL OSCILLATOR ALGORITHMS

Juhan Nam is a Ph.D. candidate in the Computer Based Music Theory and Acoustics program at CCRMA. Previously he worked at Young Chang with Hal Chamberlin on Kurzweil synthesizer products.

During the past year, Prof. Vesa Välimäki from HUT spent part of his sabbatical leave at CCRMA. A portion of this visit was a highly productive series of collaborations with Juhan Nam and others at CCRMA.

In one collaboration, Juhan et al. developed alias-free digital oscillators based on a feedback delay loop [22] (in this proceedings).

In another collaboration, Vesa's 2005 paper on "Differentiated Parabolic Waveform" (DPW) synthesis [23] was extended to higher-order polynomial waveforms. The basic idea of DPW is to generate sawtooth waveforms by first synthesizing (digitally) the *time integral* of a sawtooth waveform (*i.e.*, a parabolic wave), followed by a first-order finite-difference ($1 - z^{-1}$) which creates a slightly filtered sawtooth in the digital domain. The net result is less aliasing distortion in the audible band.[2]

Vesa, with one of his graduate students, had already worked out the extension to the third-order case. At CCRMA, we extended the idea to all orders by maximizing "flatness" at the "switch-back" point in the sawtooth waveform [24]. As in the derivations of Butterworth filters [25] and Lagrange interpolators [26], "maximum flatness" of a function $f(x)$ at some point x_0 is defined as maximizing the number of initial zero terms in its series expansion after the constant term. Thus, we desire the first N derivatives of $f(x)$ to be zero at $x = x_0$, where N is the number of degrees of freedom we have. Some further details on this technique are presented in the next section below.

An especially nice contribution from Juhan (in the author's opinion) was the extension of the DPW technique from sawtooth to triangular waves, square waves, and their integrals/derivatives (also described in [24]).

A related journal submission [27] describes other use of polynomial interpolation (especially B-splines) to reduce aliasing. Auditory masking curves were employed by determining the highest fundamental frequency for which all aliased components remain masked under the harmonic components.

6.1. Sawtooth Waveforms via Differentiated Polynomials

Let

$$f(x) = x^n + a_{n-1}x^{n-1} + ... + a_1 x + a_0 \qquad (1)$$

denote the general nth-order monic polynomial. We may differentiate $f(x)$ successively $n-1$ times to obtain the first-order polynomial

$$f^{(n)}(x) = n!\, x + (n-1)!\, a_{n-1}.$$

This polynomial generates a sawtooth waveform as x periodically traverses the interval -1 to 1. To obtain a zero-mean signal, we need $a_{n-1} = 0$. This leaves $n-1$ remaining degrees of freedom for maximizing flatness at the transition from $x = 1$ to $x = -1$ in $f(x)$. Since a_0 has no effect on this transition, we may set it to zero, leaving $n-2$ degrees of freedom.

To maximize wraparound smoothness, we compute a_n for $n \in [1, n-2]$ such that

$$f^{(k)}(-1) = f^{(k)}(1) \qquad (2)$$

for $k = 0, 1, \ldots, n-1$.

[2]Vesa's original DPW algorithm has already been added to Faust's oscillator algorithm library `osc.lib` as the line:
`saw2(F)=saw1(F)<:*<:-(mem):*(0.25'*SR/F);`

In general, every polynomial $f(x)$ can be split into a sum of its even and odd parts:[3]

$$f(x) = f_e(x) + f_o(x)$$

The even part consists of all even powers of x, while the odd part contains all odd-order terms. The even part $f_e(x)$ satisfies our smoothness constraint $f_e(-1) = f_e(1)$ spontaneously. Since the odd part obeys $f_o(-1) = -f(1)$, our constraint (2) for $k = 0$ demands $f_o(-1) = f_o(1) = 0$. In particular, $f_0(1) = 0$ requires that the *sum of the odd-order polynomial coefficients must be zero*. The even-order coefficients are unconstrained for $k = 0$.

The derivative of an even polynomial is odd, and vice versa. Therefore, setting $k = 1$ in our constraint (2), we obtain a constraint on the even part $f_e(x)$ (specifically, the coefficients of $f_e'(x)$ must sum to zero), but no constraint on the odd part $f_o(x)$ for $k = 1$.

Suppose the order n of $f(x)$ is odd. Then $O(f_o) > O(f_e)$, where $O(f)$ denotes the *order* of the polynomial f. If $f(x)$ satisfies the constraints (2) for $k = 0, 1, \ldots, n-1$, then it continues to satisfy those constraints when its even part is replaced by zero. Therefore, only odd polynomials need to be considered in the first place. Similarly, if f is even, then $O(f_e) > O(f_o)$, and f_o may be replaced by zero.

In summary, our polynomial $f(x)$ may be assumed without loss of generality to be an even polynomial (zero coefficients for all odd-power terms) when its order is even; similarly, $f(x)$ can be assumed to be an odd polynomial when its order is odd. This assumption also eliminates about half of the polynomial terms and saves computation in practice.

Given a general even or odd starting polynomial $f(x)$, every other derivative has the sum-to-zero constraint. The totality of these constraints yields an upper triangular matrix equation for the desired polynomial coefficients, which can easily be "back-solved" to produce the coefficients satisfying the constraints. This procedure quickly yields the following results for orders up to $n = 6$:

$$\begin{aligned}
f_2(x) &= x^2 \\
f_3(x) &= x^3 - x \\
f_4(x) &= x^4 - 2x^2 \\
f_5(x) &= x^5 - \frac{10}{3}x^3 + \frac{7}{3} \\
f_6(x) &= x^6 - 5x^4 + 7x^2
\end{aligned}$$

Finally, to generate sawtooth waveforms with reduced aliasing, these functions $f_n(x)$ on $[-1, 1]$ may be passed through the iterated finite difference $(1 - z^{-1})^{n-1}/(2/P_0)^{n-1}$, where $P_0 = f_s/f_0$ denotes the pitch-period in samples. (Other scaling strategies are discussed in [24].)

7. SPECTRAL DELAY FILTERS

A *spectral delay filter* is an audio effect consisting of a chain of allpass filters followed by an equalizer. In a typical application, its impulse response sounds like a "chirp" signal. The equalizer is used to keep the output amplitude more uniform. Based on some

[3]Recall that a function $f(x)$ is said to be *even* if it satisfies $f(-x) = f(x)$, and *odd* if $f(-x) = -f(x)$, for all x. The even part of $f(x)$ is given by $f_e(x) = [f(x) + f(-x)]/2$, and the odd part by $f_o(x) = [f(x) - f(-x)]/2$.

prior work by Vesa and his students, spectral delay filters were developed at CCRMA during Vesa's sabbatical visit, resulting in a paper to appear in the Journal of the Audio Engineering Society [28].

In another collaboration, spectral delay filters were further developed with Vesa's student Jussi Pekonen [29] (see elsewhere in this proceedings).

8. AUDIO FFT FILTER BANKS

Last summer, while trying to finish my spectral modeling book [4], I hit upon some apparently new methods for the design and implementation of nonuniform audio FFT filter banks [30] (elsewhere in this proceedings). This work was motivated by wanting to extend the Brown-Puckette FFT-based constant-Q transform [31] to an efficient implementation of audio filter banks such as those used for "loudness spectrogram" computations [32].

9. FLASH AUDIO PLUGINS AND FAUST TO ACTIONSCRIPT CONVERSION

Travis Skare is a Stanford EE graduate student who, for a project at CCRMA last spring, wrote a Faust architecture file and associated code which creates ActionScript Web-browser plugins from Faust source code. Several of the standard Faust examples have been successfully compiled, such as `pitch-shifter`, `freeverb`, `karplus`, `osc`, and `multibandfilter`. Travis is seeking permission from his employer to release the code in free, open-source form.

Faust compiles to C++, and this can be translated to ActionScript using Adobe Alchemy and Flex. The latest version of Flash 10 is required for runtime-generated sound support, and presently this includes only stereo, 32-bit samples at 44.1 kHz. Microphone input is not yet supported in Flash, but sound can be input to the plugin from disk files. The minimum latency was found to be 46 ms (the smallest buffer size of 2048 at 44.1 kHz). The delay from plugin GUI controls (sliders) to actual slider events was found to be about half a second, so we might as well use the largest audio buffer size in Flash (8192 samples), which alleviates load on the rest of the system.

10. VIRTUAL ACOUSTIC MODELING OF UNDERGROUND LABYRINTHS

Travis is also working on the configurable microphone array project associated with the Chavin project at CCRMA [33, 34]. The Chavin project is concerned with the measurement, archiving, and analysis of the acoustics of underground "galleries" at the pre-Inca site of Chavín de Huántar in Peru. These galleries predate Inca society by over 2000 years. This acoustic modeling project is being carried out in collaboration with Prof. John Rick (Anthropological Sciences) [35].

11. SPECIAL ISSUE OF THE IEEE ASLP

Finally, a special issue on "Virtual Analog Audio Effects and Musical Instruments" is being organized for the IEEE Transactions on Audio, Speech, and Language Processing by Vesa Välimäki, Frederico Fontana, Udo Zölzer, and myself. We received a large number of excellent submitted papers and hope to see the issue appear in March of 2010.

12. CONCLUSIONS

As evidenced above, it has been an active year for DAFx-related research at CCRMA. For further details, see the cited references themselves, and note that many have supporting websites at CCRMA and/or HUT under the lead author's home page.

13. REFERENCES

[1] Julius O. Smith, *Mathematics of the Discrete Fourier Transform (DFT), with Audio Applications, Second Edition*, `http://ccrma.stanford.edu/~jos/mdft/`, Apr. 2007, online book.

[2] Julius O. Smith, *Introduction to Digital Filters with Audio Applications*, `http://ccrma.stanford.edu/~jos/filters/`, Sept. 2007, online book.

[3] Julius O. Smith, *Physical Audio Signal Processing*, `http://ccrma.stanford.edu/~jos/pasp/`, Dec. 2008, online book.

[4] Julius O. Smith, *Spectral Audio Signal Processing*, `http://ccrma.stanford.edu/~jos/sasp/`, Mar. 2009, online book.

[5] Jonathan Abel, David Berners, Sean Costello, and Julius O. Smith III, "Spring reverb emulation using dispersive allpass filters in a waveguide structure," in *Proceedings of the 121st Convention of the Audio Engineering Society*, San Francisco, California, 2006, preprint 6954.

[6] Jonathan S. Abel, David P. Berners, and Julius O. Smith, "A spring reverb model employing coupled torsional and longitudinal modes," in review.

[7] David T. Yeh, *Digital Implementation of Musical Distortion Circuits by Analysis and Simulation*, Ph.D. thesis, CCRMA, Elec. Eng. Dept., Stanford University (CCRMA), June 2009.

[8] David Yeh, Jonathan Abel, and Julius O. Smith, "Automated physical modeling of nonlinear audio circuits for real-time audio effects," *IEEE Trans. Audio, Speech, and Language Processing*, vol. 18, no. 3, Mar. 2010, in review.

[9] David T. Yeh and Julius O. Smith, "Simulating guitar distortion circuits using wave digital and nonlinear state-space formulations," *Proc. Conf. Digital Audio Effects (DAFx-08), Helsinki University of Technology, Espoo, Finland*, Sept. 1–4 2008, `http://www.dafx.de/`.

[10] David Yeh, Jonathan S. Abel, and Julius O. Smith, "Simulation of the diode limiter in guitar distortion circuits by numerical solution of ordinary differential equations," *Proc. Conf. Digital Audio Effects (DAFx-07), Bordeaux, France*, pp. 197–204, Sept. 10–15 2007, 'Best Paper of DAFx-07' award, `http://www.dafx.de/`.

[11] David Yeh, Jonathan S. Abel, and Julius O. Smith, "Simplified, physically informed models of distortion and overdrive guitar effects pedals," *Proc. Conf. Digital Audio Effects (DAFx-07), Bordeaux, France*, pp. 189–196, Sept. 10–15 2007, `http://www.dafx.de/`.

[12] David Yeh and Julius O. Smith, “Discretization of the ’59 Fender Bassman tone stack,” *Proc. Conf. Digital Audio Effects (DAFx-06), Montreal, Canada*, Sept. 2006, `http://www.dafx.de/`.

[13] Nelson Lee and Julius O. Smith, “Low-order allpass interpolated delay loops,” in *Proc. 12th Int. Conf. Digital Audio Effects (DAFx-09), Como, Italy, September 1–4*, 2009.

[14] Nelson Lee, Julius O. Smith, Jonathan Abel, and David Berners, “Pitch glide analysis and synthesis from recorded tones,” in *Proc. 12th Int. Conf. Digital Audio Effects (DAFx-09), Como, Italy, September 1–4*, 2009.

[15] Nelson Lee, Zhiyao Duan, and Julius O. Smith III, “Excitation extraction for guitar tones,” in *Proc. 2007 Int. Computer Music Conf., Aug. 27–31, Copenhagen, Denmark*, 2007.

[16] Nelson Lee, Antoine Chaigne, Julius O. Smith III, and Kevin Arcas, “Measuring and understanding the gypsy guitar,” in *Proc. Int. Symp. on Musical Acoustics (ISMA-07), Barcelona, Spain*, Sept. 9–12 2007.

[17] Edgar Berdahl, Günter Niemeyer, and Julius O. Smith III, “Feedback control of acoustic musical instruments,” Technical report STAN-M-120, Stanford University, June 2008.

[18] Edgar Berdahl and Julius O. Smith III, “Inducing unusual dynamics in acoustic musical instruments,” in *Proc. of the IEEE Conf. on Control Applications*, Singapore, October 1-3 2007.

[19] Edgar Berdahl, Günter Niemeyer, and Julius O. Smith, “Using haptic devices to interface directly with digital waveguide-based musical instruments,” in *Proceedings of the Ninth International Conference on New Interfaces for Musical Expression*, Pittsburgh, PA, June 4-6 2009, pp. 183–186.

[20] Edgar Berdahl, Bill Verplank, Julius O. Smith, and G ünter Niemeyer, “A physically intuitive haptic drumstick,” in *Proceedings of the International Computer Music Conference*, Denmark, Copenhagen, August 2007, pp. 363–6.

[21] Edgar Berdahl, Günter Niemeyer, and Julius O. Smith, “Using haptics to assist performers in making gestures to a musical instrument,” in *Proceedings of the Ninth International Conference on New Interfaces for Musical Expression*, Pittsburgh, PA, June 4-6 2009, pp. 177–182.

[22] Juhan Nam, Vesa Välimäki, Jonathan S. Abel, and Julius O. Smith, “Alias-free oscillators using a feedback delay loop,” in *Proc. 12th Int. Conf. Digital Audio Effects (DAFx-09), Como, Italy, September 1–4*, 2009.

[23] Vesa Välimäki, “Discrete-time synthesis of the sawtooth waveform with reduced aliasing,” *IEEE Signal Processing Letters*, vol. 12, no. 3, pp. 214–217, 2005.

[24] Vesa Välimäki, Juhan Nam, Julius O. Smith, and Jonathan S. Abel, “Alias-suppressed oscillators based on differentiated polynomial waveforms,” *IEEE Trans. Audio, Speech, and Language Processing*, vol. 18, no. 3, Mar. 2010, accepted for publication.

[25] T. W. Parks and C. S. Burrus, *Digital Filter Design*, John Wiley and Sons, Inc., New York, June 1987, contains FORTRAN software listings.

[26] Andreas Franck, “Efficient algorithms and structures for fractional delay filtering based on lagrange interpolation,” *J. Audio Engineering Society*, vol. 2008, Dec. 2008.

[27] Juhan Nam, Vesa Välimäki, Jonathan S. Abel, and Julius O. Smith, “Efficient antialiasing oscillator algorithms using low-order fractional delay filters,” *IEEE Trans. Audio, Speech, and Language Processing*, vol. 18, no. 3, Mar. 2010, in review.

[28] V. Välimäki, J. S. Abel, and J. O. Smith, “Spectral delay filters,” *J. Audio Engineering Society*, vol. 57, no. 7/8, Jul/Aug 2009.

[29] Jussi Pekonen, Vesa Välimäki, Jonathan S. Abel, and Julius O. Smith, “Spectral delay filters with feedback and time-varying coefficients,” in *Proc. 12th Int. Conf. Digital Audio Effects (DAFx-09), Como, Italy, September 1–4*, 2009.

[30] Julius O. Smith, “Audio FFT filter banks,” in *Proc. 12th Int. Conf. Digital Audio Effects (DAFx-09), Como, Italy, September 1–4*, Sept. 2009.

[31] Judith C. Brown and Miller S. Puckette, “An efficient algorithm for the calculation of a constant Q transform,” *J. Acoust. Soc. of Amer.*, vol. 92, no. 5, pp. 2698–2701, 1992.

[32] Brian R. Glasberg and Brian C. J. Moore, “A model of loudness applicable to time-varying sounds,” *J. Audio Eng. Soc.*, vol. 50, no. 5, pp. 331–342, May 2002.

[33] Jonathan S. Abel, Nick Bryan, Travis Skare, Miriam Kolar, Patty P. Huang, Darius Mostowfi, and Julius O. Smith III, “A configurable microphone array with acoustically transparent omnidirectional elements,” *Audio Eng. Soc. Convention*, Oct. 2009, Paper 143.

[34] J.S. Abel, J.W. Rick, P. Huang, M.A. Kolar, J.O. Smith, and J.M. Chowning, “Chavín de Huántar archaeological acoustics project,” 2009, publications to date listed at `http://ccrma.stanford.edu/groups/chavin/-publications.html`.

[35] John W. Rick, “Evolution of authority and power at Chavín de Huántar, peru,” in *The Foundations of Power in the Prehispanic Andes*, pp. 71–89. Archaeological Papers of the American Anthropological Association 14, 2004.