About This Issue

This issue of *Computer Music Journal* presents five articles based on papers from the Tenth International Conference on Digital Audio Effects (DAFx-07), held in Bordeaux, France, in September 2007. DAFx focuses on new research in digital audio and digital music processing, covering many areas commonly found in *Computer Music Journal.* In addition to audio effects, the conference topics include sound representation, modeling, analysis, and synthesis; spatialization and localization; psychoacoustics; audio coding; time-frequency and spectral processing; software and hardware implementations; music information retrieval; automatic transcription; auditory display; source separation; audio restoration; audio for the Internet and for multimedia; performance and gestural control; and composition. The DAFx-07 conference chair, Sylvain Marchand of the Laboratoire Bordelais de Recherche en Informatique (LaBRI) at the University of Bordeaux, was a great help to us. We thank him and the DAFx-07 papers committee for answering various questions and organizing the highest-scoring papers for our consideration. The original reviewers of the five DAFx-07 papers that we selected kindly agreed to re-assess them as revised for *Computer Music Journal.* We are similarly indebted to other referees, not associated with the DAFx-07 committee, who examined some of the manuscripts for the *Journal.*

The first article, by Victor Lazzarini, Joseph Timoney, and Thomas Lysaght at the National University of Ireland, Maynooth, revives frequency-modulation (FM) synthesis and gives it a different twist by substituting the usual carrier oscillator with a complex sound source, such as an acoustic musical instrument played live. What was strictly a sound-synthesis technique becomes an audio effect; a natural-sounding timbre can be morphed into a synthesizer-like one and vice versa. Furthermore, synthetic sounds can retain salient gestural information from the input signal, producing a livelier musical timbre. The idea is not completely new; for example, Cornelius Pöpel's paper at the 2004 Conference on New Interfaces for Musical Expression replaced the modulating oscillator, rather than the carrier, with a live instrument signal. The authors explain that replacing the carrier oscillator allows timbres that are closer to the input sound. Because the carrier no longer has a frequency control built in, some means of achieving FM, or, more strictly speaking, phase modulation, must be provided. The authors present two such techniques: a variable delay line and heterodyning. By tracking the pitch of the carrier signal, the system can control the carrier-to-modulator frequency ratio, which is a classic parameter of FM synthesis. Another classic parameter, the modulation index, can be scaled according to the amplitude of the carrier, an approach that is useful in applications such as emulation of brass tones. The authors present examples of their technique using flute, clarinet, piano, and voice as inputs. Sound examples for this article, and others, will appear on the *2008 Computer Music Journal Sound and Video Anthology* DVD accompanying the Winter issue (Volume 32, Number 4).

The "best paper" designation of DAFx-07 was awarded to work by David Yeh and his co-authors (Jonathan Abel and Julius Smith at Stanford University and Andrei Vladimirescu at the University of California, Berkeley). Their article examines the feasibility of emulating audio electronics by using numerical integration methods to solve the circuits' nonlinear ordinary differential equations (ODEs) in real time. As a case study, the authors take the diode clipper circuit with an embedded low-pass filter, which forms the basis of many solid-state distortion devices such as the Boss DS-1 distortion pedal. This work points out difficulties in modeling such circuits and sheds light on why musicians find many digital emulations lacking. Such digital emulations often use cascade filtering and static nonlinearity blocks to approximate the sound of the analog device. The authors find that solving the circuit's nonlinear ODE provides a more accurate simulation, particularly when using what are known as implicit or semi-implicit solvers. They conclude that real-time emulation is a viable approach to modeling both real and fictional circuits, one that could open new avenues for creative experimentation in the design of digital audio effects.

The article by Heidi-Maria Lehtonen, Vesa Välimäki, and Timo Laakso at the Helsinki University of Technology presents a new refine-

Front cover. An image-processed version of an audio waveform from Axel Röbel's article. (Image processing: *Computer Music Journal.*)

Back cover. An illustration from the article by David Yeh et al., showing a log spectrogram of a diode-clipper response to a sine sweep.

ment to an old idea: using a multi-notch digital filter to cancel harmonics from a musical signal, or to extract them. The authors' new technique replaces the delay line in an inverse comb filter with a high-order fractional-delay filter to increase the accuracy. To cancel harmonics, one uses the inverse comb filter essentially to subtract a delayed copy of the signal from the original signal. To extract harmonics (canceling the rest of the signal), a second-order all-pole filter is cascaded with the inverse comb filter. The technique is simpler and more efficient than other methods for accomplishing these tasks, such as sinusoidal modeling or wavelet analysis. The authors describe several tests of their technique. First, harmonics were extracted from a synthesized tone, and the neighboring harmonics were able to be attenuated by more than 100 dB. Second, the residual (the non-harmonic, noisy portion) was extracted from a recording of a bowed double bass. Third, a beating harmonic was successfully extracted from a recording of the kantele, a Finnish folk instrument with plucked strings. (Preservation of the beating when extracting a harmonic is considered problematic for sinusoidal modeling techniques.)

Like the Lehtonen et al. article, the final pair of articles in this issue relate to the separation of an audio signal into constituent parts. Such separation can help effect, among many other things, convincing sonic transformations such as time-stretching. These final two articles concern the peaks in a spectrum. To decompose a signal into sinusoids, transients, and noise, it is useful to have criteria for classifying each spectral peak. The first of these articles, by Miroslav Zivanovic (Universidad Pública de Navarra), Axel Röbel (Institut de Recherche et Coordination Acoustique/Musique [IRCAM]), and Xavier Rodet (IRCAM), describes a new approach to adaptive threshold selection for classification of peaks. The authors present a compact sinusoidal model that handles nonstationary sinusoids—that is, ones whose amplitude and frequency can both vary. In the authors' model, the parameters for amplitude and frequency modulation are defined with respect to the analysis window, which is significant because short-time Fourier transform (STFT) spectra are tied to the properties of the analysis window. The authors' threshold-selection algorithm permits an intuitive control of the decision thresholds. The decision thresholds, calculated from the relationships between the noise power in the signal and the distributions of sinusoidal peaks, ensure that all peaks described as sinusoidal are in fact correctly classified. The authors also show that the threshold-selection algorithm can be used for different types of analysis windows with only a slight parameter readjustment.

Axel Röbel's own article concerns parameter estimation for sinusoids having linear modulation of amplitude and frequency. Standard techniques such as the quadratically interpolated FFT estimator (QIFFT) introduce significant estimation bias (error) when applied to nonstationary signals such as musical tones performed with vibrato. With such tones as input, the estimation error can be perceived as inappropriate "voiced" (pitched) sound within the residual, which ideally should consist only of noise. The author shows that only frequency (not amplitude) modulation creates additional estimation bias. Then he describes an enhanced algorithm for frequency-domain demodulation of spectral peaks, which can be used to obtain an approximate maximum-likelihood estimate of the frequency slope. The frequency slope estimation and demodulation are followed by the standard QIFFT estimator, yielding an estimate of the amplitude, phase, and frequency parameters that has significantly reduced bias.

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Contents

Errata

The opening paragraph, below, was omitted from "The Laptop Orchestra as Classroom" in the previous issue (*CMJ* 32:1, p. 26):
This article chronicles our pedagogical adventures in the Princeton Laptop Orchestra (PLOrk). We introduce the PLOrk classroom as well as new approaches and tools for teaching. In doing so, we explore an integrated, naturally interdisciplinary educational environment for composition, performance, and computer science. In such an environment, the learning and internalization of technical knowledge happen symbiotically with the acquisition of aesthetic and artistic awareness. There is only one explicit goal: to learn to make compelling computer-mediated music together in an academic setting. All other learning happens "along the way." The presence of experienced guest composers, who compose new works and teach them to the ensemble, allows students to learn about and experiment with varying aesthetic and technical approaches. We believe this is an exciting new environment where the learning of interdisciplinary knowledge is not only natural, but also inevitable (and fun).

The review of the book *New Digital Musical Instruments: Control and Interaction Beyond the Keyboard* (*CMJ* 31:4, pp. 75–77) mistakenly listed the reviewer's location as Belfast, Northern Ireland, UK. The reviewer, Victor Lazzarini, is affiliated with the National University of Ireland in Maynooth, Ireland.