GUEST EDITORS' NOTE
Special Issue on Audio Filter Design

Filtering is one of the most fundamental functions in signal processing, and both analog and digital filters are going strong as important elementary building blocks in more complex systems as well. The same holds true for the field of audio, where in which they are used in a wide variety of contexts.

In a strict sense, digital filters include low-pass, high-pass, bandpass, and bandreject filters, which aim at "filtering out" a part of the frequency spectrum entirely. Audio applications include rumble filters, antialiasing filters in AD and D/A converters, variable low-pass filters in subtractive synthesis, and crossover filters, to name a few. Parametric or graphic equalizers are derived from these prototypes, which, instead of simply eliminating certain spectral bands, aim at shaping the spectrum by attenuating or amplifying the various frequency regions differently. Other filters operate only or primarily on the phase of the input signal, such as all-pass filters used in artificial reverberation.

Technically, a linear, time-invariant filter is any implementation of an arbitrary transfer function in the frequency-domain, or, equivalently, an impulse response in the time-domain. In audio, such filters are used to model or equalize the response of electroacoustic devices, such as loudspeakers, headphones, microphones, radiation responses of musical instruments for sound synthesis, or to simulate the transfer function of analog gear for virtual analog modeling. They are also part of more complex nonlinear effects and reverberation algorithms, and used in various applications of spatial audio processing.

Classic filter design that covers finding the optimal set of parameters for low-pass, high-pass, bandpass, and band-reject filters has been under development for decades, and standard solutions are available both for finite impulse response (FIR) and infinite impulse response (IIR) filters. Modeling an arbitrary transfer function by an FIR filter is straightforward; however, a more complicated optimization may be needed when IIR filters are employed for the same purpose.

The situation gets even more challenging for audio, wherein the aim of reaching for the best sound quality cannot be simply described by the usual least squares or minimax error criteria, and properties of auditory perception need to be taken into account. In addition, the filter may need to model time-varying phenomena, and nonlinearities are often important; in both of these cases, it can be beneficial for the filter topology to match closely that of the device it is modeling. The current special issue presents a wide variety of aspects of audio applications, each showing its own challenges in filter design.

One important property of human hearing is its nearly logarithmic frequency resolution. Accordingly, various specialized filter design methodologies have been developed that provide a more efficient modeling or equalization in the logarithmic frequency scale compared to with traditional FIR or IIR filter design techniques. The first paper of the special issue, entitled "Warped, Kautz and Fixed-Pole Parallel Filters: A Review" by Balázs Bank, presents an overview of such methods and provides a comparison using loudspeaker-room response modeling and equalization examples.

The second paper, "Linear-Phase Octave Graphic Equalizer" by Valeria Bruschi, Vesa Välimäki, Juho Liski, and Stefania Cecchi, demonstrates how a sequence of half-band interpolated FIR filters can be efficiently applied to obtain an equalizer with a logarithmic set of center frequencies, which is an important property of graphic equalizers. A special feature of the proposed equalizer is its linear-phase response as opposed to the minimum-phase response of traditional IIR equalizers.

In the third paper, "Improving the Chamberlin Digital State Variable Filter" by Victor Lazzarini and Joseph Timoney, the authors revisit a popular digital filter realized by Chamberlin, modeling state-space-based analog structures used in several electronic music circuits. The new realization leads to a more accurate frequency response, and its performance is compared to with competing design approaches as part of a broad discussion on Digital State Variable Filters.

The fourth paper, "Real-Time Transient Reduction in Higher Order Time-Varying Musical Filters" by Nikhil Deshpande and Russell Wedelich, is logically connected to the previous one as it is concerned with spurious transients in state state-space realizations whose parameters are varied at runtime, a typical issue affecting musical filters. This paper proposes higher higher-order digital transformations of state state-space analog structures as a solution to reduce transient distortions in time varying situations, with an application to equalization filters.

An important class of audio effects is virtual analog modeling, where in which one of the approaches for discretizing analog circuits is based on wave digital filters (WDFs). The fifth paper, "Parallel Wave Digital Filter Implementations of Audio Circuits with Multiple Nonlinearities" by Riccardo Giampiccolo, Antonino Natoli, Alberto Bernardini, and Augusto Sarti, proposes a method for speeding up the computation of such simulations by the parallel version of the Hierarchical Scattering Iterative Method.

Peak reduction is commonly used in audio mastering to increase the loudness of the music material. This is usually achieved by using a compressor and/or limiter. In contrast, the sixth paper, "Audio Peak Reduction Using Ultra-Short Chirps" by Vesa Välimäki, Leonardo Fierro, Sebastian J. Schlecht, and Juha Backman, proposes two methods that change the phase of the music signal to decrease its crest factor. Among the two methods, the one based on all-pass filters is proven to be superior by the numerical and perceptual investigations.

The seventh paper, "A Workflow and Digital Filters for Correcting Speed and Equalization Errors on Digitized Audio Open-Reel Magnetic Tapes" by Niccolò Pretto, Nadir Dalla Pozza, Alberto Padoan, Anthony Chmiel, Kurt James Werner, Alessandra Micalizzi, Emery Schubert, Antonio Rodà, Simone Milani, and Sergio Canazza, reports on current developments of equalizers for compensating digitization artifacts that occur during audio tape content archiving and restoration. After introducing a workflow wherein these equalization filters are employed, results from listening tests are presented measuring the quality of the resulting restoration process.

Cylindrical radial filters are used in spatial signal processing applications such as sound field synthesis and active noise control. The eighth paper, "Cylindrical Radial Filter Design with Application to Local Wave Field Synthesis" by Nara Hahn, Frank Schultz, and Sascha Spors, proposes an analytical method for designing such filters starting from the continuous-time impulse responses. The aliasing spoiling the discretization of such impulse responses is reduced by adding a residual function to the sampled responses approximated by spherical radial functions. The addition of residual functions to reduce aliasing was developed previously for bandlimited synthesis of square and sawtooth waves in virtual analog modeling.

The last paper, "Resynthesis Of of Spatial Room Impulse Response Tails With Anisotropic Multi-Slope Decays" by Christoph Felix Hold, Georg Götz, Thomas McKenzie, Sebastian Schlecht, and Ville Pulkki, is also related to spatial audio processing. Spatial room impulse responses are used together with ambisonics for modeling the reverberation of acoustic spaces. The present paper proposes a method to restore the late part of measured reverberation responses that is otherwise buried in the noise floor. The directional signals are extracted by a spatial filter bank, and the envelope of the direction-dependent reverberation is modeled by multiple exponential decays. The results are evaluated both by objective measures and listening tests.

We hope that readers of the AES journal will find the present collection of papers useful and inspiring. While Although the methods and algorithms are proposed to solve particular problems for each application area, they can find use in other contexts as well, and are likely to prompt further research.

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