Ph.D. in Information Technology
Thesis Defenses

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Riccardo Giampiccolo– XXXV Cycle

Multiphysics Modeling of Audio Systems in the Wave Digital Domain

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Abstract:

Audio systems, such as audio sensors and actuators, are pervasive devices that have taken over many different markets, ranging from the classical consumer electronics market up to that of biomedical devices. Many are the components and the transduction principles on which audio systems rely, as well as many are the physical domains - such as electrical, acoustic, mechanical, magnetic, etc. - involved in the digital/analog processing chains characterizing such complex systems. The physical domains do not only interact in a nonlinear fashion but are in turn affected by many different types of nonlinearities which are highly responsible for the peculiar characteristics of the audio gear. In order to outline, digitally replicate, and process the signals of such systems, multiphysics models able to account for such nonlinear behavior must be derived. In the literature, the physical modeling of audio systems has been addressed following both distributed and lumped approaches. In this thesis, we decide to employ Lumped-Element Models (LEMs), although being generally characterized by less descriptive power than distributed models, since they show many different properties which can be highly valuable for our purpose. Our ultimate goal, in fact, is to derive real-time digital signal processing algorithms for enhancing the acoustic performance of audio systems leveraging the underlying physics. In particular, we are interested in the modeling of multiphysics systems that have as input or output signals an electrical quantity. It has been demonstrated that, in such cases, LEMs are a well-suited modeling methodology given that, for example, they can be easily interfaced with such kinds of systems and they can be digitally implemented in an efficient fashion. Moreover, it is possible to exploit the analogies between electrical and other physical domains in order to derive equivalent circuit representations of audio systems that allow us to analyze all the physical domains in a unified fashion. Among the methods for the implementation of LEMs, we take into account Wave Digital Filters (WDFs) since, over the past few years, they have demonstrated good properties for the digital realization of nonlinear audio circuits or, more in general, of physical systems described by means of electrical equivalents. Introduced by A.~Fettweis in the late 70s, WDFs are a particular class of digital filters based on physical modeling principles. Circuit elements and topological interconnections are realized as input-output blocks characterized by scattering relations. Port voltages and currents are substituted by a linear combination of incident and reflected waves; in addition, a free parameter is introduced into the port description leading to several numerical advantages. Amongst others, the possibility of solving circuits with up to one nonlinear element (characterized by explicit mapping) in a fully explicit fashion can come in handy for the processing and emulation of multiphysics circuits. In general, WDFs preserve the stability properties of the reference circuit; moreover, they are modular, accurate, and efficient, proving to be suitable to be employed in digital audio signal processing algorithms for consumer electronics applications. In this thesis, we thus propose new Wave Digital (WD) modeling techniques able to efficiently

describe the different physical domains of audio systems in a modular fashion. For example, we provide new models of audio transformers, piezoelectric loudspeakers, and guitar pickups. As far as audio transformers are concerned, we take into account the modeling of magnetic saturation by means of Canonical Piecewise-Linear functions, and the modeling of rate-dependent hysteresis by encompassing a particular Recurrent Neural Network trained on measurement data into a WD block. We then generalize the Scattering Iterative Method (SIM), i.e., a fixed-point iterative method able to solve WD structures characterized by an arbitrary number of nonlinear elements and one single topological junctions, deriving a hierarchical version - which we called Hierarchical SIM (HSIM) - able to accommodate both multiple nonlinearities and junctions; in fact, this is typically the case in multiphysics systems. We then exploit the high number of embarrassingly parallelizable operations of HSIM for deriving parallel implementations able to speed up the circuit emulation and cope with the latest developments of Digital Signal Processors (DSPs) as far as multi-core Central Processing Units (CPUs) are concerned. We then show how it is possible to apply Newton-Raphson routines to nonlinear WD structures organized in the form of a connection tree for enhancing the convergence speed. Once both models and methods for multiphysics emulation of audio systems are introduced, we show how it is possible to use them for deriving virtualization algorithms. In fact, we introduce a new class of digital signal processing algorithms able to impose the behavior of target sensors/actuators using novel processing chains based on circuit inversion. Such algorithms can be exploited for the compensation/linearization of audio transducers, but especially for modifying their behavior making them sound like other transducers. We analyze from a theoretical standpoint such algorithms, providing ready-to-use schemes and block diagrams for deriving the circuital inverse of physical models, as well as for applying the virtualization chains to all the possible input/output configurations. Finally, in applications in which virtualization algorithms might not be able to impose a target behavior due to very strict physical constraints of the audio transducers, we propose to exploit psychoacoustic effects. We address the case of low-frequency enhancement in small-size transducers, and we propose to use Virtual Bass Enhancement (VBE) systems, which exploit psychoacoustics to trick the human perception of sound. We first propose a circuit equivalent realization of a generic time-domain VBE system, enabling the design of new VBE algorithms in analog, digital, and hybrid analog/digital audio applications. Then, we introduce a novel VBE system that relies on deep learning techniques for music demixing. The method solves known problems of common techniques and overcomes the state-of-the-art method as far as bass enhancement is concerned.

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