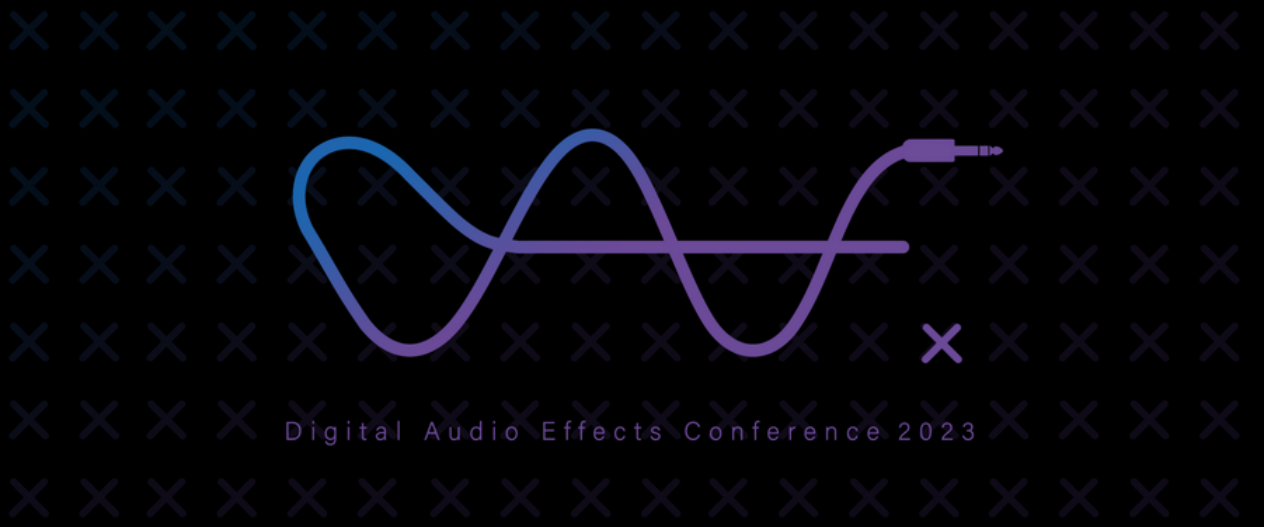

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Welcome to DAFx23!

This year the DAFx conference responds to the tireless creativity our community with a lot of novelty. By substantially shortening oral presentations in favour of a prolonged discussion after each session, and by considering late breaking results whenever a team of researchers proposed an ongoing yet convincing idea, the 2023 edition disrupts the traditional setting. Additionally, the logo, which was established as early as 1998 for the first conference edition in Barcelona, has been renovated. With its five thematic oral sessions, a poster session, a session including demos and late breaking results, two mindopening keynotes and one special session animated by the industry, DAFx23 testifies the continuously increasing interest for digital audio effects as flagship products, squeezing the best from what acoustics, virtual analog, software engineering and product design can offer today. In these proceedings the interested reader in fact will discover unexpectedly rich and advanced applications of sound wave theory, musical circuit modelling, artificial intelligence for music, high-performance algorithms, software coding, real-time programming, human-computer interaction and subjective evaluation, to mention some.

The social part further revolutionises the get together at the conference hall, the breaks and the social events: we will be invited to extend our discussions and to follow off-events later in the evening, yet without setting the alarm clock just in time as yoga early morning sessions will be organised to engage us, and optimally set our minds to receive the science to come next across the day.

And, last but not least, let us not miss the chance to submit an extended version of our research to the special issue on Directions of Digital Audio Effects, to appear on the EURASIP Journal of Audio, Speech and Music processing under the editorial supervision of Stefania Serafin, Federico Fontana and Silvin Willemsen.

Twenty-five years have passed since the first edition in Barcelona, and DAFx shines younger than ever. So, loud and again, welcome to DAFx23!

Stefania, Federico, Silvin
Copenhagen, September 2023

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Keynotes

Sustainable perceptual evaluation for digital audio applications: motivation, methods, and best practice

Hanna Järveläinen, Zurich University of the Arts

Listener tests are a standard, partly even standardized, procedure in the development of new audio technology. This kind of testing is often carried out to measure the degradation of perceived audio quality. Experimental research with participants is also performed in basic and applied psychoacoustic research, user experience studies with digital musical instruments and interfaces, and in designing applications for special groups. Within the DAFx community, methods from the audio testing field are presently most widely used. However, long-term development requires considering the end user both in passive perception and active interaction, often in multisensory, ecological, or creative settings. The presentation will discuss state-of-the art procedures and analysis methods that could contribute to digital audio research in this challenging environment.

Linking Sound, Morphology and Perception: Towards a Language of Sound

Mitsuko Aramaki, CNRS – Aix-Marseille University

The Sciences of Sound and Music have considerably grown notably thanks to the development of digital audio processing. Today, we have numerous synthesis tools, models and methods for sound creation to generate sounds of impressive realism. However, the perceptual control of these synthesis processes remains a current challenge. Based on the ecological approach to perception, a synthesis control paradigm enabling the creation of sounds from evocations has been proposed and led to the development of environmental sound synthesizers which can be directly controlled from perceptual attributes. In this talk, we will present the methodology through a series of perceptual studies to better understand the relationship between sound morphology and human perception for synthesis purposes with an interdisciplinary perspective.

Tutorials

Virtualization of Acoustic Transducers based on Direct and Inverse Circuital Modeling

Alberto Bernardini, Politecnico di Milano

Audio systems can be often accurately described using equivalent circuit models that are capable to represent their behavior in multiple physical domains (e.g., electrical, magnetic, mechanical, acoustic) in a unified fashion. Moreover, such models allow us to efficiently emulate audio systems in the digital domain by employing circuit simulation methods. In this tutorial, we will highlight a further advantage of representing audio systems using circuits. In fact, we will show how, given an audio system represented as a circuit with input and output signals, it is possible to design the corresponding inverse circuital system. As a first example of application of this inverse system design approach, we will describe a method for loudspeaker virtualization through digital audio signal preprocessing. This method can be used to alter the behavior of a physical loudspeaker in such a way to match that of a target loudspeaker. Special cases of this approach are loudspeaker linearization and equalization. The proposed virtualization algorithm is extensively tested both through simulations and applications to real loudspeakers. Moreover, we will show how a similar reciprocal approach can be used for the virtualization of acoustic sensors, like microphones or guitar pickups. Finally, further possible examples of application and related future research works are discussed.

Performance analysis of DSP algorithms

Stefano D'Angelo, Orastron Srl

Performance analysis of DSP algorithms in scientific literature is usually limited to counting the number and type of operations involved and sometimes determining their algorithmic complexity. While these metrics are important, they can only give a rough idea on the computational cost of actual implementations. This tutorial touches on theoretical and practical aspects of trying to achieve high-performance when implementing DSP algorithms on modern platforms, such as computer architectures, instruction sets, operating systems, and numerical analysis.

Recent developments in Topological Signal Processing

Georg Essl, University of Wisconsin-Milwaukee

This workshop will present recent developments in Topological Signal Processing that spring from combinatorial Hodge theory. Graph and simplicial versions of frequency analysis and filter constructions have emerged in this framework. We will cover combinatorial Hodge theory and its relationship to Homology and classical vector calculus. We then develop graph signal processing and basic notions such as the graph Fourier transform. Finally, we will extend these ideas to arrive at simplicial signal processing, which is a topological version of higher dimensional signal processing.

Workshops

Applying Machine Learning to Virtual Analog Modeling

Boris Kuznetsov, TikTok UK

How do we apply advances in the machine learning and deep learning fields to the discipline of classical signal processing and virtual analog modeling? We will talk about how you can use machine learning to both help you design and iterate on virtual analog models faster and how to potentially replace entire subsets of your work with deep learning.

Daisy Dub Hackathon for Digital Audio Effects Enthusiasts and DSP developers (Max/gen, Rust, C++, PureData, Arduino)

Rasmus Kjærbo & Leo Fogadić, Componental

Join us for an exhilarating hackathon workshop centered around the groundbreaking Daisy Dub pre-production prototype: the ultimate swiss army knife tailored for music producers, DSP developers, and DJs. Dive into the expansive world of quadrophonic real-time audio engineering and live performance and experience the processing power of Electro-Smith's Daisy Seed DFM module.

Co-designing Tactile Experiences for Musical Creativity

Doga Buse Cavdir, Multisensory Experience Lab, Aalborg University

This design workshop invites all participants to co-create new mappings for audio-tactile languages. The workshops will introduce and demo new haptic interfaces while developing discussions on the relationships between vibrotactile stimuli and music. During this two-part workshop, we hope to explore the relationships between musical and tactile communications and identify new frameworks which allow Deaf/Hard of Hearing (DHH) individuals to learn and create music using vibrotactile experiences.

Special sessions

Advances, Barriers, and Future Direction for Hearing Aid Effects

Niels H. Pontoppidan, Danish Sound

During the last 20 years advanced applications for hearing instruments only possible with machine learning (ML) emerged. However, for many and for long the computational complexity did not allow for actual implementation. Nevertheless, in 2020 core signal processing based on ML principles came in use for enhancing speech in the presence of noise. It is interesting to look back at the interplay of applications, algorithms, connectivity, and hardware to speculate about the next core signal processing areas in hearing instrument that ML will enhance.

Leveraging Deep Learning for Enhanced Signal Processing in Telecommunication Devices: A Step towards Futuristic Audio

Clément Laroche, Jabra

With the rapid evolution of artificial intelligence and deep learning algorithms, it's essential to discern their transformative impacts on telecommunication devices' audio performance, specifically headsets, speakerphones, and videobars. This presentation will start by delineating the challenges faced by conventional signal processing techniques in the current digital age, such as the inability to effectively filter ambient noises in varying environments or adapt to different speech characteristics. We then explore how can deep learning-based approaches aid in overcoming these challenges by learning complex, non-linear relationships from vast audio data. However, the computational demand and memory footprint of such advanced models can often pose a challenge for their deployment on resource-constrained embedded devices. Limitations imposed by the computational and memory requirements of deep learning models, highlighting the importance of model optimization for their practical use in real-time telecommunication devices. Spotlight will be put on dynamic neural networks, a compelling concept that allows 'early exiting' from computations. This approach facilitates rapid decisions when the network encounters less complex tasks, thus conserving computational resources — a valuable attribute for real-time applications on embedded devices. In addition to the technical aspect, we believe in the invaluable role of human listeners in validating our models. Hence, we will share results from a study conducted on Amazon's Mechanical Turk platform, where a diverse crowd sourced the rating of audio quality. The insights gathered from these human ratings provided a more nuanced understanding of the perceived audio quality, underlining the importance of a human-centric approach in our technical advancements.

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