A Trainable Cochlear Filter for Audio Processing

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We investigated the possibility of implementing online learning rules in analog VLSI. Specifically, we intend to develop an aVLSI circuit design for a trainable adaptive filter for audio processing that feeds the output of one of Andre's cochleas into a simple neural network that is adapted on-chip by Nici's stochastic meta-descent (SMD) rule, reformulated as three coupled continuous-time differential equations.

No analog or nueromorphic chips seem to exist that incorporate online learning, even though adaptation features heavily in neural systems. While other attempts in this direction involve either floating gate technology or binary weights, we believe a continuous-time adaptation process with capacitive parameter storage well-suited for online aVLSI signal processing.

The expected outcome is a first circuit design for the trainable cochlear filter, which will serve as a basis for a full-scale grant proposal. Collaboration is necessary because Nici has experience in online learning, but not in analog VLSI whereas Andre has experience in analog VLSI but not in learning rules. A lot of the details of how to adapt the learning rules to circuit implementation and how to best interface this with a silicon cochlea still need to be worked out before full research can be developed.

In addition to the standard advantages of aVLSI design (simple circuits, low power consumption, etc.), a cochlear filter should be especially suited for applications relating to human perception, and thus speech signals in particular. We therefore want to use cochlear filtering to tweak adaptive noise suppression techniques such that they leave speech signals intact, thus enhancing the comprehensibility of speech in a noisy audio signal.

We set up and ran simulations of several cochlear models on speech signals, followed by lateral inhibition and adaptive reconstruction of the input signal. These experiments showed that good reconstruction of the speech is possible even when some cochlear bands are knocked out (as could happen e.g. to suppress noise), but only if reconstruction coefficients are adapted accordingly. Online adaptation will thus be an essential ingredient in our circuit designs.