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Dynamic Audio Sensor

The Dynamic Audio Sensor is an asynchronous event-based silicon cochlea. The board takes stereo audio inputs; the custom chip asynchronously outputs a stream of address-events representing activity in different frequency ranges. As such it is a silicon model of the cochlea, the auditory inner ear. The system has also been called AER-EAR, where AER stands for Address-Event Representation – the protocol used to transmit spikes.

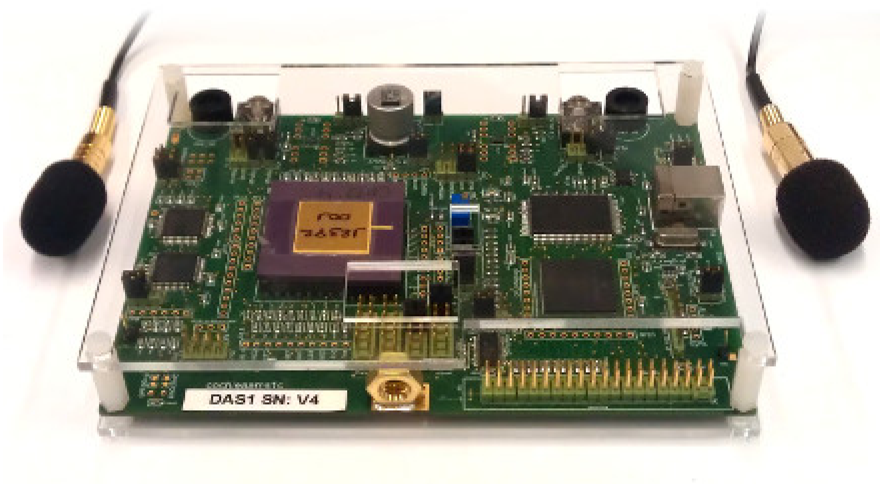


Fig. 1. Dynamic Audio Sensor USB board (DAS1), with on-board microphones, preamplifiers, and digitally controlled biases.

The Dynamic Audio Sensor (DAS1) is a binaural cochlea intended for spatial audition and auditory scene analysis. It is explained most thoroughly in this paper: [Liu et al 2014](). It uses cascaded second-order sections (SOS) to model the physical oscillation of the basilar membrane. These drive half-wave rectifier circuits, which model inner hair cells. These in turn drive multiple pulse frequency modulator circuits, which model ganglion cells with different spike thresholds. The resonance of individual sections can be adjusted with local digital-to-analog converters.

This chip includes a variety of features including a matched binaural pair of cochleas, on-chip digitally controlled biases, on-chip microphone preamplifiers, and open-sourced host software APIs and algorithms. A bus-powered USB board enables easy interfacing to standard PCs for control and processing (Fig. 1), whilst a parallel AER port allows direct connection to other dedicated spiking neuromorphic hardware. The responses of the binaural DAS to speech and a chirp are shown in Fig. 2 and Fig. 3 respectively.

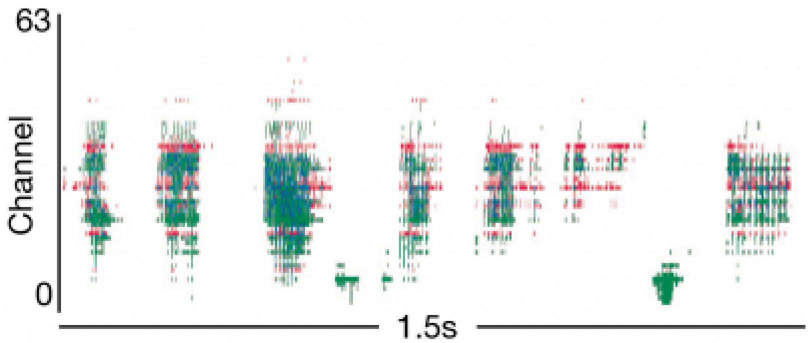


Fig. 2. Response to speech “The quick red fox jumped over the lazy dog”. The different colors correspond to channels of the different ears. Each dot is one event. The mean event rate is 17keps.

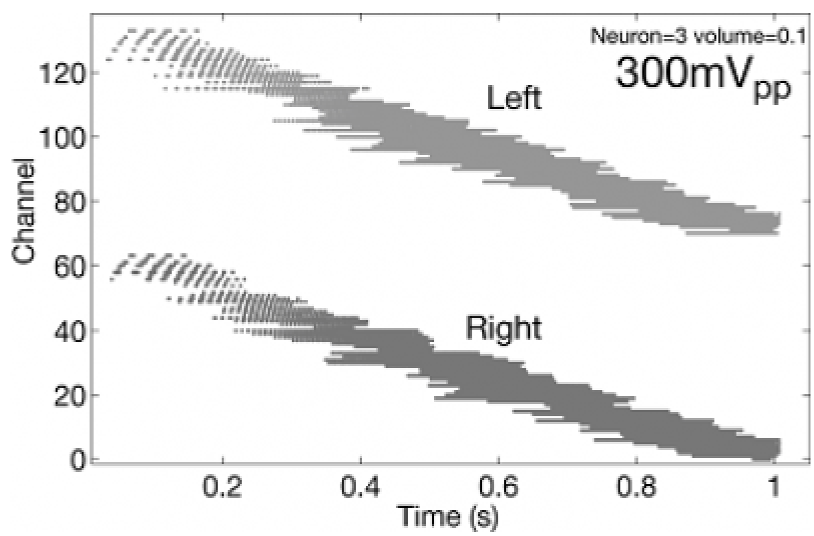


Fig. 3. Event rasters recorded from the 64 channels of both ears. Frequency is logarithmically swept from 30Hz to 10kHz with input amplitude 300mVpp.

Spike-Based Auditory Processing

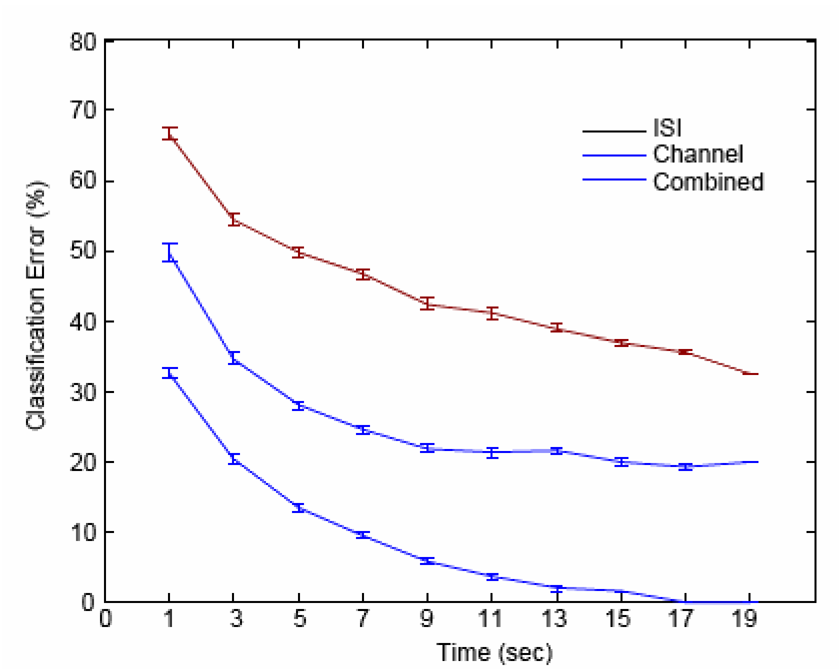
Localization

The timing information from the DAS can be used for inferring the location of an acoustic source, for example, from the interaural time differences between sounds arriving to the two ears [37] or by emulating the echolocalization mechanism of bats (Abdalla and Horiuchi, 2005). The spike outputs can also be used to extract higher-level auditory features suitable for tasks such as harmonicity detection (Yu et al, 2009) and speaker identification (Liu et al, 2010b).

One useful application of the binaural DAS is in the extracted of localization cues. One of the main cues is the interaural time difference (ITD) cue. This cue comes from the difference in the timing arrival of sound waves at the two ears from a sound source can code for the location of a sound source in the azimuth direction. The use of the spike timing arrival difference of the output spikes of the binaural DAS can be used for sound localization (Chan et al, 2007). The data in Figure 9 of this work show how well the reconstructed ITD matches the actual source ITD illustrating that the address events preserve timing information for the extraction of interaural time differences.

An on-time localization algorithm running in jAER can be used to extract the sound location with approximate resolution and lower latency than using a cross-correlation algorithm on the microphone outputs (Holger and Finger, 2011).

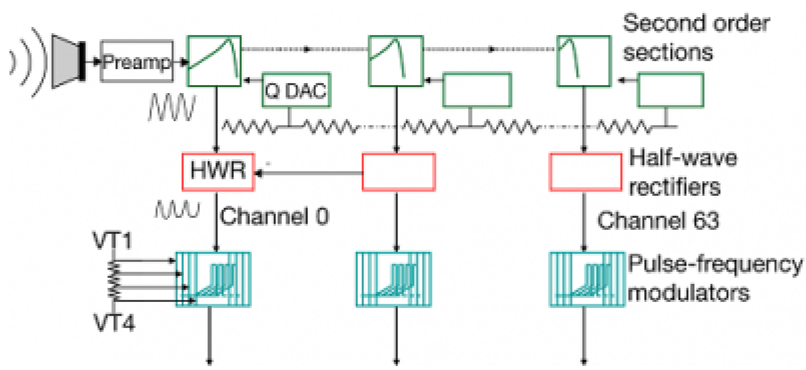
Speaker Verification



An early experiment involving speaker identification on the TIMIT database (Liu et al 2010) with 20 males and 20 females together with Mesgarani and Hermansky shows possibilities of spike coded features from the DAS outputs. Three different feature sets are extracted in identifying speakers: 1) average ISI across channels 2) average ISI across time and 3) combination of 1 and 2. Classification was performed by training 40 linear Support-Vector-Machines (SVM) for each speaker versus the remaining speakers. The results show that the classification error of the speakers reduces over longer durations of speech.

DAS1 Design

The binaural chip has two separate 64-stage cascaded filter banks allowing connection to two electret microphones. The architecture of one of the two cochleas on the chip is shown in Fig 1. Each cochlea consists of a 64-stage cascaded filter bank stage. The cascaded architecture is preferred over a coupled bandpass architecture so we can achieve better matching and sharp high frequency roll-off. The coupled architecture is particularly susceptible to destructive interference at mismatched stages. A voltage-mode implementation is chosen over a current-mode implementation because of better robustness to fabrication variances. The impact of the smaller linear input range is reduced by including global automatic gain control (AGC) on the front of the filters, using off-chip microphone preamplifiers. The voltage-mode filter bank implementation also reduces variability compared with log-domain current-mode implementations which are very susceptible to current copying mismatch. However, noise accumulation and time delay along the cascade favor a small number of sections per octave, making it harder to maintain high Q. But maintaining acceptably high Q is important for spectral selectivity and is why this chip includes Q adjustment and lateral suppression circuits, as will be explained.



Each filter stage consists of a second-order-section (SOS) filter which is biased by a Complementary Lateral Bipolar Transistor (CLBT) ladder to improve matching [11]. A differential readout of each SOS output drives its own halfwave rectifier (HWR) circuit, and the HWR output drives 4 pulse-frequency modulators (PFMs). The PFM circuits implement an integrate-and-fire model with a threshold (VT). The four PFMs have individual global thresholds (VT1 to VT4), allowing volume encoding by selective activation of PFMs. Compared with regularly-sampled audio systems, the PFM outputs are transmitted asynchronously, reducing latency to the analog delay along the filter bank and increasing temporal resolution to microseconds.

The resonance of individual sections can be adjusted by a local digital-to-analog converter (QDAC). This chip includes a variety of features including a matched binaural pair of cochleas, on-chip digitally controlled biases, on-chip microphone preamplifiers, and open-sourced host software APIs and algorithms – the highly-usable USB2 implementation, and the jAER event-based processing software. A bus-powered USB board enables easy interfacing to standard PCs for control and processing (Fig. 9).

System Integration

The DAS1 is integrated with a USB2.0 high-speed interface that plugs into any PC or laptop. The host software presently stands at >200 Java classes. The open source jAER software project lets you render events in a variety of formats, capture them, replay them, and most important, process them using events and their precise timing.

Specifications

Functionality	Asynchronous cochlea + ganglion cell output
Fabrication process	4M 2P 0.35um standard CMOS
Channel count	64×2
Chip size size mm2	3.5 x 6
Chip interface	12-bit word-parallel AER active low Req and Ack 4-phase handshake
Computer interface	USB 2.0, Windows XP driver Java API & Matlab output file format
Power consumption	Chip: 18.4mW to 26mW (DVdd) 33mV (AVdd) USB System: approx. 70mA
Dynamic range to produce PFM output	36dB (25mVpp to 1500mVpp) at microphone preamp output
Frequency range	50Hz to 20kHz (adjustable)
PFM Best characteristic frequency (BCF) matching	+-16% between ears at 150mVpp

PFM Q and Q matching (BCF/width at 0.7 of BCF)	1.5 +- 0.4 (+-27%) at 450mVpp
Event timing jitter, 1kHz input	+- 2mus at 250mVpp
PFM peak bandwidth	10M events/sec
Total PFM typical speech rate	20k events/sec

Acknowledgements

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