Ingredients of An Embedded AI Solution Software as Main course with Hardware as a Garnish and AI as Condiment

Al Hardware – Munich Urban Colab

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Outline

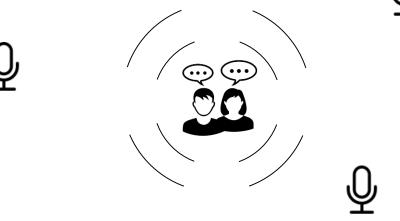






Goal

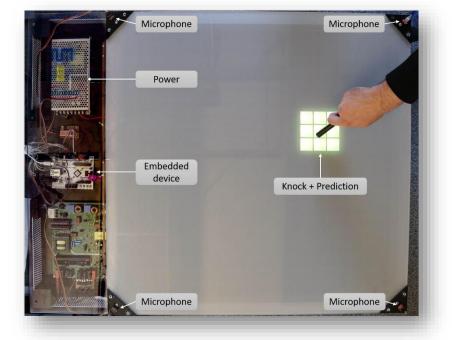
- > Audio source localization is an interesting topic nowadays
 - E.g. supermarket or industrial environment



- > Why not use ML and an embedded device to do that?
 - Embedded devices are extensively used in industrial environment due to low cost
 - Problem: limited hardware resources

Proof – of -Concept

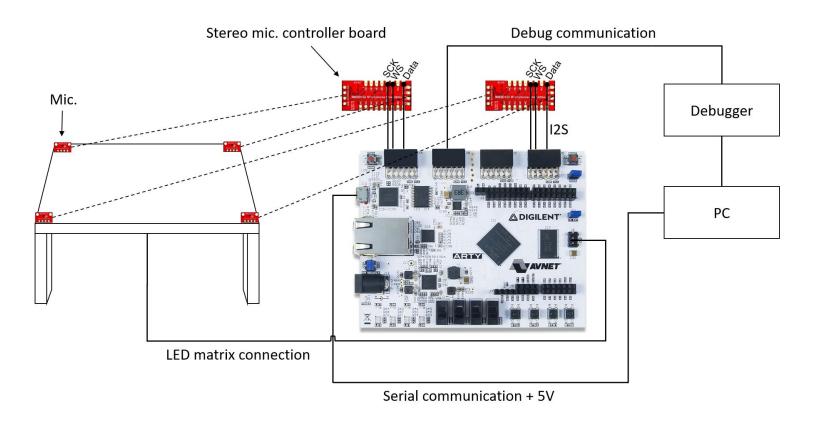
- > Setup:
 - 4 synchronized microphones
 - 1 FPGA running a RISC-V CPU with special ML instructions
 - 64 KBytes RAM
 - 64 KBytes ROM
 - 20 x 20 LED matrix
- > Use the setup for
 - Data collection
 - Proof-of-Concept





Setup

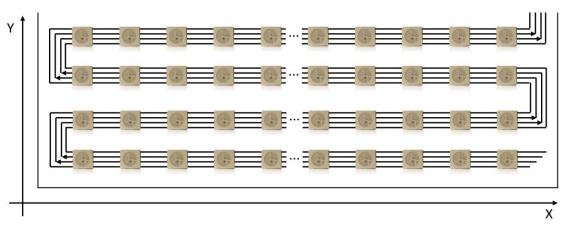






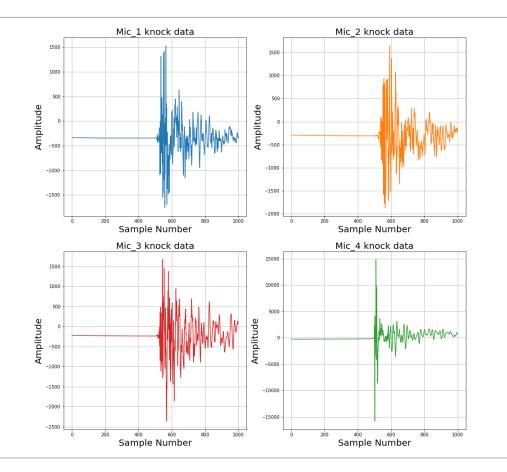
Capturing the dataset

- > Coordinate system with origin at the left bottom of the LED matrix pad
- > Capturing process
 - Random permutation of all LEDs is created
 - A LED lights up
 - A knock is recorded
 - The audio data is sent to the PC for further processing



Signal characteristics





- Plot of sample audio data corresponding to position (20,1)
- High Amplitude for Mic 4
- Less Echo for Mic 4

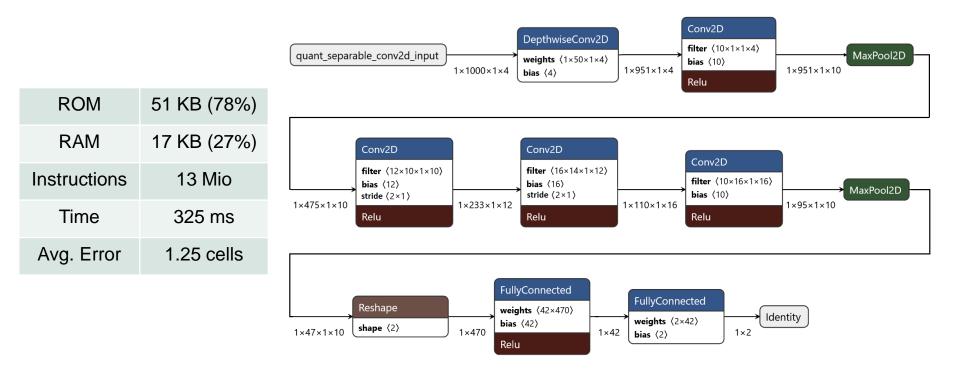
Outline







AI model





Convolutional

- Well understood & easy to compute
- > Supported in ML frameworks for edge devices (e.g. TFLite)
- > Best accuracy

Depthwise convolutional

- Adds to the concept of convolution
- > Reduces the amount of expensive multiplications

Temporal convolutional

- Causal convolutions
- > Brings time dependency into CNNs
- Not supported well in TFLite



How to fit an AI model in 64KB RAM and 64KB ROM (1)

- Reduce model complexity
 - Use stride where possible
 - Use max pooling if strides don't perform well
 - Replace multiplication heavy convolutions with depthwise convolutions
 - Analyze the model in regards to under- and overfitting
- > Use quantization
 - 32 Bit float if no quantization is used
 - 16 Bit int quantization: 50% memory reduction
 - 8 Bit int quantization: 75% memory reduction
- > Do scalable pruning

	ROM usage	RAM usage
Final AI model	51KB (78%)	17KB (27%)



How to fit an AI model in 64KB RAM and 64KB ROM (2)

- > Tensor Flow Light Interpreter is with several 100k binary code size way to big
- > Use Tensor Flow Lite Micro
- > Compile Model
 - Apply compression techniques to weights
 - Use pre-interpreter to identify required functions and re-implement to support compression
 - Generate Skeleton
 - Merge minimum number of required functions with Skeleton and Compile
 - Consider compression in hardware accelerator
- > Use TVM in future

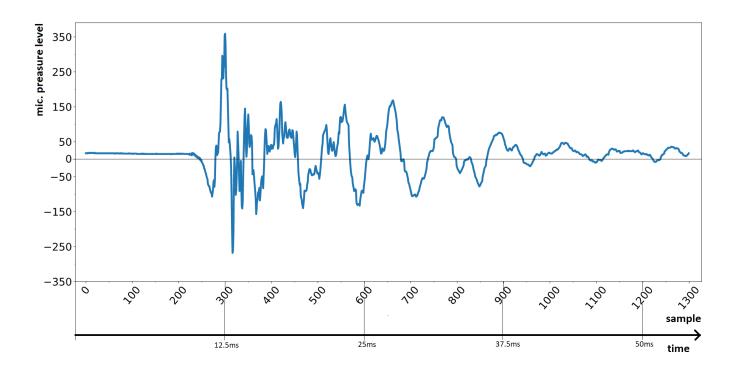
Outline

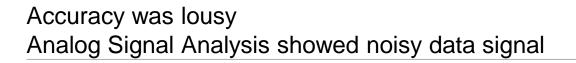






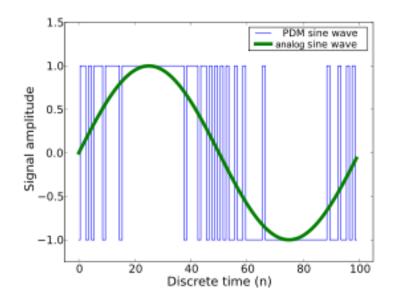
Accuracy was lousy







- I²S Data Line Signal was partially corrupted
 - Replaced by PDM

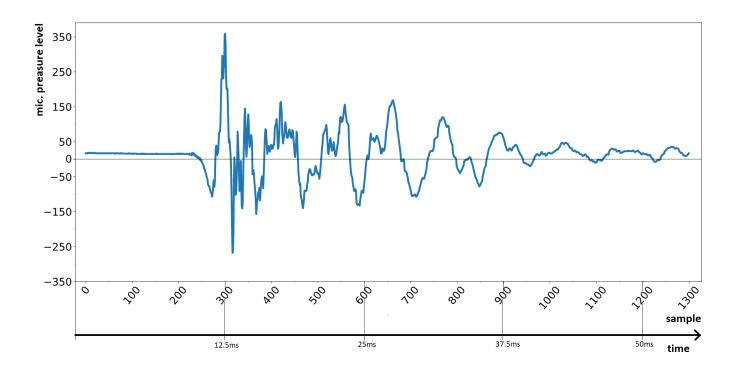


Source:

https://en.wikipedia.org/wiki/Pulse-density modulation



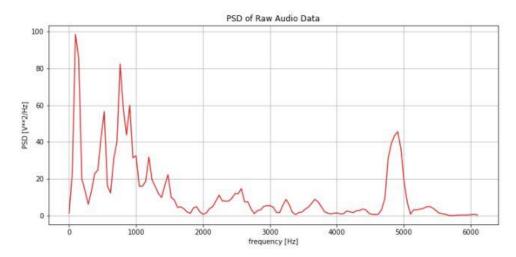
Accuracy was less lousy but still not good



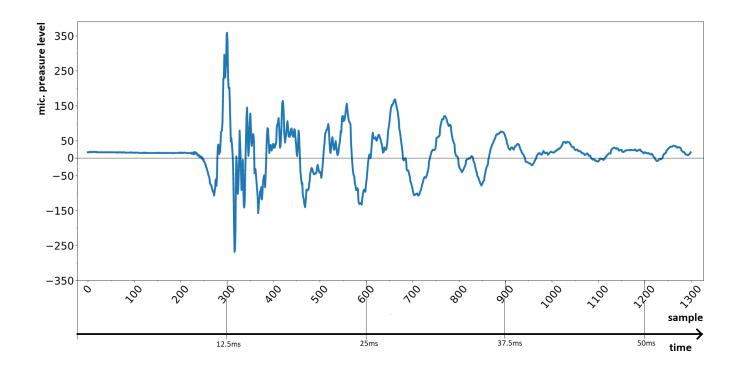
Accuracy was less lousy but still not good Moving to Power Spectral Density



- Time Series of input values (sound) was transformed to power spectral density
- Result, i.e. nose localization, got worse



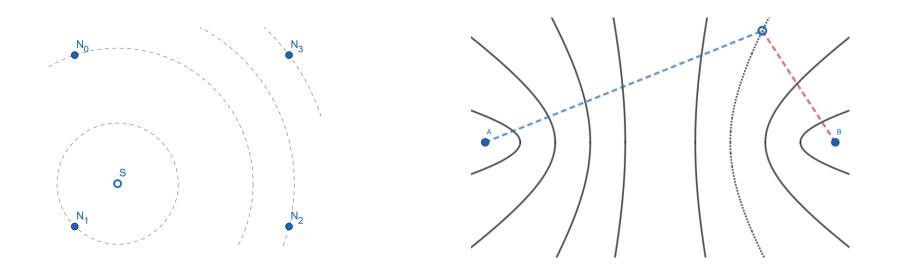




Accuracy was less lousy but still not good Non ML-Solution: Time Difference of Arrival (TDoA)



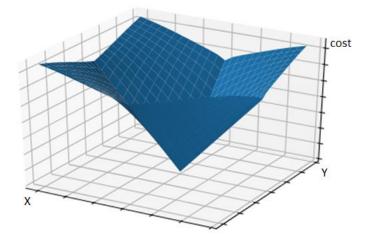
- Circles can not be used for localization => use hyperbolas
 - Time Difference of Arrival can be measured





Time Difference of Arrival (TDoA)

- > A cost function can be created
 - Mic. positions are known
 - Time Difference of Arrival is known
- Computation costly but lest accurate than ML





Machine learning (ML) vs. Conventional

- > ML provides better accuracy
 - Possible reason: Echo noise from the PVC surface of the LED matrix pad

Distance in cells	0	1	2	3	4	5
ML (hit rate)	24%	78%	91%	96%	99%	99%
Conventional (hit rate)	1%	10%	20%	34%	64%	81%

- > Conventional is faster
 - Neural networks are computational heavy

	Instructions	Execution time
ML	~13 Mio	~325 ms
Conventional	~2,1 Mio	~53 ms

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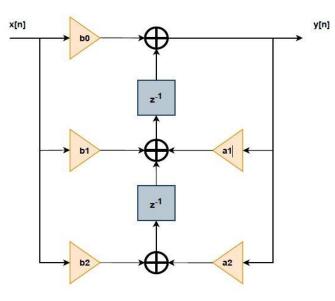


A Mixed NN and DSP solution Applying a High Pass Filter on the Inputs (1)



- Each Biquad stage implements a second order filter using the difference equation:
 - > b0 * x[n] + d1

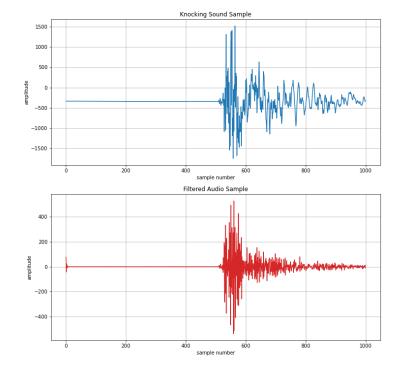
- d2 = b2 * x[n] + a2 * y[n]
- > Coefficients calculated using Octave.
 - > b0 = 0.3045
 - > b1 = -0.3045
 - > b2 = 0
 - > a1 = -0.3910
 - > a2 = 0



A High Pass Filter Biquad Cascade II T Filter with Direct Form II Transposed Structure

A Mixed NN and DSP solution Applying a High Pass Filter on the Inputs (2)





A Mixed NN and DSP solution

High Pass Filter substantially improved the results



Model Type	Mean Cell Distance	0 Cell	1 Cell	2 Cell	3 Cell	4 Cell
CNN without HP Filter	1.8	5.83%	48.33%	84.16%	94.16%	97.48%
CNN with HP Filter	1.4	10.10%	75.10%	91.66%	97.49%	99.99%

Cumulative Accuracy comparison for CNN model with the usage of High Pass Filter in place v/s raw audio signal. A total of 1696 training audio *datapoints*, trained for 50 epochs and learning rate of 0.01 were used. The cell results are based on 120 test *datapoints*

Open Question: Why could't this be achieved already with CNNs?

Outline





Software as Main course with Hardware as a Garnish and AI as Condiment

- > Findings on ML application
 - The data driven approach with ML was able to deal with body noise – it "detected" that effect
 - The best solution could be achieved with a mixed ML/DSP solution
- > Most effort went into SW development
 - Software Infrastructure for Micro including runtime system, driver, application code
 - ML compiler extension
- > Second most effort went in HW development
 - Setting up the system (!), building the SoC, building the AI accelerator
- Least effort went into ML
 - Most hereof was data collection and running experiments

