Adaptive Audio Filtering on Dynamic Signals

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Background/Motivation

- Denoising and localizing sources in the presence of noise is a common problem in auditory processing
- The human ear does this well. Computers? Not so much
- Noise interruptions - major issue with audio streaming (Zoom, Skype, etc.)

Goals

- To implement adaptive filtering techniques in an auditory focused, dynamic application in order to reduce or eliminate noise to a sufficient level
- Demonstrate and test various filtering techniques to propose what the best solution could be
- Propose and/or implement a hardware solution based on the best filter design

Methods

- Audio samples from Noizeus website (6)
- Algorithms processed audio samples at three different SNRs: 0 dB (baseline), 5 dB, and 10 dB
- We used three different filter types: time, frequency, and collaborative functional link
- Multiple speech sources were tested with various noise sources
- We will focus on one speech and noise combination (Restaurant, Train, Exhibition Hall) at 0 dB and 10 dB SNR levels

Background on Time Domain Filter: Kalman Filter

- Comprised of Hidden State Variable (System) and Observed Variable (Measurement) Equations
- Generalization of the RLS filter

Benefits:

- Allows for filter adaptivity (improvement on Wiener Filter)

Shortcomings:

- Must derive the state prediction correlation matrix through direct observation (must be known)

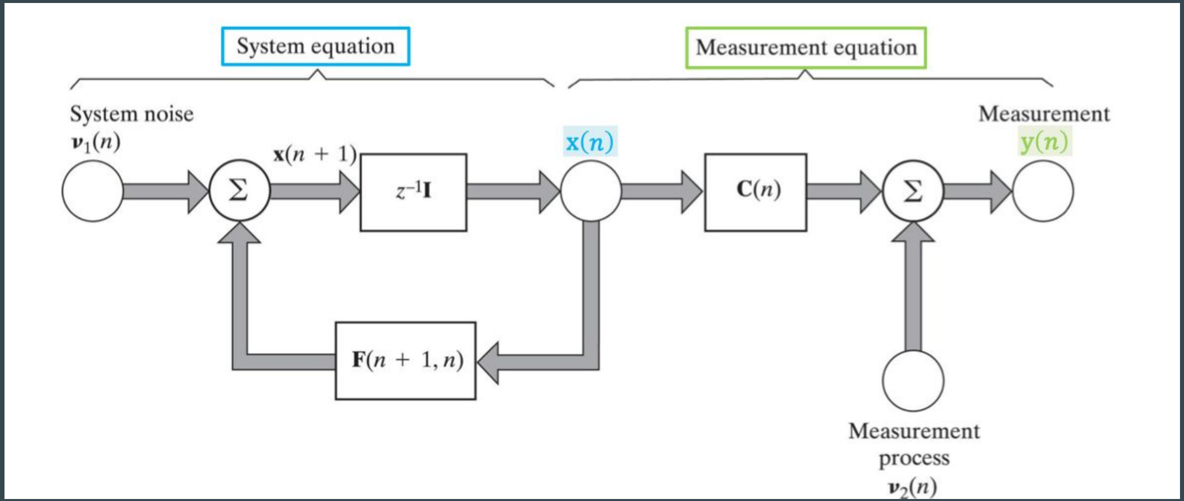


Figure 1: Kalman Filter Block Diagram (9)

Background on Frequency Domain Filter: Frequency Domain Kalman Filter

- Kalman Filter run in frequency domain
- More useful for filtering audio components and frequency specific adaptive filtering
- Has the same shortcomings of the time domain Kalman filter

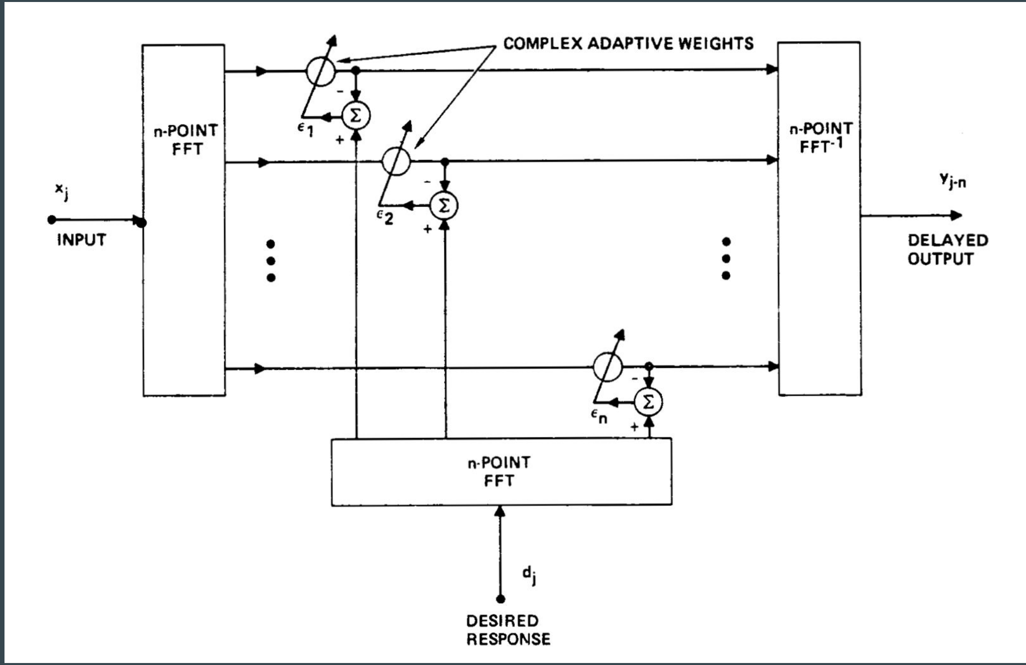


Figure 2: Adaptive Filtering in Frequency Domain (8)

Background: Collaborative Functional Link Adaptive Filter

- Filter architecture includes a linear filter and a non linear filter in “collaboration”
- Noise error correction feedback loop is driven by LMS algorithm
- d(n) driven by a combination of nonlinear and linear filtering

Benefits:

- Adaptivity and error information is driven by correlation between nonlinear and linear filters and information exchange between the two filters

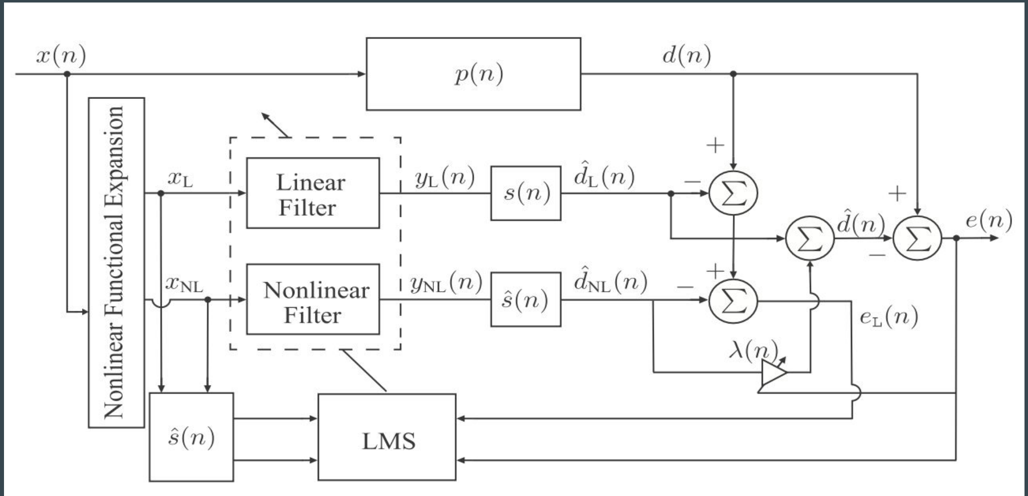
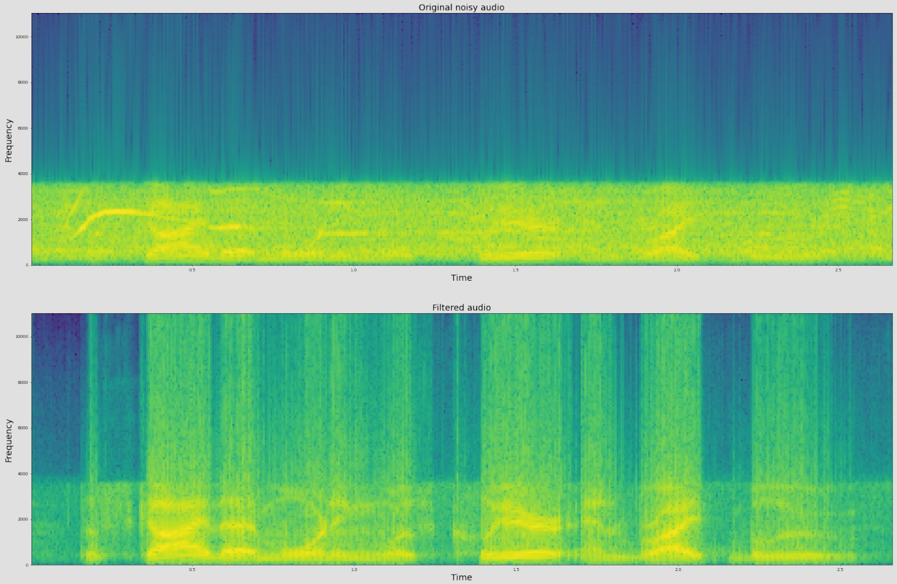
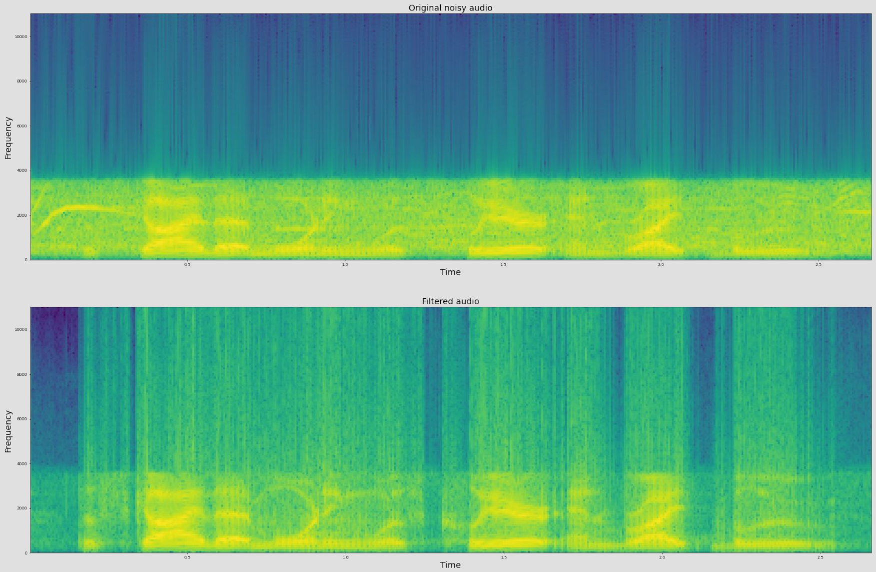


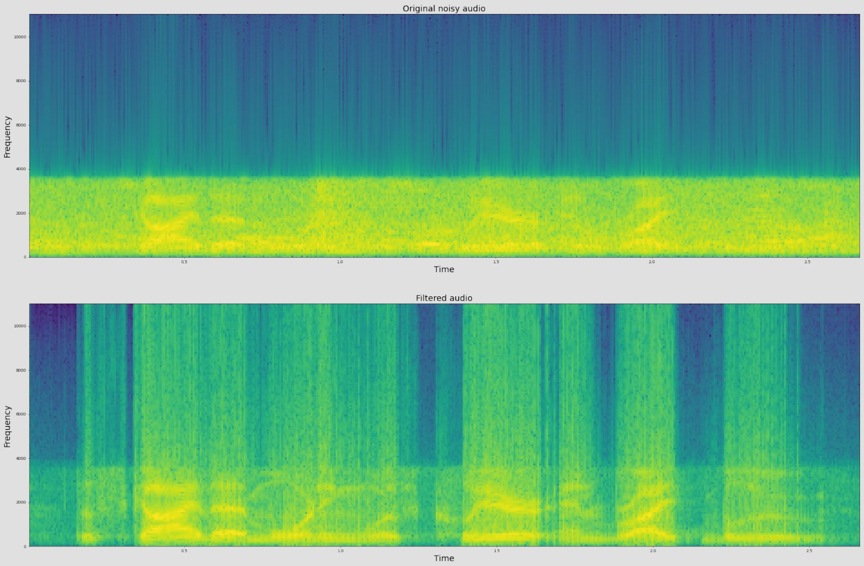
Figure 3: Block Diagram of Collaborative Function LInk Adaptive Filter (10)

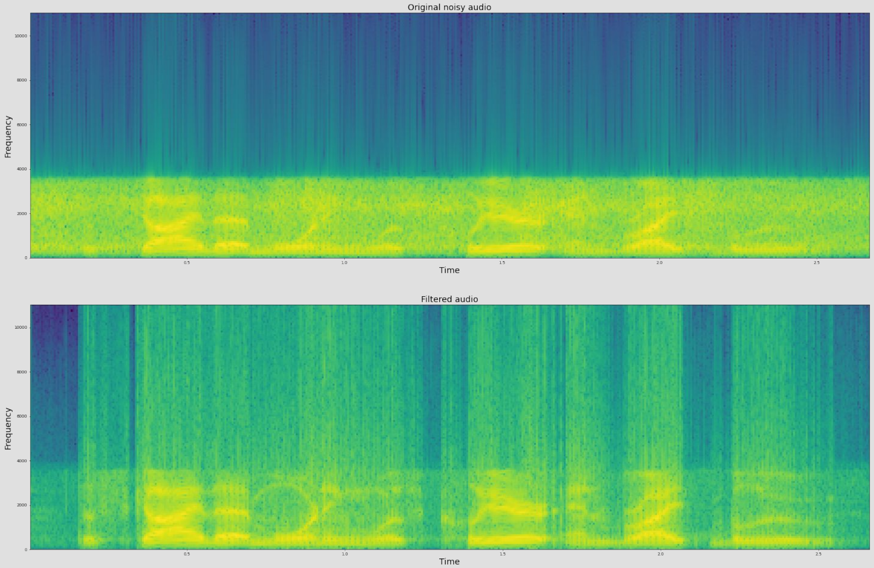
Results: Kalman Filter: Exhibition hall 0 dB and 10 dB



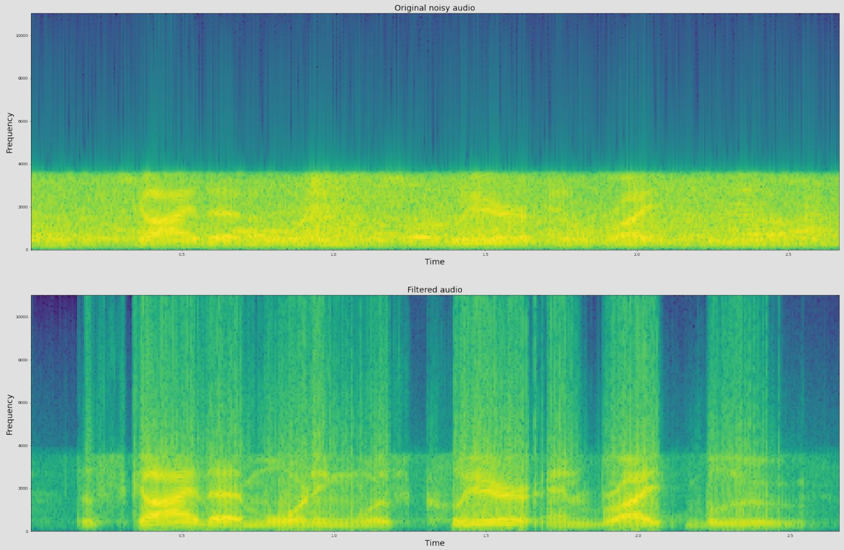


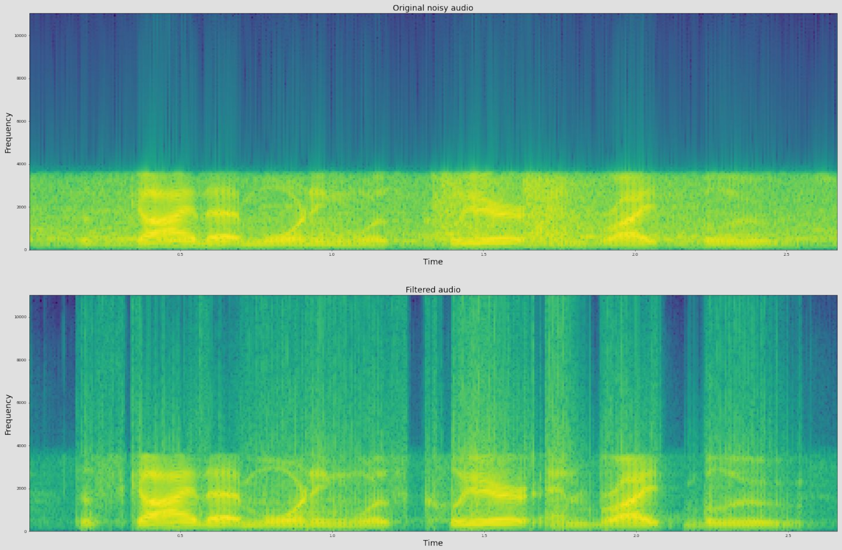
Results: Kalman Filter: Train (0 dB and 10 dB)



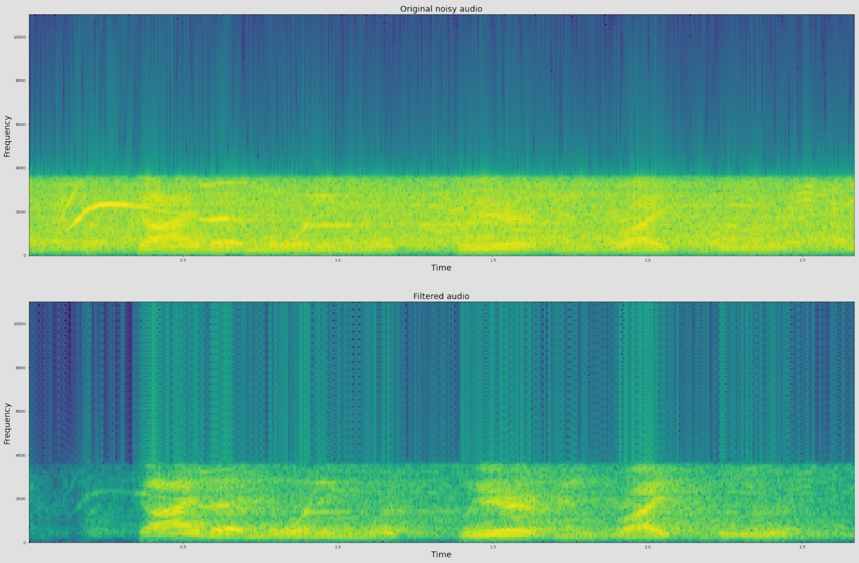


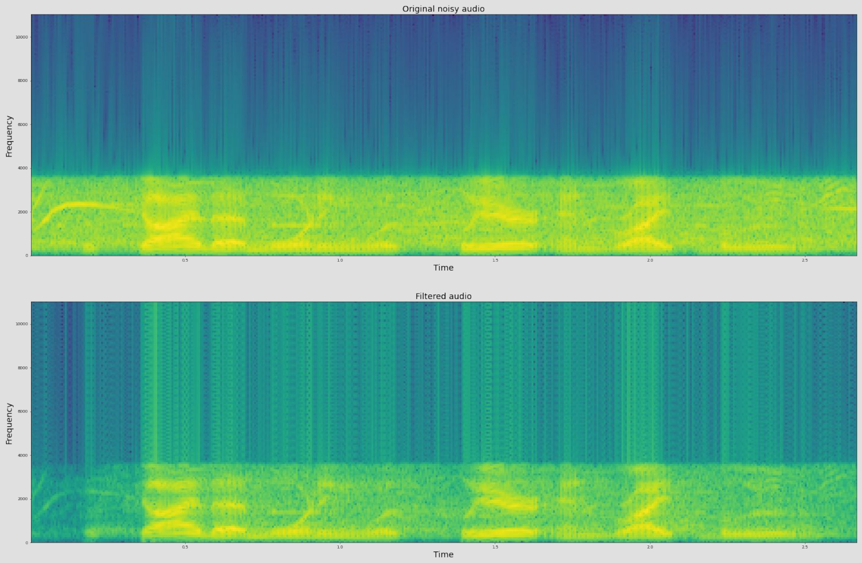
Results: Kalman Filter: Restaurant (0 dB and 10 dB)



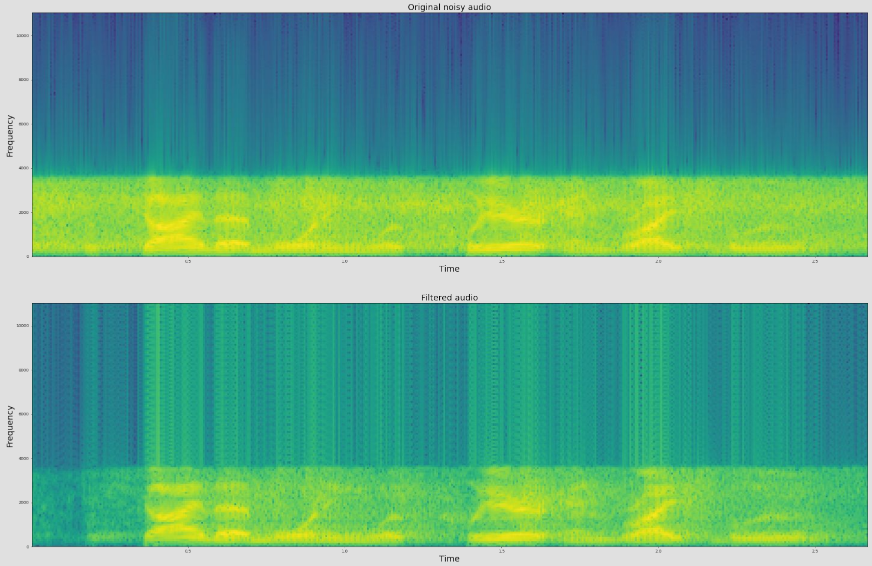


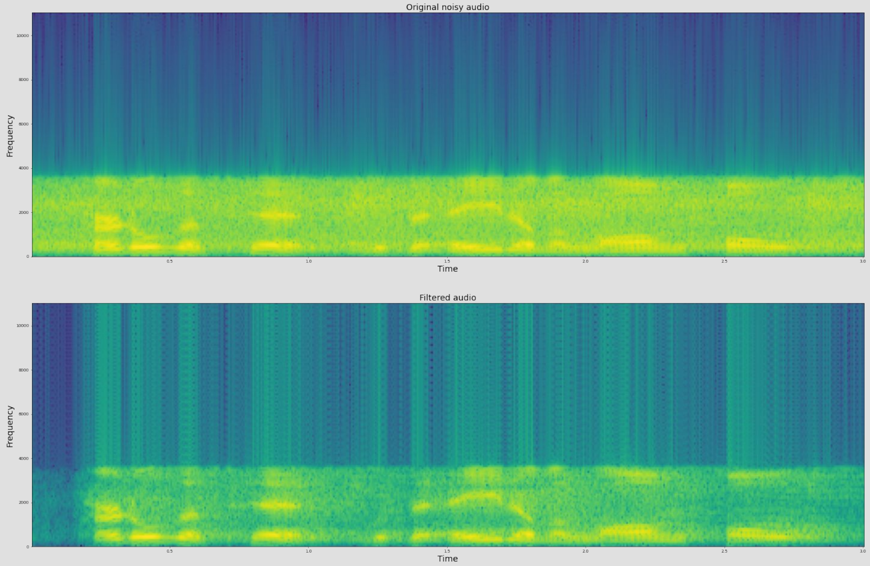
Results: Frequency Domain Kalman Filter: Exhibition: 0 dB and 10 dB)



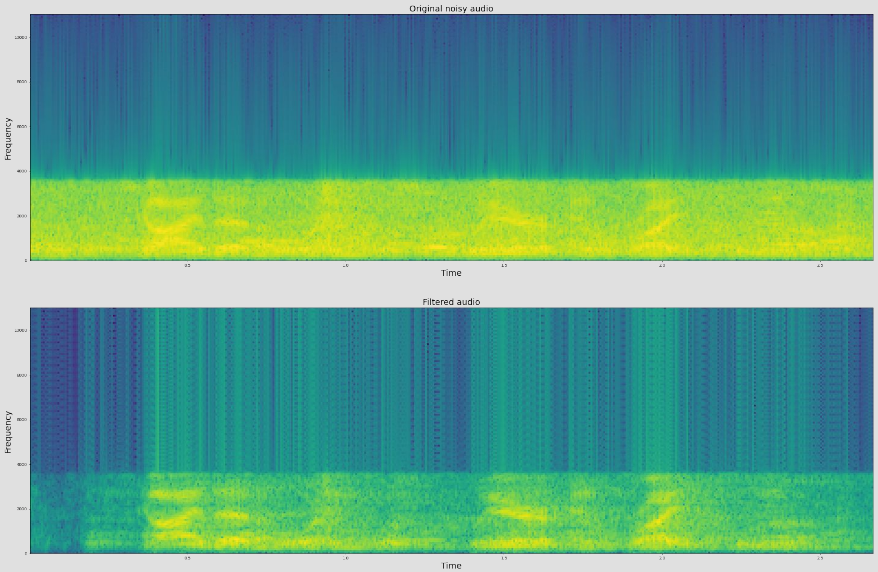


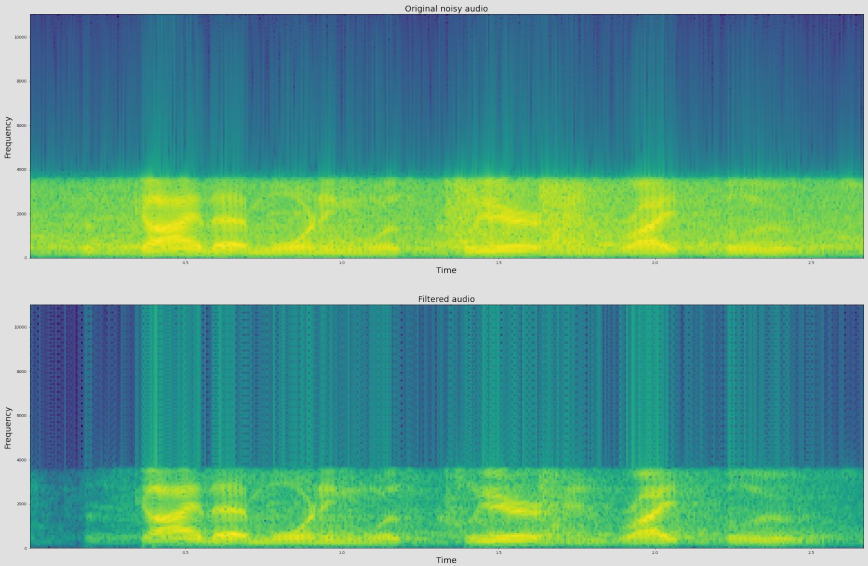
Results: Frequency Domain Kalman Filter: Train: 0 dB and 10 dB)



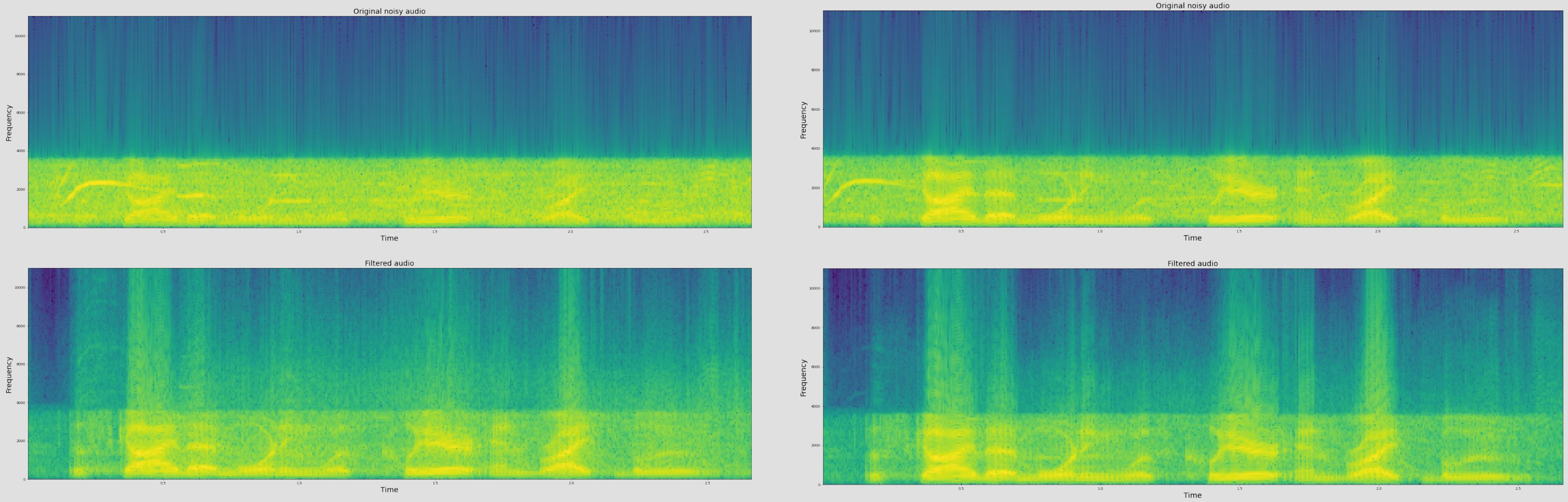


Results: Frequency Domain Kalman Filter: Restaurant: 0 dB and 10 dB)

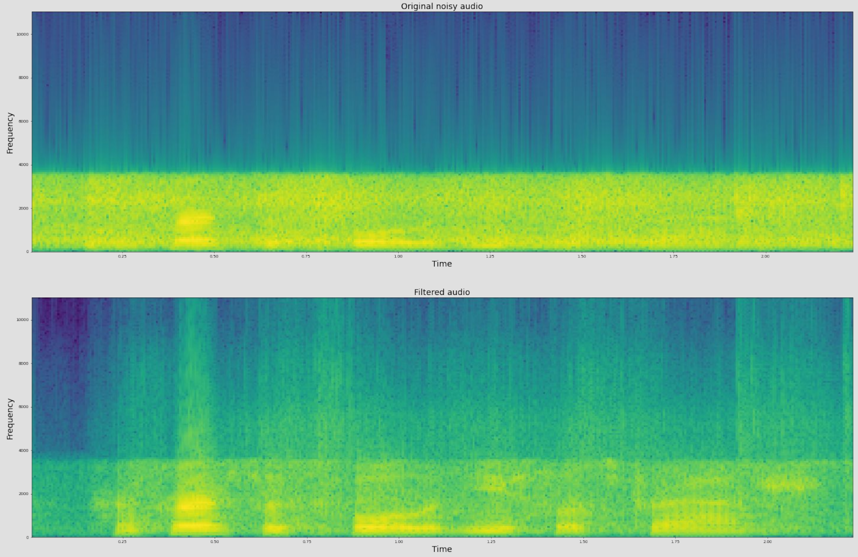


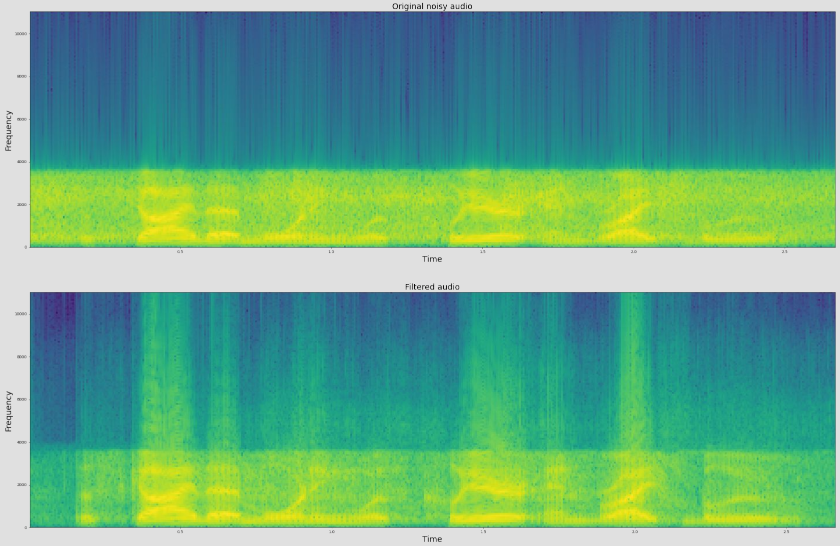


Results: Collaborative Functional Link Filter: Exhibition: 0 dB and 10 dB)

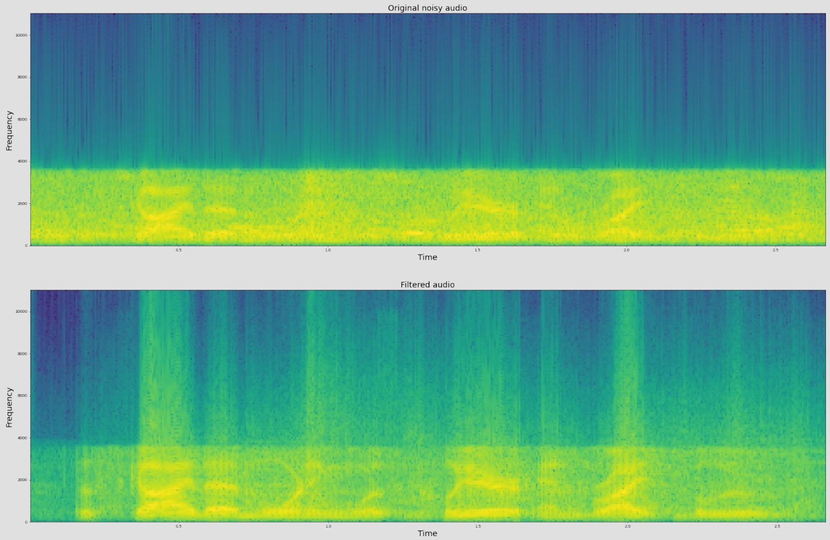


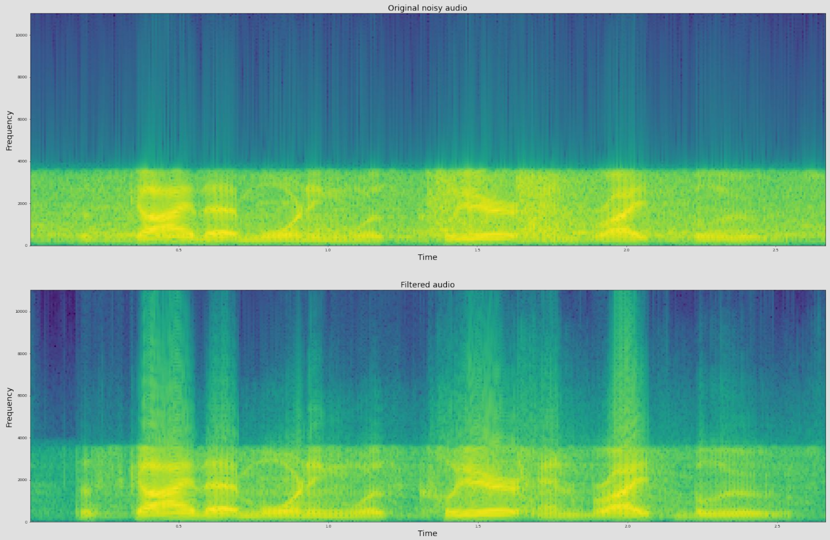
Results: Collaborative Functional Link Filter: Train: 0 dB and 10 dB)





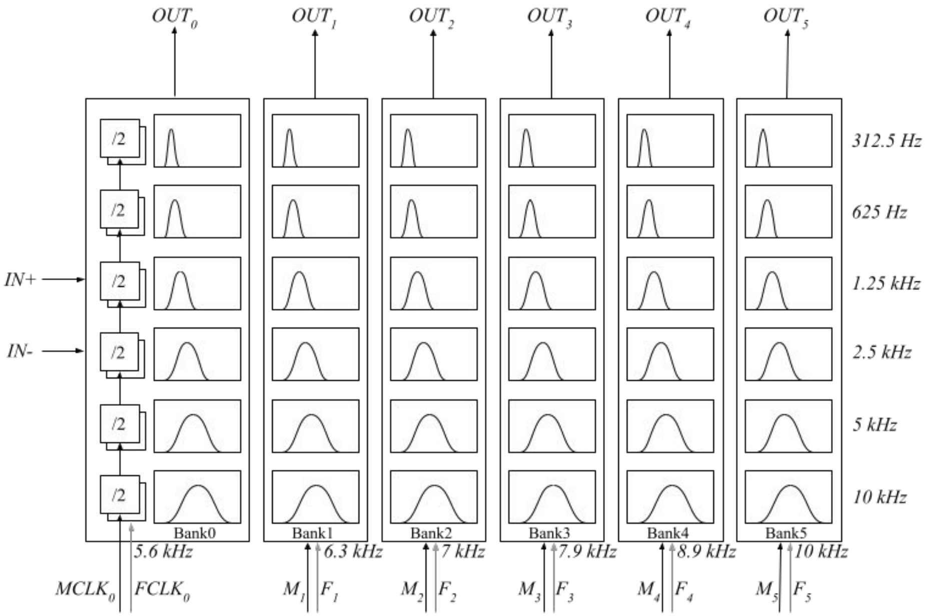
Results: Collaborative Functional Link Filter: Restaurant: 0 dB and 10 dB)



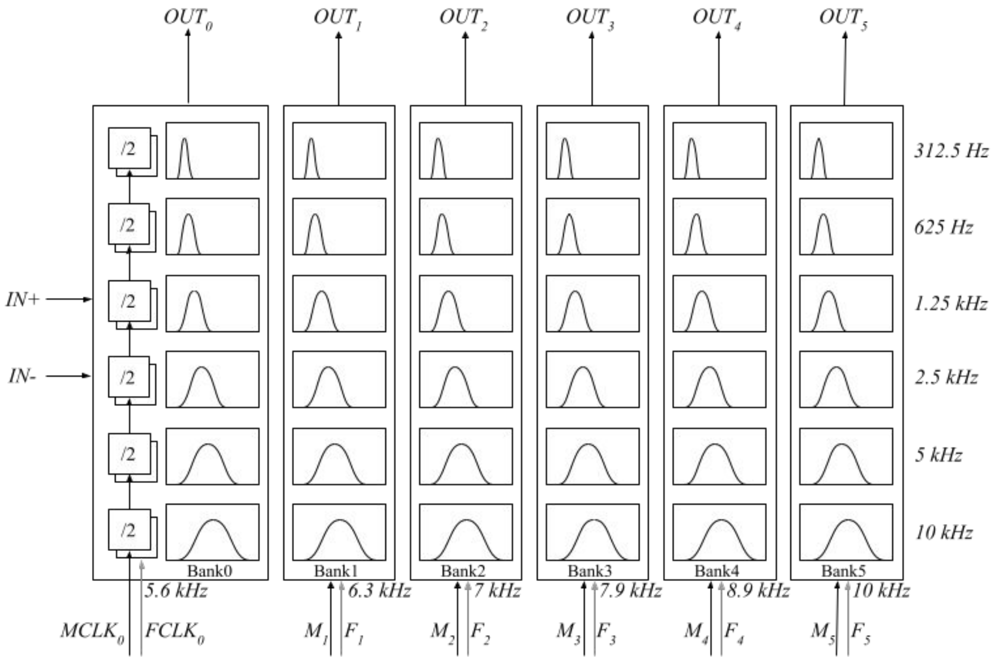


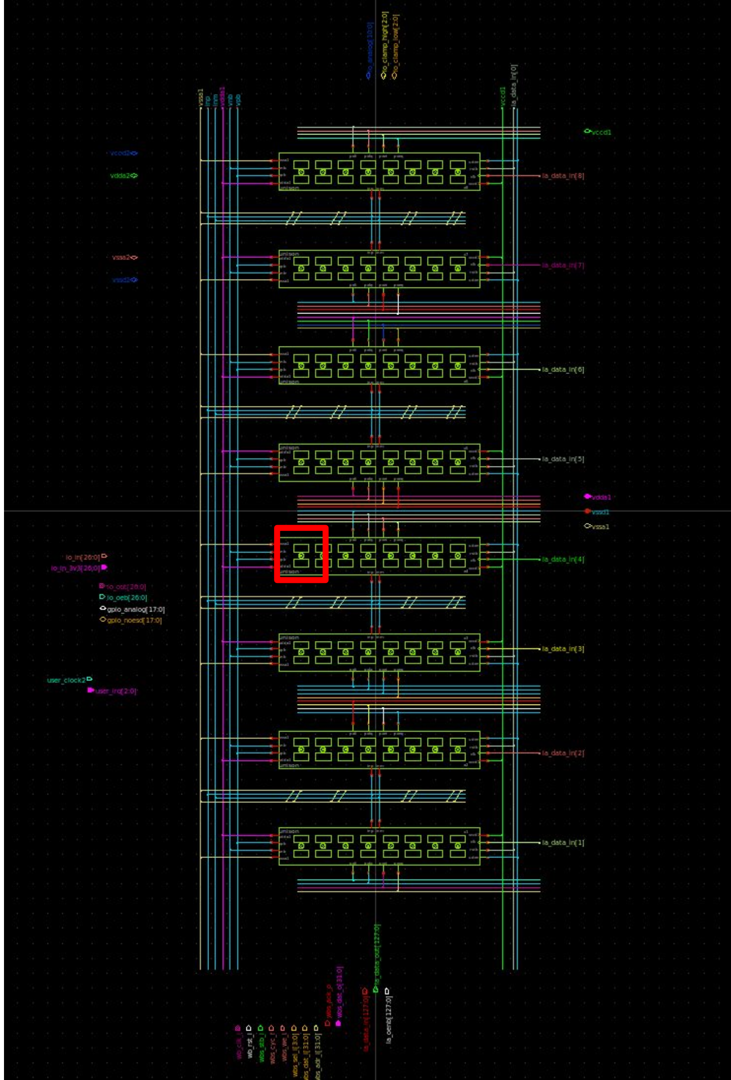
Do these filters work in low power hardware?

- In FPGA... maybe, but in IC - No!
- In cochlea design: there is a comparator which is very noise sensitive.
- This comparator is also a source of noise itself.
- The noise distribution can never be known before fabrication
- Hence, we need an adaptive filter to compensate for the error in each comparator.



Top Level Diagram of the chip





A low power hardware adaptation strategy

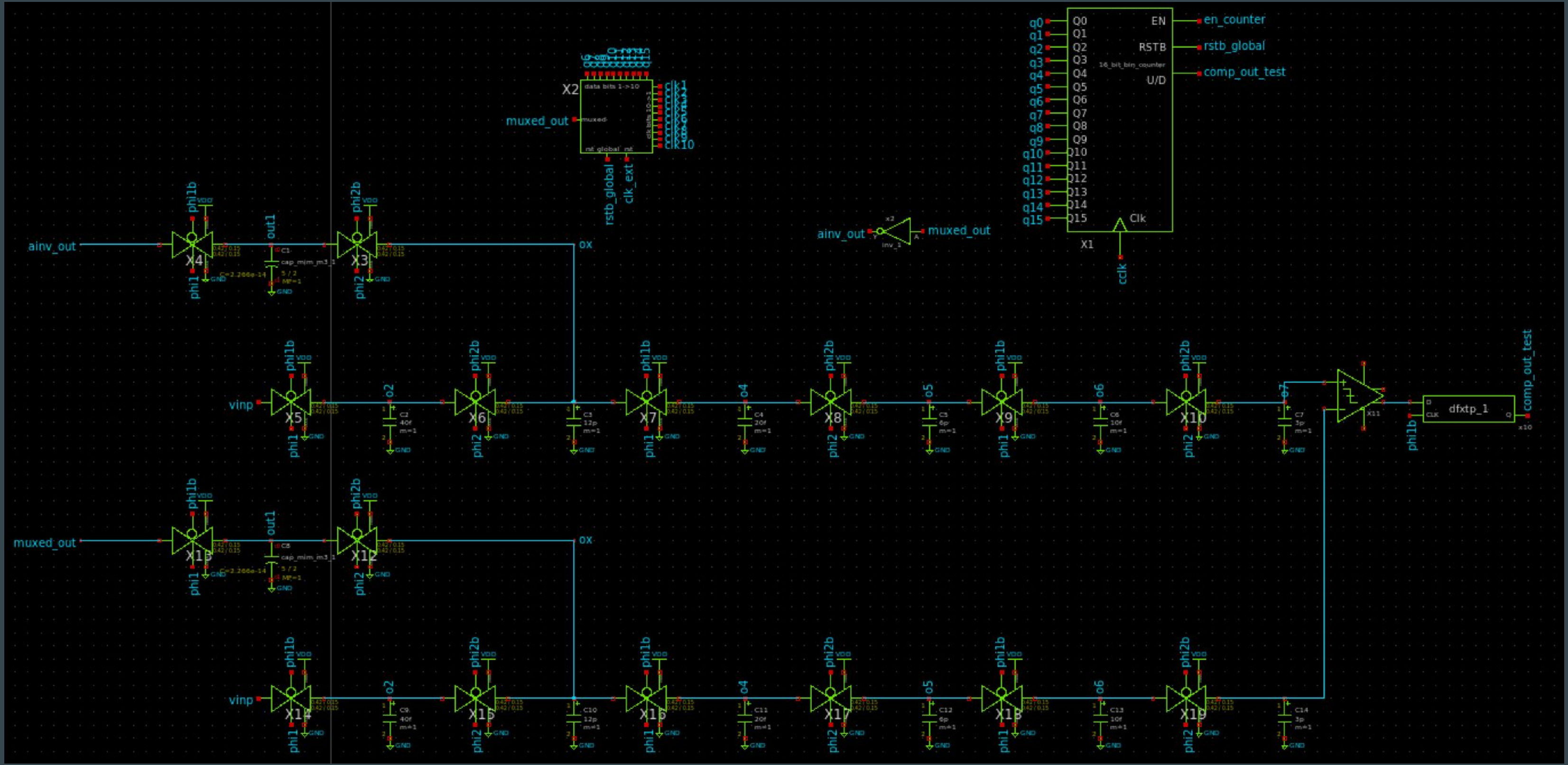


Fig 4. The block diagram of adaptive feedback

How does the adaptive compensation work?

1. Comparator is a sensitive piece of hardware that determines whether one signal is higher than the other.
2. If comparator is leaning more towards one signal, we count these increments.
3. Depending on the count, we select the 10 most significant bits in gray fashion
4. This is sent back through the filter as a compensation on top of the biggest capacitor in the circuit.
5. When the digital bits are passed through the low pass filter, we are only left with the DC component of the feedback which changes the common mode of the signal.

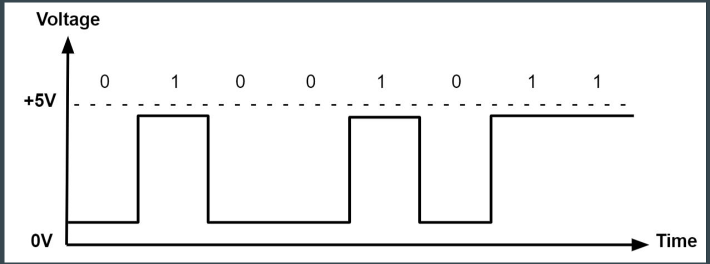
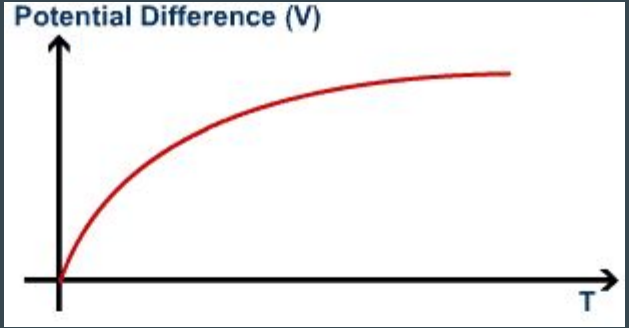


Fig 5. Digital to analog conversion (Monolithic power systems)

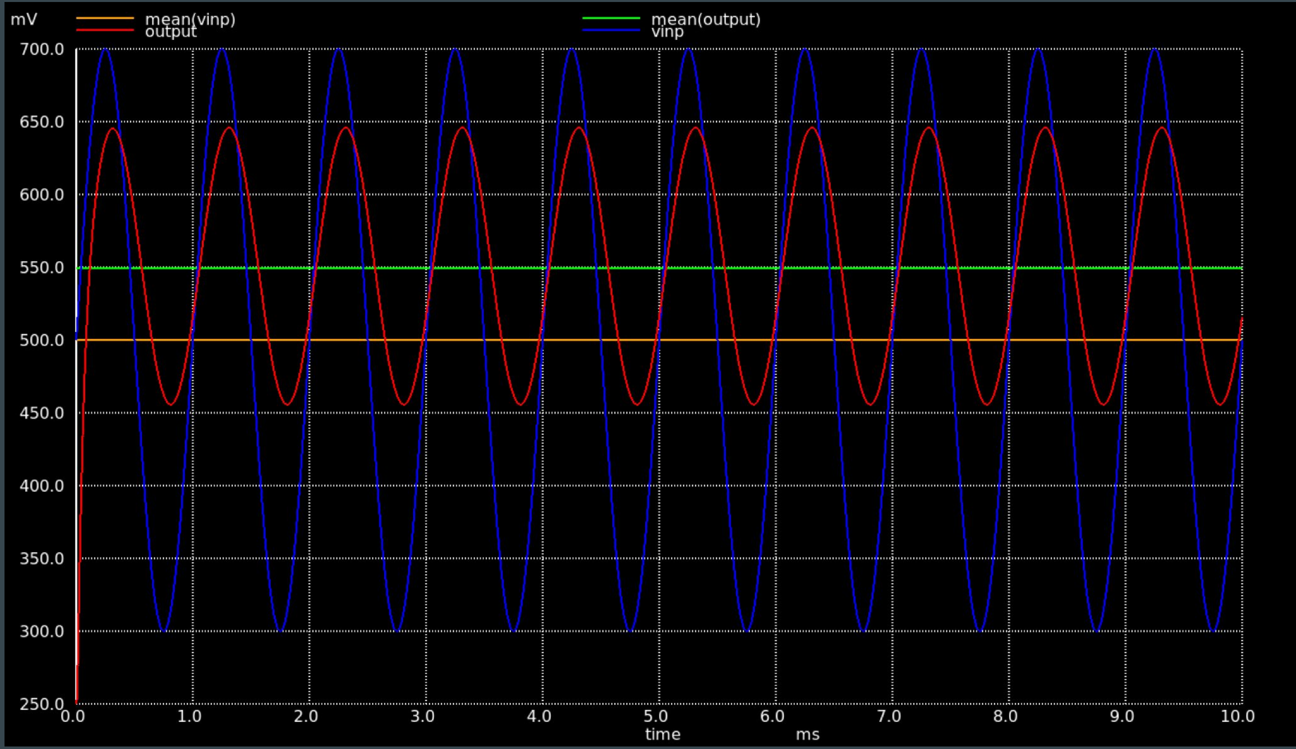


Output of the compensated filter

- Initial common mode of the signal: 500mV

- Compensated common mode of the signal: 550mV

This change is extremely important for the sensitivity of our comparator.



More to add in the final report:

- Simulink models for various filters
- Details of more adaptive filters tested and their results

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Thanks!