Active Noise Control of Speech in Headphones
using Linear Prediction

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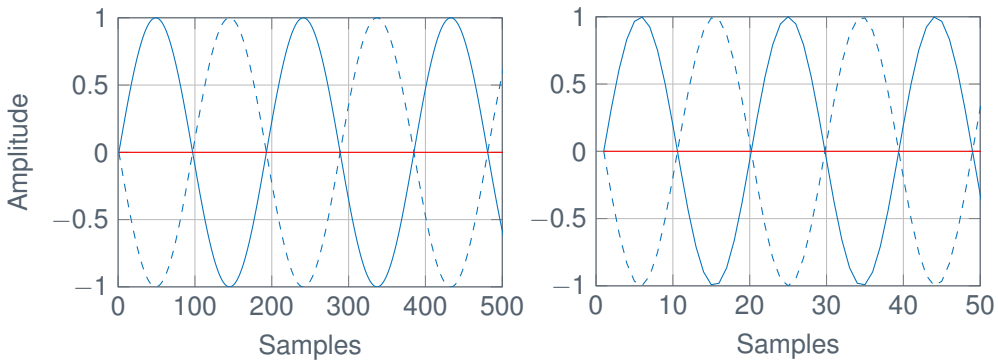
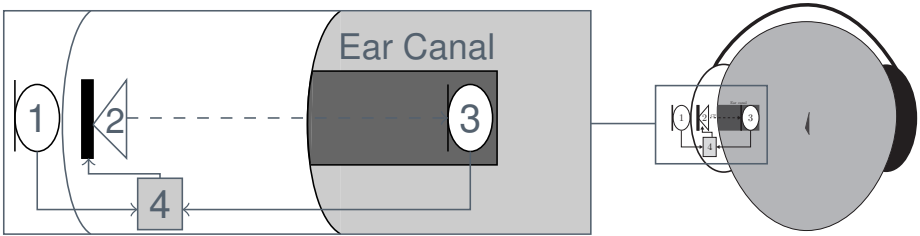
Agenda

- The basic theory of ANC
  - 250 Hz
  - 2500 Hz

Original signal
Counterphase signal
Error



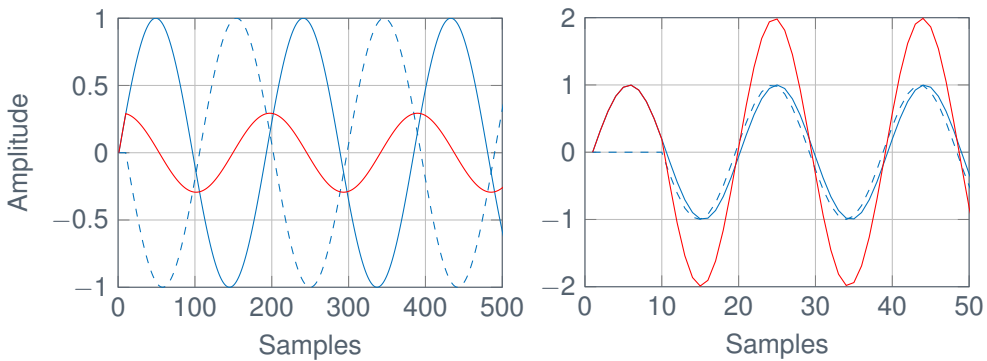
- Headphone cups attenuate high frequencies passively
- Lower frequencies must be attenuated actively
- Feedforward system
  - 1: Reference microphone
  - 2: Headphone loudspeaker
  - 3: Error mirophone
  - 4: Digital signal Processor (DSP)



- Feedforward problem
- Sampling and reconstruction delay.
  - Anti Aliasing filter
  - Reconstructions filter
- The measured delay of a Sigma Delta converter TLV320AIC3204
- Spacing between microphones
  - Min: 75.5 mm
  - Max: 302 mm

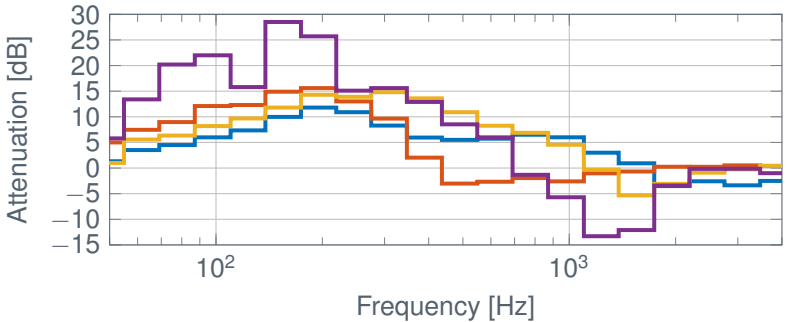
f_s [kHz]	48	96	192
Delay [μs]	900	450	225
Delay [samples]	43	43	43

- Counter phase signal delayed 10 samples
  - 250 Hz
  - 2500 Hz



- Signal Characteristics
  - Periodic Signals
    - Periodic
    - Strict Sense Stationary (SSS)
  - Speech Signals
    - Quasiperiodic
    - 50 Hz – 4000 Hz
    - Can be assumed Wide Sense Stationary (WSS) for 20 *m*s – 30 *m*s
- Periodic noise is easy to cancel
- Speech noise is difficult to cancel

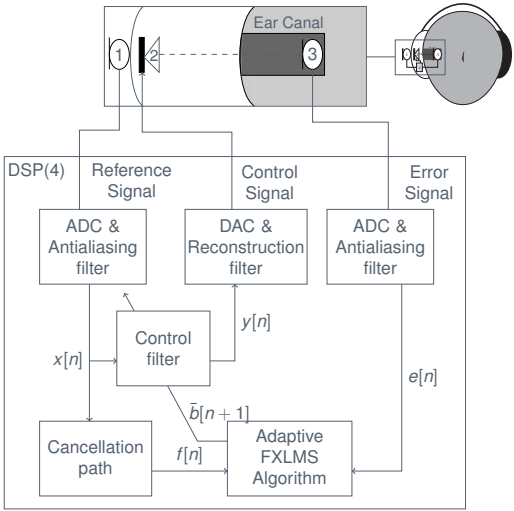
- How well does the consumer headphones attenuate?
  - Denon AH-GC20 2.200 kr (2016)
  - Bose QC25 2.799 kr (2016)
  - Bose QC15 2.696 kr (2011)
  - BeoPlay H8 3.495 kr (2016)



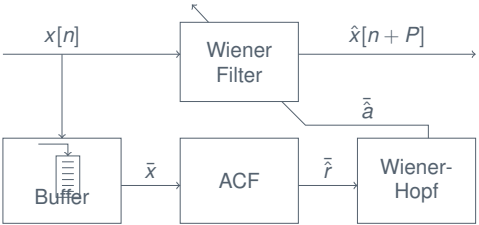
- ANC attenuate infinitely ideally
- Delays are introduced by sampling and reconstruction
- Periodic signals can be attenuated infinitely
- Speech signals are not attenuated very well

Combining a feedforward ANC algorithm with a Linear prediction (LP) scheme to compensate for delay.

- Control filter
  - Transfer function from (1) to (2)
  - Adaptive
- Cancelation path
  - Transfer function from (2) to (3)
  - Linear Time Invariant
- Adaptive FXLMS Algorithm
  - Optimization problem



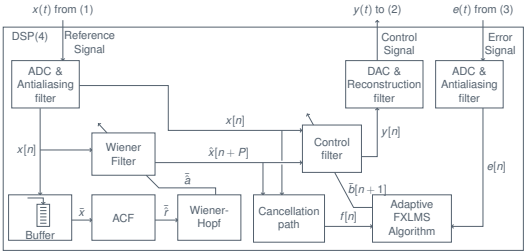
- Auto Correlation Function estimation
  - Frame length N
  - Overlap O
- Wiener hopf equation: $\hat{R}\bar{a} = -\bar{\hat{r}}_x$
  - Inverting matrix
  - Levinson Durbin
- Wiener filtering in cascade
  - Prediction order P



- Multirate Processing for reducing conversion delay
  - High sample rate
  - Low processing rate
- Smaller prediction order
  - 10 instead of 43

f_s [kHz]	48	96	192
Delay [μs]	900	450	225
Delay [samples]	43	43	43

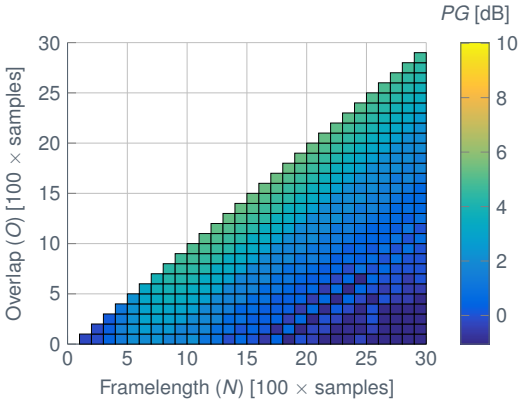
- Input for control filter and CP
  - $x[n]$
  - $\hat{x}[n+P]$
- CP delayed for compensation of error microphone delay



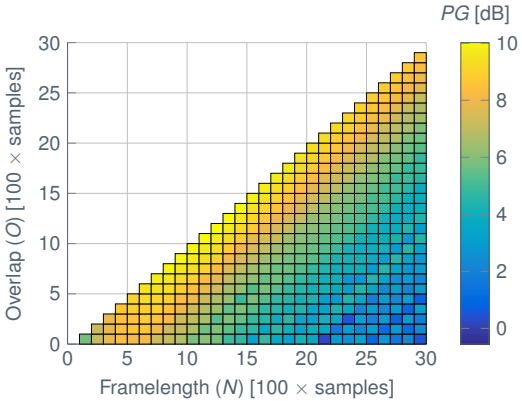
- Delays are introduced due to sampling and reconstruction
- Delays are reduced using multirate processing
- Compensation by Linear Prediction using Wiener filtering
- Noise cancelling using a feedforward FXLMS algorithm

- Simulink
- Prediction Gain
- Cross correlations
- Not entirely sure what to put here

- Prediction order P = 43
- Optimal parameters
  - Frame length N = 1600
  - Overlap O = 1500
- Prediction Gain PG = 5.4 dB

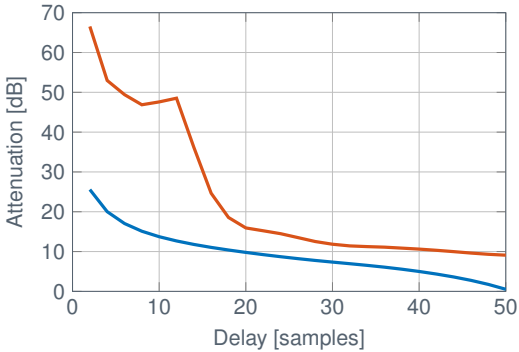


- Prediction order P = 10
- Optimal parameters
  - Frame length N = 1200
  - Overlap O = 1100
- Prediction Gain PG = 10 dB



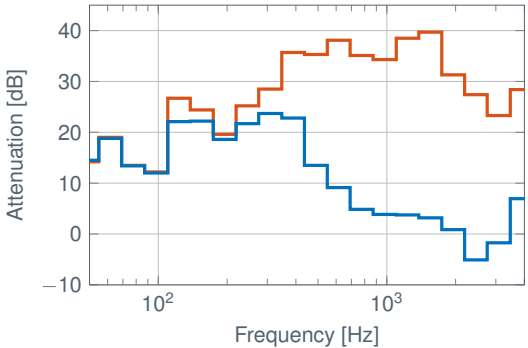
Simulation Results
Attenuation Performance

- ANC attenuation with varying system delay
  - Feedforward FXLMS
  - Feedforward LP FXLMS



Simulation Results
Frequency response

- Frequency response
  - Feedforward FXLMS
  - Feedforward LP FXLMS



- Computational cost of System (instructions pr. sample)
  - Linear Prediction: 55000
  - Feedforward FXLMS: 4000
  - Multirate: ≤ 100
- Figure missing of different DSP maximum instruction pr sample at 48 kHz

Questions?

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