CHARACTERIZATION AND APPLICATION OF ECHO CANCELLATION METHODS

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Motivation

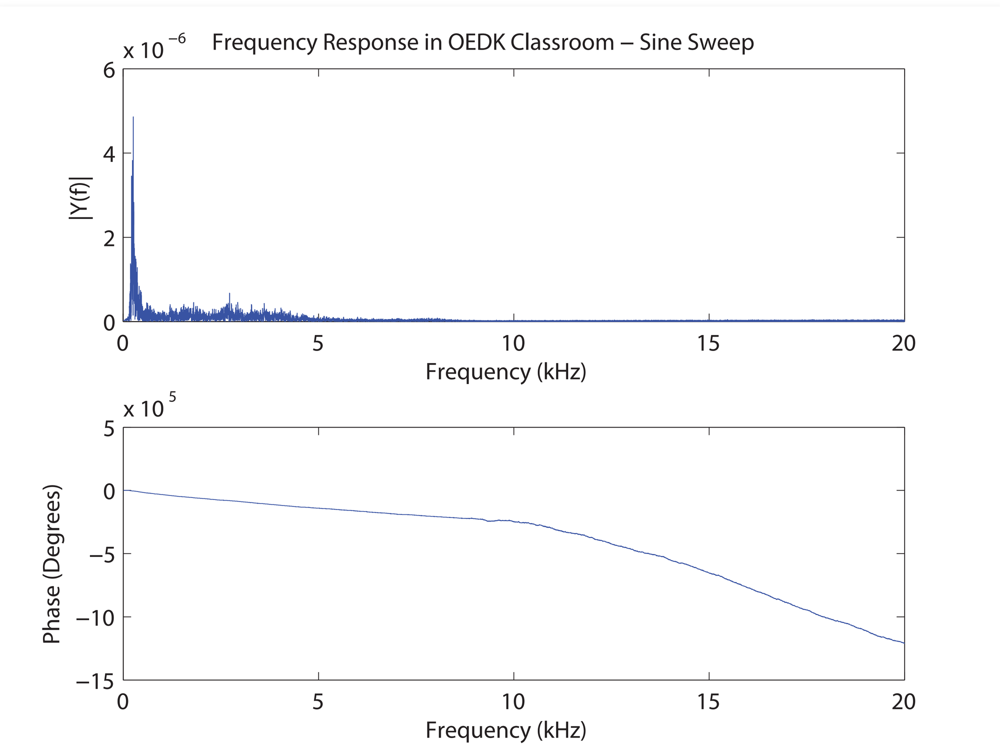
The quality of audio recordings is largely dependent on the acoustic properties of the surrounding space. If a room is acoustically active, then the quality of audio recordings will decrease. Take for example the Oshman Engineering Design Kitchen (OEDK) classroom, which suffers from excess acoustic reverberation. Because physically altering the room is expensive, we approached the problem from a digital signal processing point of view.

Implementing an echo cancellation filter to the microphone's signal will improve the quality of the presentation's audio when being played back by students or other future listeners to the presentations that have been recorded in the classroom.

Introduction

Echoes in recorded presentations in the OEDK classroom are distracting and unpleasant. Our team will:

- research and implement methods of echo cancellation to be used for rectifying the classroom's acoustic problem.
- make a recommendation for approaching echo cancellation in OEDK classroom



Deconvolution

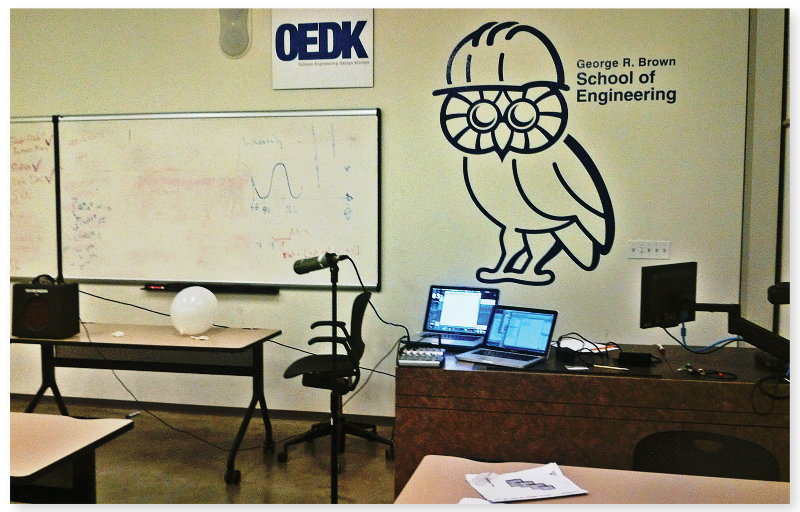
Finding the Impulse Response

In characterizing a system at all frequencies, one would ideally excite it with a Dirac delta function, an infinitely high and infinitely thin signal spike. We approximated and recorded the impulse response of the room using the following methods:

- balloon popping: simulating impulses
- pseudo-dirac: sounding Matlab's dirac() function to generate a single click from a guitar amplifier
- sine-sweep method: generating a logarithmically increasing sine wave over a desired frequency range and performing a deconvolution using the inverse chirp filter

$$x(t) = \left[\sin \frac{T\omega_1}{\ln(\frac{\omega_2}{\omega_1})} \left(e^{\frac{t}{T}\ln(\frac{\omega_2}{\omega_1})} - 1 \right) \right]$$

where ω_1 is the initial radian frequency and ω_2 is the final frequency of the sweep of duration T.



Deconvolution Method

- Assume the echoed signal is the convolution of the near-end speech signal and the impulse response of the room (the room is linear and time-invariant)
- Deconvolve the echoed signal with the impulse response after room's ambient noise is removed

Adaptive Filtering

Adaptive Filters

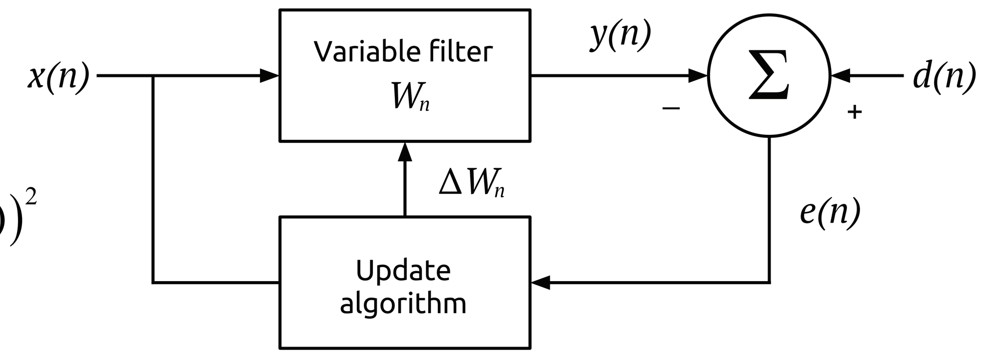
The impulse response of the room is not constant; people move around and the environment changes. Time-varying adaptive filters are more sensible for locations such as the OEDK classroom. Adaptive filters take in a desired signal and an input signal and update their parameters using feedback driven by the error signal. For our project, we trained adaptive filters using signals from a low-echo environment using three algorithms to filter out echoes in an echoic room and in the OEDK classroom.

Least Mean Squares (LMS) Algorithm

The LMS Algorithm estimates the filter coefficients that minimizes the expected value of the cost function $(d(n)-y(n))^2$ using its gradient. There are three steps per iteration:

1. Calculate output of filter: $y(n) = w^T(n)x(n)$
2. Calculate estimate of error: $e(n) = d(n) - y(n)$
3. Adjust FIR coefficient weights: $w(n+1) = w(n) + 2\mu e(n)x(n)$

Computational complexity: $2N$ additions, $2N+1$ multiplications



Adaptive Filtering Block Diagram

$x(n)$ – input signal
$e(n)$ – error
$y(n)$ – output signal
μ – step size
$w(n)$ – FIR coefficients
$d(n)$ – desired signal

Normalized Least Mean Squares (NLMS) Algorithm

Normalized LMS allows for a different step size $\mu(n)$ for each iteration.

Computational complexity: $2N$ additions, $3N+1$ multiplications

Block Least Mean Squares (BLMS) Algorithm

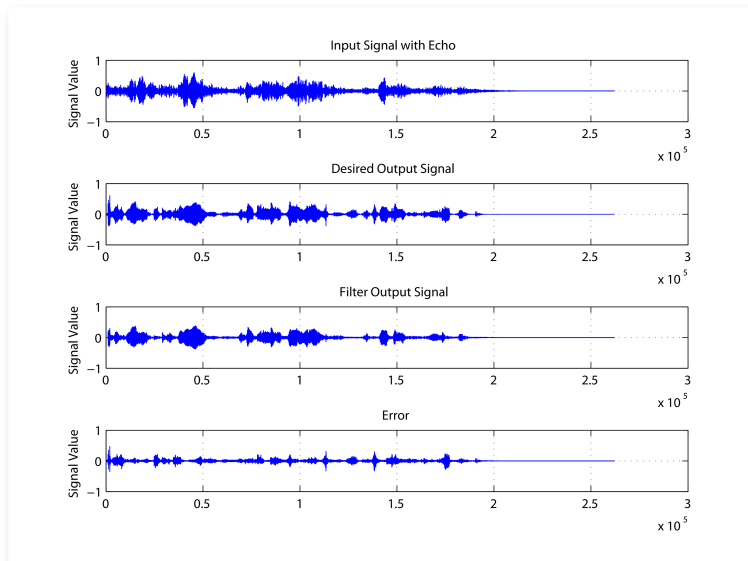
Instead of computing by sample, the Block LMS algorithm calcutes in blocks of data of length B reducing computation cost and improving the convergence rate.

Computational complexity: $\frac{4N}{B} + \left(\frac{N}{B} + 3\right)\log_2(2B)$

Echo Cancellation Results

Conclusions

- Sine-sweeping covers more frequencies; Psuedo-dirac is unreliable
- Deconvolution method is over-sensitive to noise and depends on an unrealistic, static environment. Recommend adaptive filter
- NLMS has better average error RMS but LMS has better signal quality
- LMS best for short signals



Echo Cancellation of Male Speech Using LMS Method

Error Signal Root Mean Square (RMS) of each Algorithm

Signals	LMS	BLMS	NLMS
Echoic Classroom			
Impulse	0.0240	0.0490	0.0182
Female Speech	0.0705	0.1328	0.0385
Male Speech	0.0235	0.0564	0.0133
OEDK Classroom			
Impulse	0.0253	0.0480	0.0166
Female Speech	0.0560	0.1141	0.0373
Male Speech	0.0223	0.0590	0.0117
Average RMS Value	0.0369	0.0752	0.0226

Acknowledgements and References

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OEDK Oshman Engineering Design Kitchen
RICE ece redefining limits
RICE