### **Active Noise Cancellation**

Using the LMS and FxLMS algorithms



November 7, 2016

**Marko Stamenovic** 

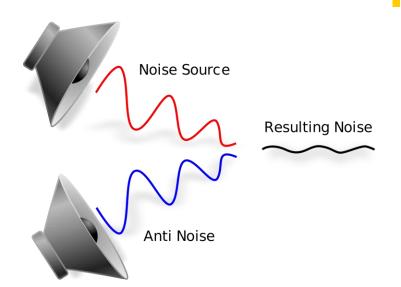
Recurse Center Lightning Talk https://github.com/markostam

### **Background and Motivation**

Let's start at the beginning



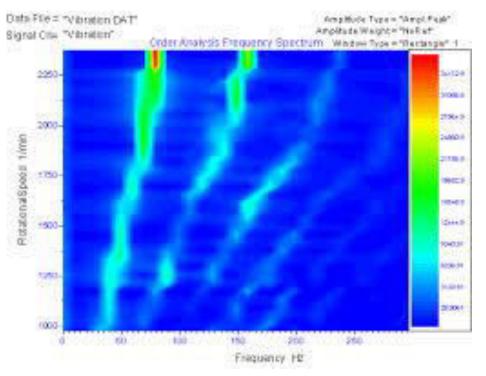
#### **Background**



#### **Active Noise Cancellation (ANC)**

- Technique which uses destructive interference to cancel unwanted noise signals.
- Machine learning algorithms are employed to quickly learn the characteristics of the unwanted signal in near real time.

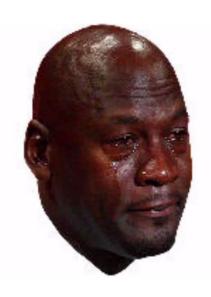




# Benefits over passive attenuation

- Lighter weight and smaller than passive noise attenuation.
- Targets specific frequencies.
- Actively adapts to offending noise spectrum.





#### Limitations

- Expensive.
- Not good with impulsive sounds.
- Relatively complex (requires specialized hardware and software).
- Requires constant power supply.

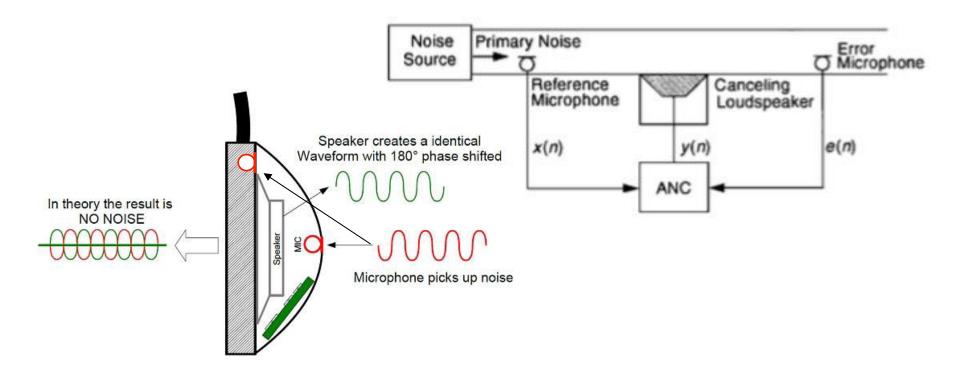
## **System Overview**

Equations & block diagrams ahead



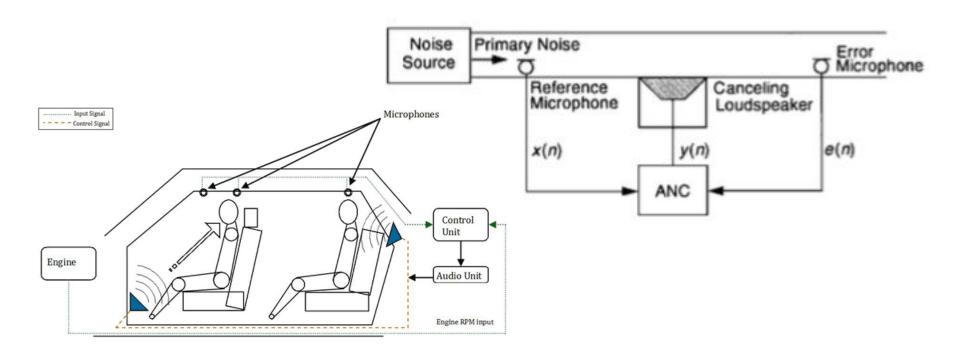


#### **Conceptual System Overview**





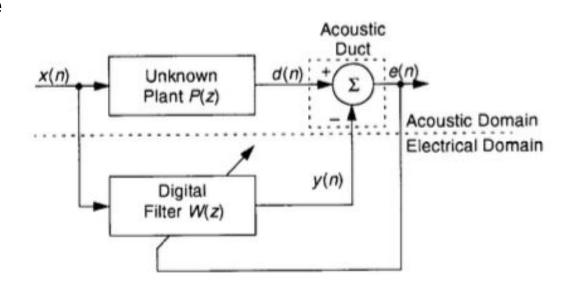
#### **Conceptual System Overview**





#### LMS for ANC Block Diagram

- Unknown plant P(z) is the transfer function b/w engine and passenger in car our outside world and ears in headphones.
- As e(n) approaches 0,
   W(z) becomes equal to
   P(z)





#### **LMS for ANC Equations**

$$e(n) = d(n) - \mathbf{w}^{T}(n)\mathbf{x}(n)$$
$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu\mathbf{x}(n)e(n)$$

Least Mean Squares (LMS)
 algorithm. Uses stochastic
 gradient descent to minimize
 e(n). Simple and powerful.

Unknown d(n) +  $\Sigma$  ; e(n) Acoustic Domain

Digital Filter W(z)

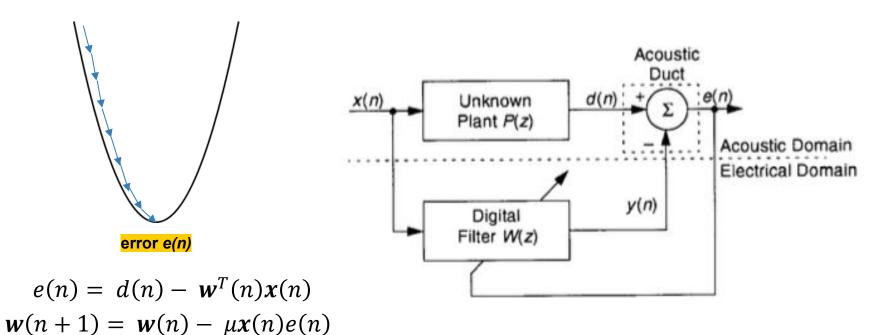
Acoustic

Duct

•  $\mu$  = Learning rate



# LMS for ANC Block Diagram and Equations





#### Cool story, show me dez code

Setup stuff and synthetic transfer function

```
interrupt void interrupt4(void)
    {
        short i;
        float input, refnoise, signal, signoise, wn, yn, error;
        codec_data.uint = input_sample();
        refnoise =(codec_data.channel[LEFT]); // noise sensor
        input = refnoise;

        for (i=0; i < N; i++)
        // filter refnoise to emulate transfer
        // function of firewall (3rd order lp filter)
        {
            wn = input - a[i][1]*w[i][0] - a[i][2]*w[i][1];
            yn = b[i][0]*wn + b[i][1]*w[i][0] + b[i][2]*w[i][1];
            w[i][1] = w[i][0];
            w[i][0] = wn;
            input = yn;
        }
}</pre>
```

**Meat of the ANC**: filter the signal & update weights

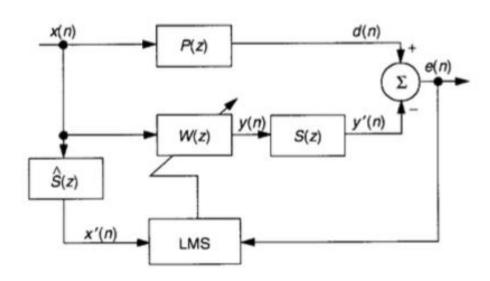
```
yn=0.0;
x[0] = refnoise;
for (i = 0; i < N; i++) // compute adaptive filter output (w'x)</pre>
     yn += (weights[i] * x[i]);
error = - yn; // compute error
for (i = N-1; i >= 0; i--) // update weights and delay line
     weights[i] = weights[i] + mu*error*x[i];
     x[i] = x[i-1];
codec_data.channel[LEFT]= ((uint16_t)(error));
codec data.channel[RIGHT] = ((uint16 t)(error));
output sample(codec data.uint);
return;
```

(also update the delay line and output the filtered signal)



#### **FxLMS Block Diagram**

$$e(n) = d(n) - \mathbf{w}^{T}(n)\mathbf{x}(n)$$
$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu \mathbf{x}'(n)e(n)$$

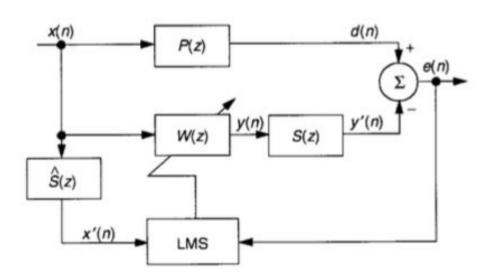


- One thing missing from the LMS model in practical ANC applications is the path from the speakers to the ear for the correction signal.
- This is called the Secondary Path or S(z).
- The FxLMS or Filtered LMS algorithm takes care of the secondary path.



#### **FxLMS Block Diagram**

$$e(n) = d(n) - \mathbf{w}^{T}(n)\mathbf{x}(n)$$
$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu \mathbf{x}'(n)e(n)$$

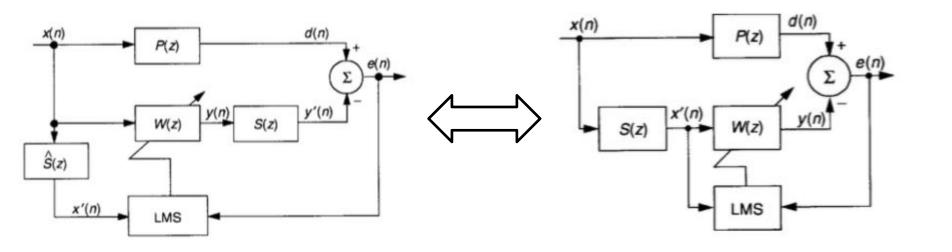


- By adding in a filter \$\hat{S}(z)\$ equal to \$S(z)\$, we can account for the secondary path signal.
- Ŝ(z) is learned by playing a known excitation signal through the LMS block diagram shown previously.



#### **FxLMS Block Diagram**

$$e(n) = d(n) - \mathbf{w}^{T}(n)\mathbf{x}(n)$$
$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu \mathbf{x}'(n)e(n)$$

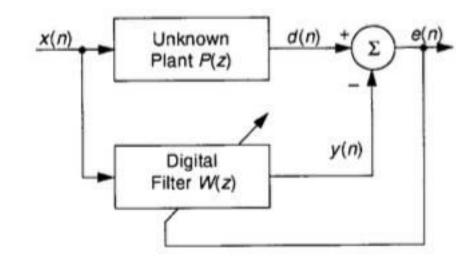


# 4 Experimental Results



#### **Experimental Setup**

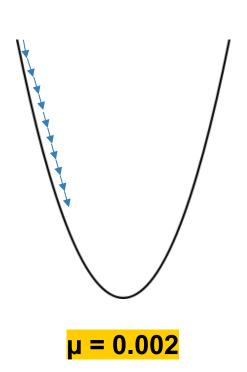
- Unknown plant P(z) is modeled by a digital LPF.
- Input x(n) is pink noise.
- Stays in digital domain so there is no secondary path.

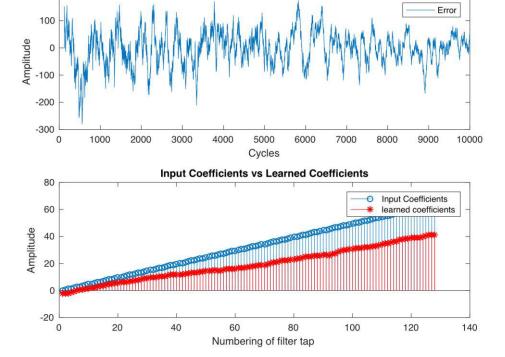




#### **Learning Rate Too Small**

200

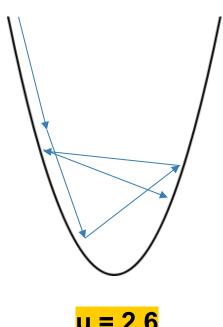


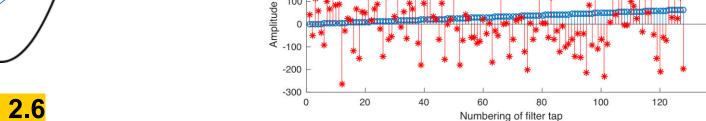


**Convergence Time in Cycles** 

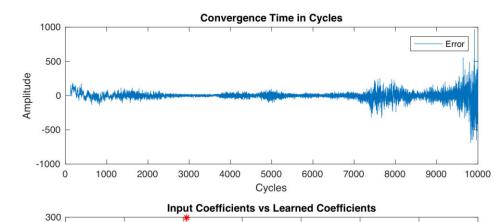


#### Learning Rate Too Large





200



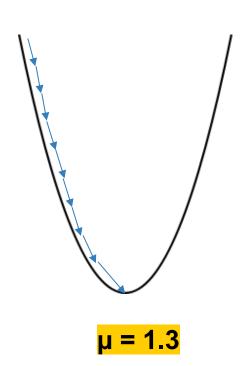
Input Coefficients

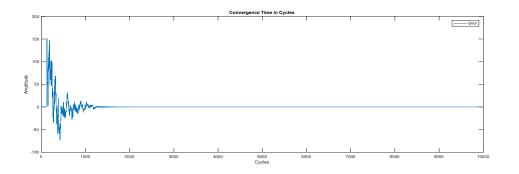
learned coefficients

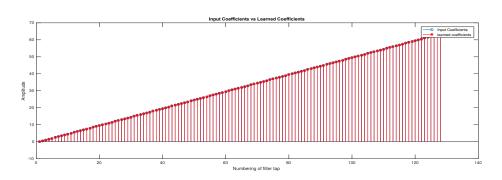
140



#### Learning Rate Juust Right







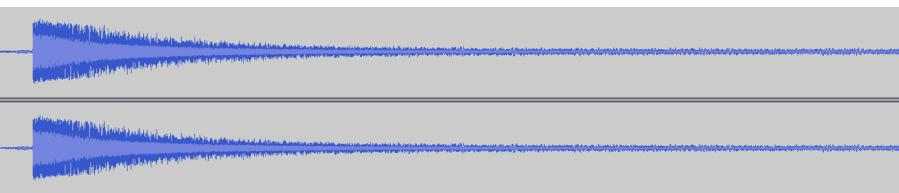
Demo



#### Yes, it's really time for the:







#### **Not Perfect Attenuation**

But a lot better than a full pink noise waveform.



# Thanks!

## Any questions?

#### Find me at:

markostam@gmail.com

#### Find my code at:

 https://github.com/markostam/active-noisecancellation