ECE 3640 - Discrete-Time Signals and Systems
Adaptive Noise Cancellation

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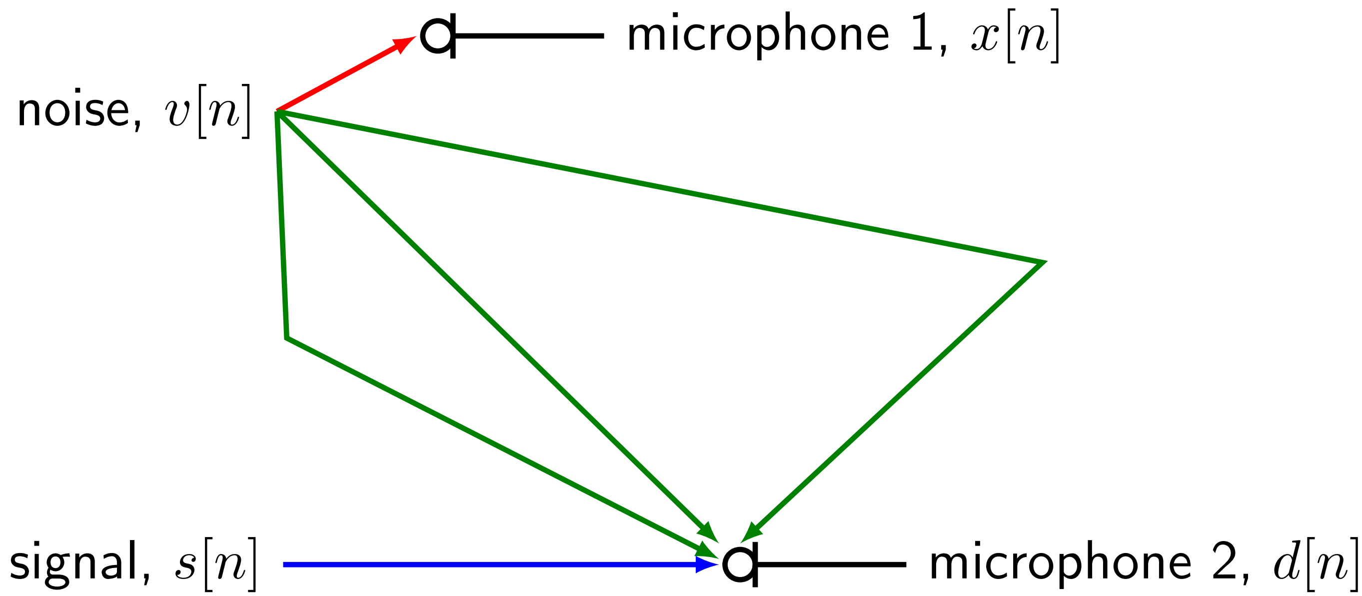
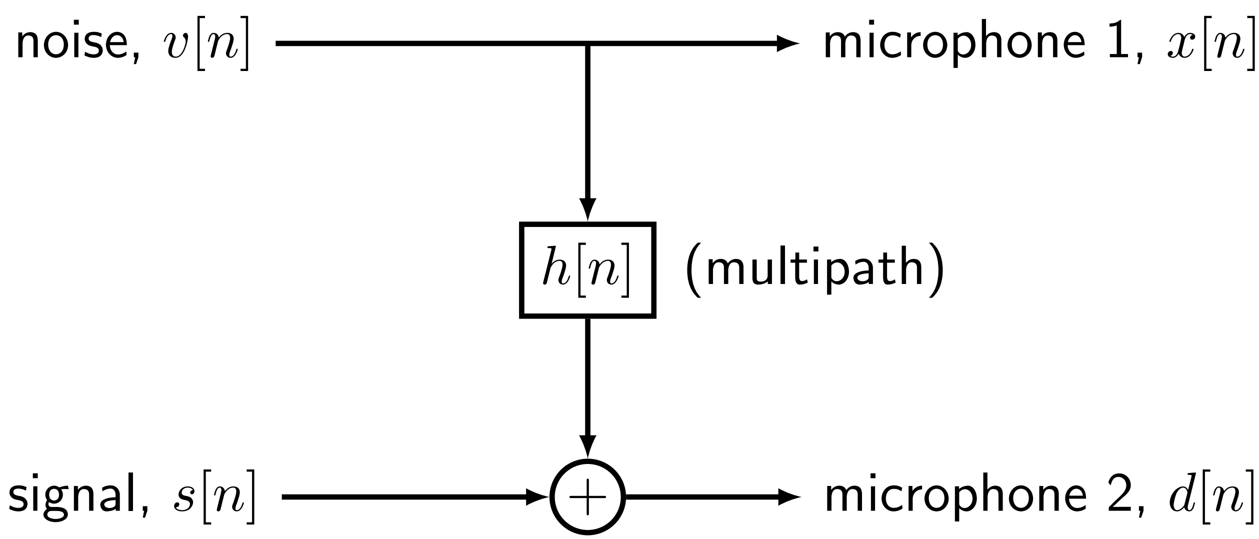
physical system



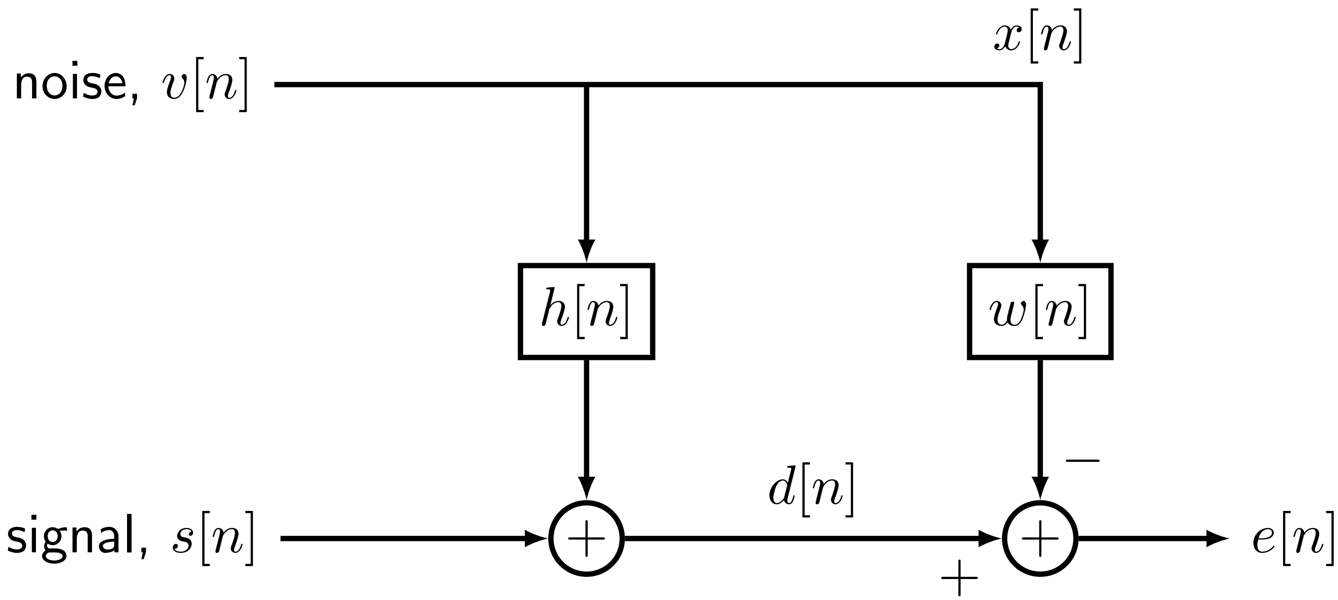
- problem: we want to measure the signal $s[n]$
- constraint: we can not place a microphone near $s[n]$
- reality: we place a microphone (microphone 2) far from $s[n]$
- reality: there is background noise $v[n]$
- reality: microphone 2 picks up the signal plus noise
- idea: we can place a microphone (microphone 1) close to the noise source and then subtract from microphone 2 to cancel the noise
- reality: this won't work because the noise has been filtered by the time it reaches microphone 2
- idea: we can filter the noise before subtracting it (adaptive filter)

signal processing model



- $h[n]$ models the unknown system between the noise source and microphone 2
- we don't know $h[n]$
- we don't know $v[n]$
- we don't know $s[n]$
- recovering $s[n]$ from microphone 1 $x[n]$ and microphone 2 $d[n]$ seems impossible

noise cancellation system



- if $w[n] = h[n]$, then $e[n] = s[n]$
- we can check this out as follows

$$e[n] = s[n] + (h[n] - w[n]) * v[n]$$

adaptive filtering

- let $w_n[k]$ be the kth filter "tap" at time n
- given one new sample of $x[n]$ and $d[n]$ do the following
  1. compute adaptive filter output: $y[n] = \sum_{k=0}^{N-1} w_n[k]x[n-k]$
  2. compute the error: $e[n] = d[n] - y[n]$
  3. update the filter: $w_{n+1}[k] = w_n[k] + \mu e[n]x[n-k], \;\; k = 0, 1, \cdots, N-1$
- μ is called the step size
- this algorithm can be derived by minimizing the squared error

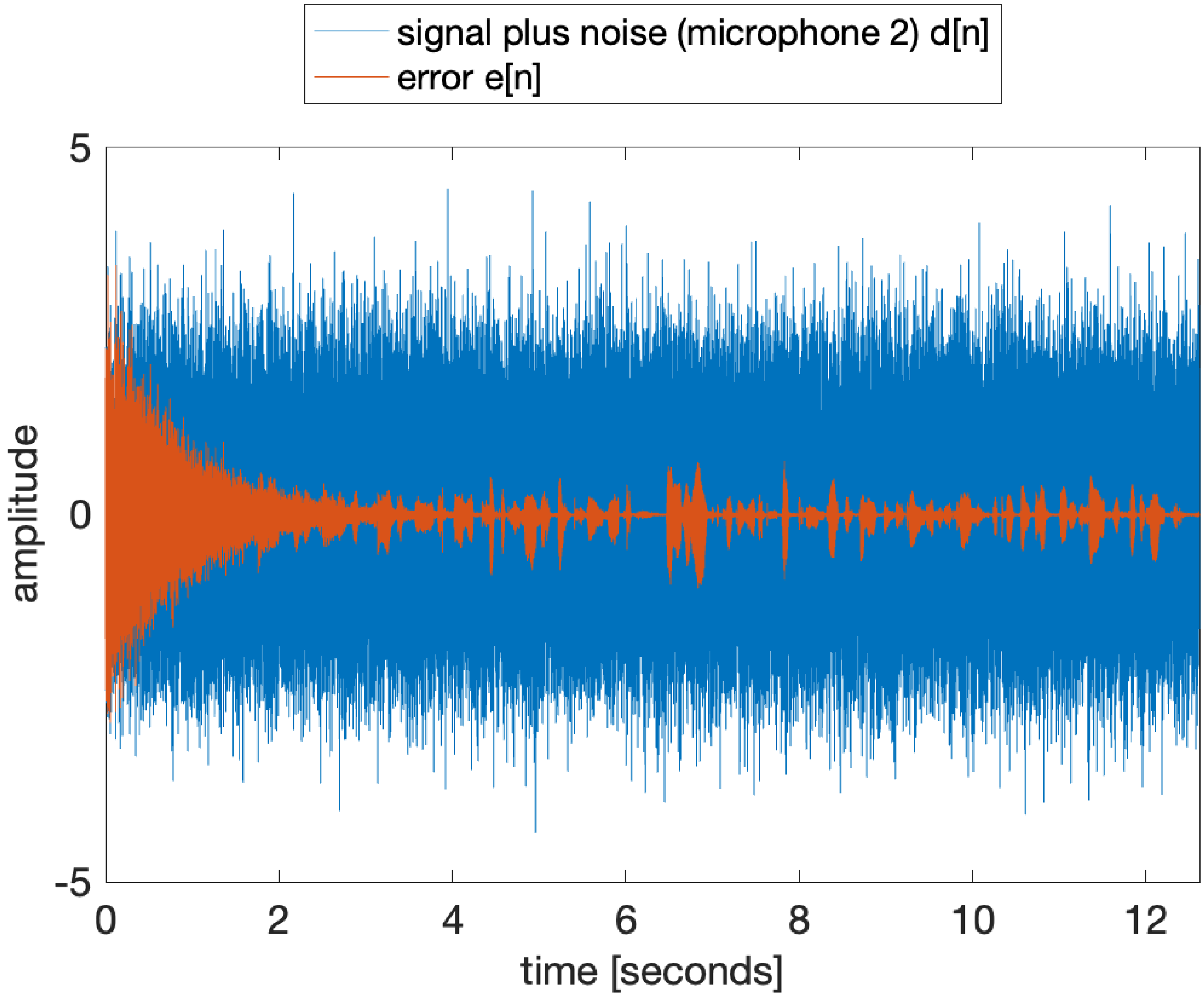
$$(e[n])^2 = \left(d[n] - \sum_{k=0}^{N-1} w_n[k]x[n-k] \right)^2$$

$$\frac{\partial (e[n])^2}{\partial w_n[k]} = -2e[n]x[n-k]$$

- now do gradient descent

$$w_{n+1}[k] = w_n[k] - \frac{\mu}{2}\frac{\partial (e[n])^2}{\partial w_n[k]} = w_n[k] - \mu e[n]x[n-k]$$

it works



signal plus noise (microphone 2) d[n]
error e[n]
5
0
-5
amplitude
0
2
4
6
8
10
12
time [seconds]

requirements

program 1: wave file → C → speaker and wave file

1. download `mic1.wav` and `mic2.wav`
2. listen to these two noise files (can you hear any speech in `mic2.wav`?)
3. write a C program to perform perform adaptive noise cancellation and recover the speech signal
   (a) experiment with the length of the adaptive filter $w[n]$
   (b) experiment with the value of step size μ

(more to come ...)