

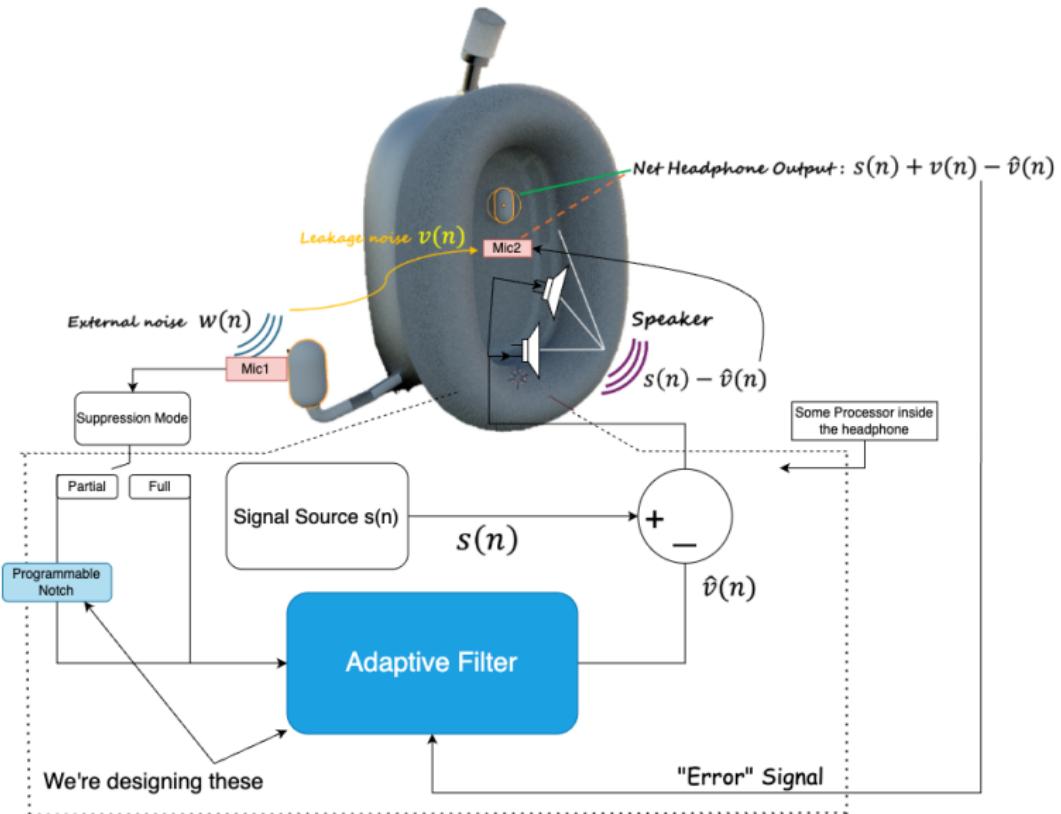
# Course Project

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# Block Diagram of The Design



# Design Choices and Justification

- We have used **RLS(Recursive Least Squares)** to design the adaptive filter shown in the block diagram.
- In the method of least squares, a deterministic sum of exponentially weighted squared errors is minimised

$$\zeta(n) = \sum_{k=1}^n \rho_n(k) e_n^2(k) \quad (1)$$

- $e_n(k)$  Error computed using current weights at past time  $k$
- $\rho_n(k)$ : Here  $\rho_n(k) = \lambda^{n-k}$ , where  $\lambda \in (0, 1]$  is the forgetting factor

## WHY RLS

- The  $\zeta$  here is a better estimate of MSE than per-sample estimates used by LMS which gives more SNR gain.
- RLS uses  $3N^2 + 11N + 8$  multiplications and  $3N^2 + 7N + 4$  additions whereas  $2N + 3$  multiplications and  $2N + 5$  additions in LMS.

# Our Design

**For full suppression**, we give the RLS filter the reference noise signal from the external microphone, and the noisy speech signal from the headphones.

**For partial suppression**, we remove the tones from the reference noise signal, and give it to RLS filter. The tones are removed using a notch filter.

- Filter size, forgetting factor affect the performance of the RLS filter, and the bandwidth of the notch filter affects its performance.
  - **Forgetting factor ( $\lambda$ )**: We chose  $\lambda = 0.99$ .
  - **Filter Size ( $N$ )**: We chose  $N = 8$ .
  - **Bandwidth of Notch Filter**: We chose a bandwidth of 20 Hz.

## Assumptions:

- The external noise is uncorrelated with the clean signal.
- The external noise provides a good estimate of the leakage noise reaching the ears.

## Performance Metrics:

- **SNR Gain**: We measure the pre and post cancellation SNR, and measure the difference, to quantify the performance of the filter.
- **NRNTN**: We measure the Non tonal noise in Noisy speech, and the residual non tonal noise in the output signal, and compare them.

$$\gamma = \frac{\|e(n) - P_{tonal} - P_{clean}\|^2}{\|d(n) - P_{tonal} - P_{clean}\|^2}$$

# Time Domain

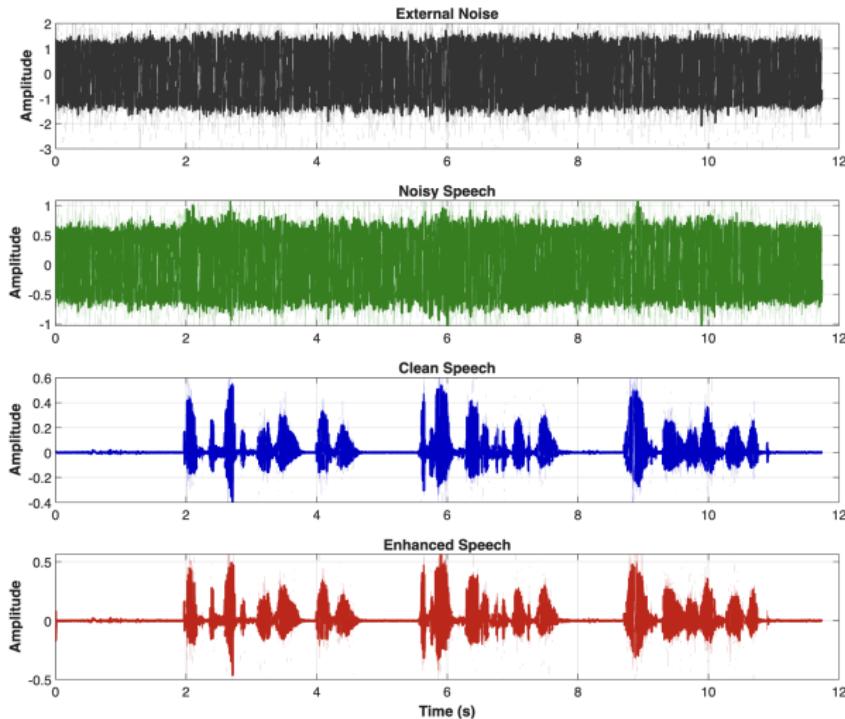


Figura 1: Time Domain Signals

# Spectrograms

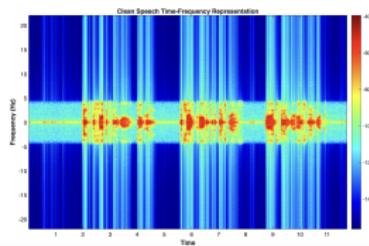


Figura 2: Clean Speech

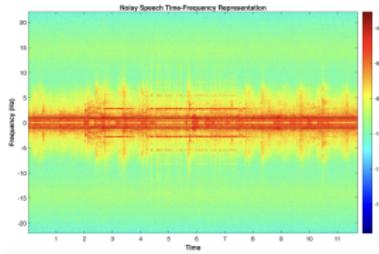


Figura 3: Clean Speech

# More Spectrograms

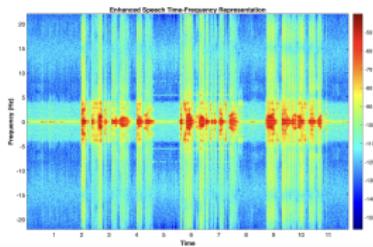


Figura 4: Result with  $\lambda = 0.99$

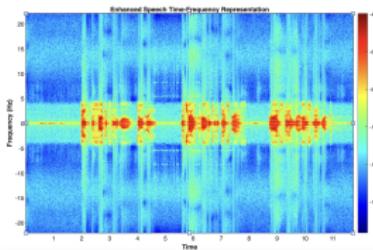


Figura 5: Result with  $\lambda = 0.9999$

# Comparison: NLMS vs RLS

## NLMS (Normalized LMS)

- **Cost Function:**

$$J(n) = e^2(n)$$

- **Error:**

$$e(i) = d(i) - \mathbf{w}^T(i)\mathbf{x}(i)$$

- **Complexity:**

- $2N + 3$  Multiplications
- $2N + 4$  Additions
- 1 Division

- **Parameter:** Step size  $\mu$

- **Accuracy:** Ensemble average behavior

- **Memory:** Low (only vectors)

## RLS (Recursive Least Squares)

- **Cost Function:**

$$J(n) = \sum_{k=0}^n \lambda^{n-k} e^2(k)$$

- **Error:**  $e(k) = d(k) - \mathbf{w}_{n-1}^T \mathbf{x}(k)$

- **Complexity:**

- $3N^2 + 11N + 8$  Multiplications
- $3N^2 + 7N + 4$  Additions
- 1 Division

- **Parameter:** Forgetting factor  $\lambda$

- **Accuracy:** Optimized for specific signal, hence more accurate

- **Memory:** High (stores matrices)

# References and Design Influence I

## References

- Farhang-Boroujeny, B. (2013). *Adaptive Filters: Theory and Applications* (2nd ed.). Wiley.
- Chapter 12 Method Of Least Squares : The Algorithm was taken from this book and all the derivations are influenced by this book.

# RLS DERIVATIONS

We aim to minimize the exponentially weighted least-squares error:

$$\zeta(n) = \sum_{k=1}^n \lambda^{n-k} e_n^2(k) \quad (12.1)$$

where  $0 < \lambda \leq 1$  is the forgetting factor.

$$y_n(k) = \mathbf{w}^T(n) \mathbf{x}(k) \quad (12.2)$$

$$e_n(k) = d(k) - y_n(k) \quad (12.3)$$

We define the filter output and error at time  $k$  using weights at time  $n$ .

# Vector Notation

$$\mathbf{d}(n) = [d(1) \ d(2) \ \dots \ d(n)]^T \quad (12.4)$$

$$\mathbf{y}(n) = [y_n(1) \ y_n(2) \ \dots \ y_n(n)]^T \quad (12.5)$$

$$\mathbf{e}(n) = [e_n(1) \ e_n(2) \ \dots \ e_n(n)]^T \quad (12.6)$$

$$\mathbf{X}(n) = [\mathbf{x}(1) \ \mathbf{x}(2) \ \dots \ \mathbf{x}(n)] \quad (12.7)$$

These help simplify the matrix-based formulation.

# Matrix Form of Output and Error

$$\mathbf{y}(n) = \mathbf{X}^T(n)\mathbf{w}(n) \quad (12.8)$$

$$\mathbf{e}(n) = \mathbf{d}(n) - \mathbf{y}(n) \quad (12.9)$$

$$\zeta(n) = \mathbf{e}^T(n)\mathbf{e}(n) \quad (12.10)$$

Substitute expressions into the cost function:

The cost function becomes a quadratic in  $\mathbf{w}(n)$ .

$$\zeta(n) = \mathbf{d}^T(n)\mathbf{d}(n) - 2\theta^T(n)\mathbf{w}(n) + \mathbf{w}^T(n)\Phi(n)\mathbf{w}(n) \quad (12.11)$$

Where:

$$\Phi(n) = \mathbf{X}(n)\mathbf{X}^T(n) \quad (12.12)$$

$$\theta(n) = \mathbf{X}(n)\mathbf{d}(n) \quad (12.13)$$

# Optimal Weights from Normal Equation

Set derivative of  $\zeta(n)$  w.r.t.  $\mathbf{w}(n)$  to zero:

$$\Phi(n)\hat{\mathbf{w}}(n) = \theta(n) \quad (12.14)$$

$$\hat{\mathbf{w}}(n) = \Phi^{-1}(n)\theta(n) \quad (12.15)$$

Introduce forgetting factor:

$$\rho_n(k) = \lambda^{n-k}, \quad 0 < \lambda \leq 1 \quad (12.33)$$

Then define:

$$\Lambda(n) = \text{diag}(\lambda^{n-1}, \lambda^{n-2}, \dots, 1) \quad (12.35)$$

# Exponentially Weighted Normal Equation

$$\Phi_\lambda(n) = \mathbf{X}(n)\Lambda(n)\mathbf{X}^T(n) \quad (12.37)$$

$$\theta_\lambda(n) = \mathbf{X}(n)\Lambda(n)\mathbf{d}(n) \quad (12.38)$$

$$\hat{\mathbf{w}}(n) = \Phi_\lambda^{-1}(n)\theta_\lambda(n) \quad (12.36)$$

Recursive updates:

$$\Phi_\lambda(n) = \lambda\Phi_\lambda(n-1) + \mathbf{x}(n)\mathbf{x}^T(n) \quad (12.41)$$

$$\theta_\lambda(n) = \lambda\theta_\lambda(n-1) + \mathbf{x}(n)d(n) \quad (12.42)$$

To recursively compute inverse we use matrix inversion lemma :

$$(\mathbf{A} + \alpha\mathbf{a}\mathbf{a}^T)^{-1} = \mathbf{A}^{-1} - \frac{\alpha\mathbf{A}^{-1}\mathbf{a}\mathbf{a}^T\mathbf{A}^{-1}}{1 + \alpha\mathbf{a}^T\mathbf{A}^{-1}\mathbf{a}} \quad (12.43)$$

# Gain Vector and Weight Update

Using the matrix inversion lemma, the inverse of the autocorrelation matrix is updated as:

$$\mathbf{P}(n) = \lambda^{-1} \mathbf{P}(n-1) - \frac{\lambda^{-2} \mathbf{P}(n-1) \mathbf{x}(n) \mathbf{x}^T(n) \mathbf{P}(n-1)}{1 + \lambda^{-1} \mathbf{x}^T(n) \mathbf{P}(n-1) \mathbf{x}(n)} \quad (12.44)$$

Define gain vector:

$$\mathbf{k}(n) = \frac{\lambda^{-1} \mathbf{P}(n-1) \mathbf{x}(n)}{1 + \lambda^{-1} \mathbf{x}^T(n) \mathbf{P}(n-1) \mathbf{x}(n)} \quad (12.45)$$

Substitute the gain vector to simplify the inverse update:

$$\mathbf{P}(n) = \lambda^{-1} [\mathbf{P}(n-1) - \mathbf{k}(n) \mathbf{x}^T(n) \mathbf{P}(n-1)] \quad (12.46)$$

# Rewriting the Gain Vector

The gain vector can also be expressed in terms of current inverse:

$$\mathbf{k}(n) = \lambda^{-1} [\mathbf{P}(n-1) - \mathbf{k}(n)\mathbf{x}^T(n)\mathbf{P}(n-1)] \mathbf{x}(n) \quad (12.47)$$

Another simplification gives:

$$\mathbf{k}(n) = \mathbf{P}(n)\mathbf{x}(n) \quad (12.48)$$

This is sometimes used directly in the RLS weight update.

From recursive forms of  $\theta_\lambda(n)$  and  $\mathbf{P}(n)$ :

$$\hat{\mathbf{w}}(n) = \lambda \mathbf{P}(n) \theta_\lambda(n-1) + \mathbf{k}(n)d(n) \quad (12.49)$$

# Final Recursive Weight Update Form

Rewriting with previous weight vector:

$$\hat{\mathbf{w}}(n) = \hat{\mathbf{w}}(n-1) + \mathbf{k}(n) [d(n) - \hat{\mathbf{w}}^T(n-1)\mathbf{x}(n)] \quad (12.50)$$

This is the core RLS weight update formula. Simplifying further

$$\hat{\mathbf{w}}(n) = \hat{\mathbf{w}}(n-1) + \mathbf{k}(n)e_{n-1}(n) \quad (12.52)$$

# Initialization of the RLS Algorithm

Before starting the RLS iterations, we need to initialize the key variables:

- **Initial inverse correlation matrix:**

$$\mathbf{P}(0) = \delta^{-1} \mathbf{I} \quad (12.55)$$

where  $\delta$  is a small positive constant (e.g.,  $10^{-2}$ ).

- **Initial weight vector:**

$$\hat{\mathbf{w}}(0) = \mathbf{0} \quad (12.57)$$

These settings ensure numerical stability and allow recursive updates to begin.

# Second-Order IIR Notch Filter

A **notch filter** removes narrowband tonal interference.

For normalized notch frequency  $w_0$  and bandwidth  $bw$ , the pole radius:

$$r = 1 - \frac{bw}{2}$$

The transfer function of a second-order IIR notch filter is:

$$H(z) = \frac{1 - 2\cos(\pi w_0)z^{-1} + z^{-2}}{1 - 2r\cos(\pi w_0)z^{-1} + r^2z^{-2}}$$

From  $H(z) = \frac{B(z)}{A(z)}$ , we get the recursive form:

$$y[n] = b_0x[n] + b_1x[n-1] + b_2x[n-2] - a_1y[n-1] - a_2y[n-2]$$

with:

$$b_0 = 1, \quad b_1 = -2\cos(\pi w_0), \quad b_2 = 1$$

$$a_1 = -2r\cos(\pi w_0), \quad a_2 = r^2$$