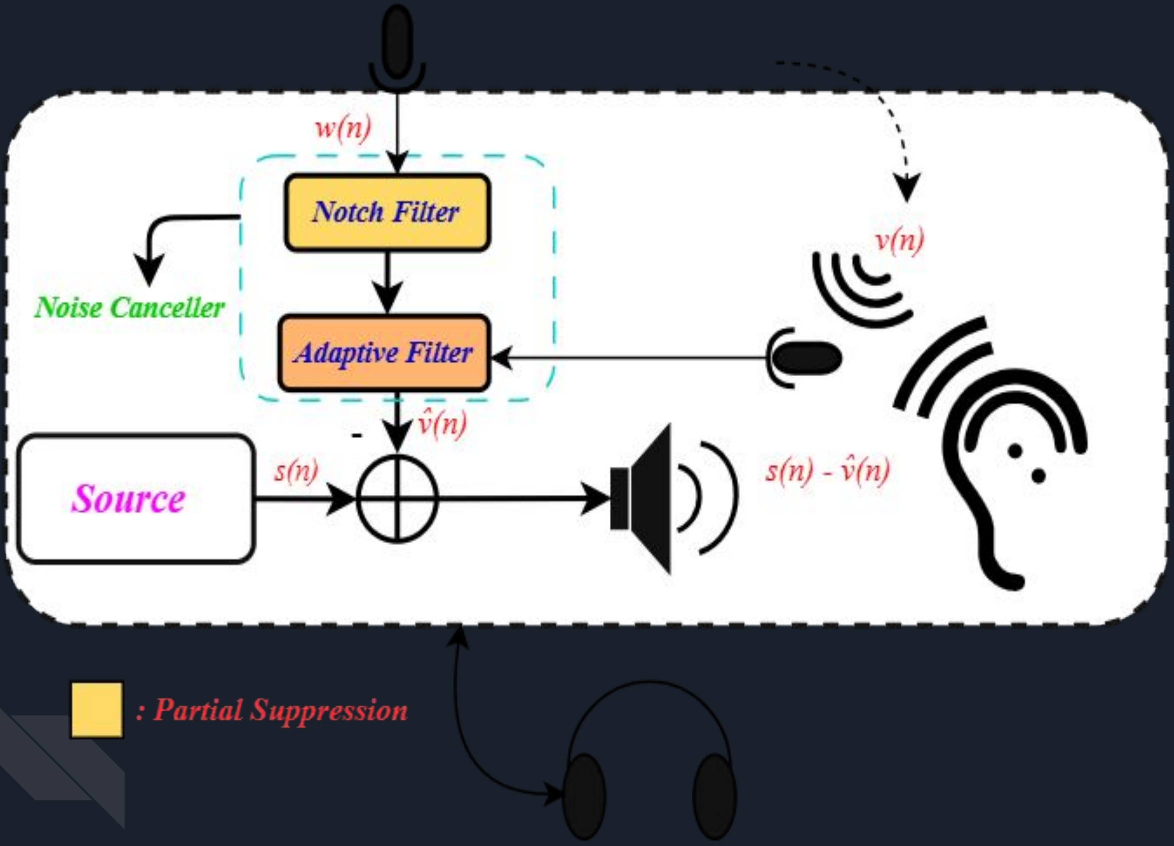
EE2800 - DIGITAL SIGNAL PROCESSING

EE23BTECH11011,EE23BTECH11012,
EE23BTECH11013,EE23BTECH11014

Problem Statement : Noise Cancellation



The project aims to design a programmable noise canceller which estimates the leakage noise using the w(n) captured by outer microphone. The design includes two modes :

➢ **Full Suppression**

Cancels all the noise components i.e., by maximizing the SNR (v(n) ≈ v^(n)).

➢ **Partial Suppression**

Retains the programmable tonal frequencies and suppress the remaining non tonal components.

Design Choices & Justifications

- **Adaptive Filter (RLS):**
  - *RLS was chosen because it was providing better results than LMS & NLMS in terms of SNR, suppressing the noise effectively.(Slide 6: Plots)*
  - *RLS uses the actual data for the weight updates, minimizes exponentially weighted cost function including the past samples and adjusts the step size using the autocorrelation matrix resulting in faster convergence and high SNR.*
  - *Filter Length(L) determines the buffer length of the filter, Forgetting factor(λ) controls how quickly past data has to neglected and Delta(δ) helps in initializing the filter and the convergence speed.*
  - *Filter length affects the performance and computational cost. Higher λ won't work in rapidly changing environments as they adapt slowly. Lower δ will take more iterations to converge and higher value might not converge at all.*

- **Notch Filter:** (Refer to *Slide 8: Notch Filter Design*)
  - *Notch filter was chosen to suppress the programmable tonal frequencies from the external noise captured by the mic.*
  - *To achieve narrow bandwidth poles were placed close to the zeros which are on the unit circle. But by making pole radius too close to the unit circle i.e., r ≈ 1 can cause the system to become unstable.*

Design Details

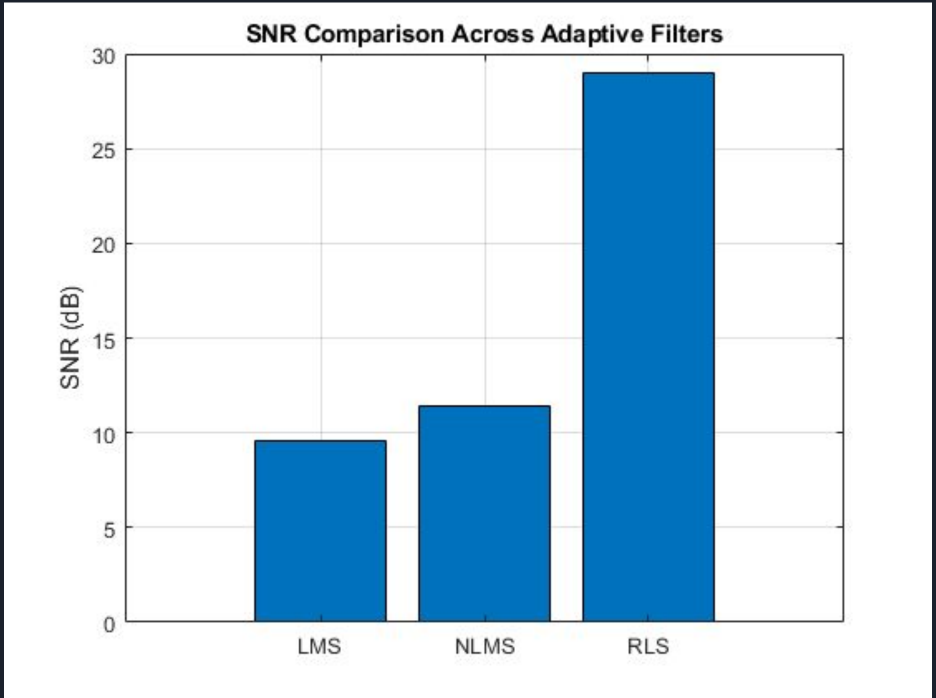
- **Full Suppression:**
  - *In this mode the notch filter block is off. The external noise captured by the mic is passed to the adaptive filter which estimates the leakage noise entering the headphone . Internal mic present in the headphone captures the output produced by the speaker(s(n) - v^(n)) and the leakage noise which is passed to the adaptive filter as feedback to update the weights, eventually the algorithm converges, resulting in v(n) ≈ v^(n) receiver hearing s(n). (Refer to* *Slide 7* *for update equations)*
  - *The parameters L = 8 , λ = 0.9999, δ= 0.001 were determined through iterative testing of the data. These values produced the best results.*
- **Partial Suppression:**
  - *In this mode, we need the output to be s(n) + tonal frequency so the adaptive filter must estimate the non tonal noise so that upon subtracting from s(n) + v(n) we get desired output.*
  - *A notch filter must be placed before the adaptive filter so that, notch suppresses the tonal frequency and passes the remaining signal to the adaptive filter and estimates only the non tonal.*
  - *The parameters r = 0.993 results in a narrow bandwidth of 98 Hz.*
- **Pros:**
  - *Effective noise reduction resulting in high Signal-to-Noise Ratio (SNR).*
  - *The use of cascaded notch filters ensures that desired tonal components are retained.*
  - *The design supports real-time processing of the data.*
- **Cons:**
  - *The pole radius r is fixed, which may not produce the optimal bandwidth for all frequencies.*
  - *For storage of matrix P we need large memory for higher L.*

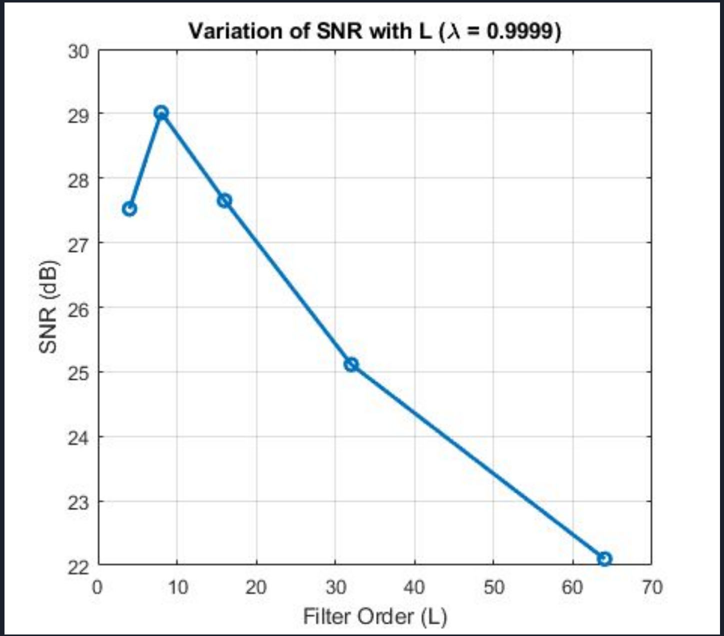
References

- *M.H. Hayes, "Statistical Digital Signal Processing and Modeling", John Wiley & Sons, 1996. Chapter 9 (9.2.2-9.4).*
- *S. Haykin, Adaptive Filter Theory, Fourth Edition, Pearson Education LPE, 2007. Chapter 9.*

Plots

Full Suppression:





LMS /NLMS : L = 4, μ = 0.005 & RLS : L = 8, λ = 0.9999

Recursive Least Squares Algorithm

The algorithm is initialized with : $\hat{w}(0) = 0, \quad P(0) = \delta^{-1} I$

The algorithm is updated recursively as :

$$e(n) = s(n) + v(n) - \hat{w}^T(n-1)x(n)$$

$$z(n) = P(n-1)x(n)$$

$$g(n) = \frac{z(n)}{\lambda + x^T(n)z(n)}$$

$$P(n) = \lambda^{-1}P(n-1) - \lambda^{-1}g(n)x^T(n)P(n-1)$$

$$\hat{w}(n) = \hat{w}(n-1) + g(n)e(n)$$

Notch Filter Design

The Second Order Notch Filter is used in the design as :

$$H(z) = \frac{1 - 2\cos(\omega_0)z^{-1} + z^{-2}}{1 - 2r\cos(\omega_0)z^{-1} + r^2 z^{-2}}$$

From the difference equation we get the filter coefficients as :

$$\text{Numerator coefficients (b):} \quad b_0 = 1, \quad b_1 = -2\cos(\omega_0), \quad b_2 = 1$$

$$\text{Denominator coefficients (a):} \quad a_0 = 1, \quad a_1 = -2r\cos(\omega_0), \quad a_2 = r^2$$

The relation between the bandwidth and the pole radius :

$$B.W = \frac{(1-r)f_s}{\pi}$$

Thank You