EE2800- Course Project

Team 16

Block Diagram

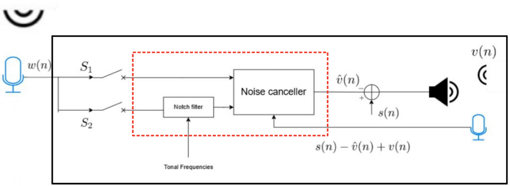


Figure: Block Diagram of the Noise Canceller

Here, the black rectangle represents the headphone cup. $w(n)$ represents the direct external noise measurement. $s(n)$ is the speech signal which the user desires to hear. $v(n)$ is the leakage noise heard by the user, which is different from $w(n)$ because this sound travels through the headphone cup.

The noise canceller, is the block which is being built, is shown by the red dotted rectangle. $\hat{v}(n)$ is the leakage noise estimated by this block.

The switches S_1 and S_2 are complementary switches. S_1 is ON when Full Suppression mode is on, and OFF when Partial Suppression mode is on.

Design Choices and Justifications

For the RLS algorithm, the following design parameters have been used.

- A total number of 12 taps have been used (filter order 11).
- The forgetting factor (λ) is chosen to be 0.9995. This value of λ forgets errors which are older than around 14000 samples, and will track the changes in the signal better.
- δ is taken to be 10^{-4}. This is taken because we are not very confident about the correlations between different dimensions initially.

For the notch filter, r is taken to be 0.999, which gives a bandwidth of the notch of about 14 Hz (analog) around each tonal frequency.

Metric to measure non-tonal suppression: The clean speech, the noisy speech and the filtered speech are notched at the given tonal frequencies. The SNR for these signals is now compared. Since, there are no tonal parts in these signals, this method quantifies the non-tonal suppression performed by the Noise Canceller.

The SNR gain observed after Full Suppression is around 39 dB and around 15 dB after Partial Suppression, for the given signals. The tonal frequency is taken to be 1 kHz for the Partial Suppression mode.

Design- Pros and Cons

- Full Suppression
  - Approach: The leakage noise, comprising both tonal and non-tonal components, is estimated from the external noise. This estimate of the leakage noise signal is pre-subtracted from the speech signal and is fed to the speaker.
  - The RLS algorithm is used over other algorithms like NLMS because it has given better SNR gains (around 36 dB!), though it is computationally expensive than NLMS. NLMS gave gains of around 20 dB.
  - The parameters have to be carefully chosen, or else the performance might not be up to our expectation for RLS.
- Partial Suppression
  - When the Partial Suppression mode is ON, the external noise is passed through a *notch filter*, and then the leakage noise is estimated. Since, the tonal frequencies are not present in the input to the noise canceller, these frequencies cannot be suppressed and are preserved.
  - Since there is a given bandwidth for the notch filter, this would also preserve those tonal components which are within this bandwidth, which might be undesirable.

References

- Digital Signal Processing: Proakis, Monolakis, 4/e, Chapter 5.4- Notch Filters.
- Adaptive Filter Theory: Fourth Edition, Simon Haykin, Chapter 9 - 9.3, 9.4- RLS Algorithm.
- Stastical Digital Signal Processing and Modelling: M.H Hayes, Chapter 9- 9.4- RLS Algorithm.

The RLS Algorithm

The RLS algorithm minimizes the following cost function:

$$J(n) = \sum_{k=1}^{n} \lambda^{n-k} e^2(k)$$

Where $e(\cdot)$ is the error signal $(d(n) - v(\hat{n}) = s(n) + v(n) - \hat{v}(n)$ in this case). λ is the forgetting factor, which exponentially gives lesser weightage to older samples. Here, if $\mathbf{W}$(n) is the weight vector of the noise canceller, and $\mathbf{x}(n) = [w(n)\ w(n-1)\ \ldots w(n-N+1)]^\top$, is the vector containing N recent samples of the external noise signal, then $\hat{v}(n) = \mathbf{W}(n)^\top \mathbf{x}(n)$ is the estimate of the leakage noise by the Noise Canceller.
Minimizing the cost function, we get the following update equations.

$$\begin{aligned}
\alpha(n) &= d(n) - \mathbf{W}(n-1)^\top \mathbf{x}(n) \\
\mathbf{g}(n) &= (\lambda + \mathbf{x}(n)^\top P(n-1)\mathbf{x}(n))^{-1} P(n-1)\mathbf{x}(n) \\
P(n) &= \lambda^{-1}(P(n-1) - \mathbf{g}(n)\mathbf{x}(n)^\top P(n-1)) \\
\mathbf{W}(n) &= \mathbf{W}(n-1) + \alpha(n)\mathbf{g}(n)
\end{aligned}$$

Here, $P(n)$ is the inverse autocorrelation matrix $\mathbf{g}(n)$ represents the gain vector, and $\alpha(n)$ represents the error between the current signal and the estimate using

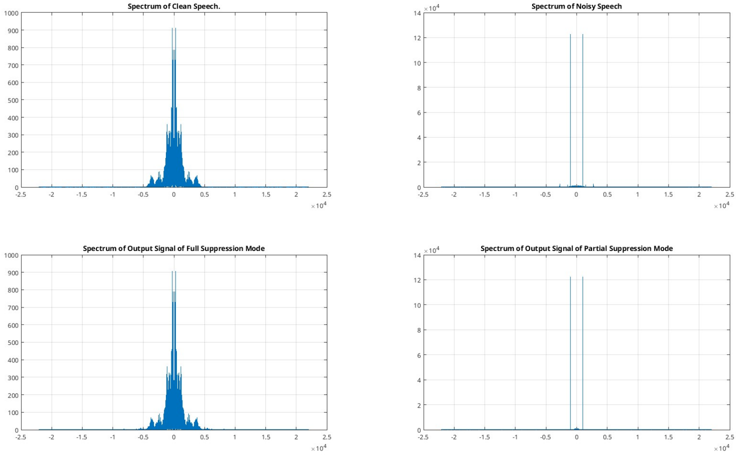
The Notch Filter

A notch filter is a filter that has a deep "notch" at a given frequency. Its gain is very small or approximately zero at this notch frequency. Its transfer function is given by:

$$H(z) = \frac{(1 - e^{j\omega_b} z^{-1})(1 - e^{-j\omega_b} z^{-1})}{(1 - re^{j\omega_b} z^{-1})(1 - re^{-j\omega_b} z^{-1})} = \frac{1 - 2\cos\omega_b z^{-1} + z^{-2}}{1 - 2r\cos\omega_b z^{-1} + r^2 z^{-2}}$$

This has zeros at $z = e^{\pm\omega_b}$, and poles at $z = re^{\pm\omega_b}$, where r is a constant approximately, but not exactly equal to 1. The presence of these poles ensures that the gain of the filter is nearly one for $\omega \neq w_b$ on the unit circle.
Its 3 dB bandwidth is given by: $\Delta\omega \approx 2(1 - r)$ rad / sample or $\Delta F = \frac{(1-r)F_s}{\pi}$ Hz (analog).
The parameter r: Taking r to be very lesser than 1 (around 0.9), the notch becomes less steeper. If r is taken to be very close to 1 (around 0.99999 or higher), there might be numerical issues. So, we choose r to be around 0.999.

Plots



As visible, the tonal frequencies are preserved in the Partial Suppression mode of the Noise Canceller.