

Wiener Filter

1 Questions

1. What are the design criteria for the Wiener Filter?
2. What assumptions are made for the basic solution of the Wiener Filter?
3. What is the basic principle when deriving the solution of the Wiener Filter by the principle of orthogonality?
4. What problems occur when using the time domain solution?
5. How are these problems solved by using the frequency domain solution?
6. Look at slide II-19. In the equation “Approximation using short-time estimations”, you find the maximum operator. Why do you have to use it at this place? **Hint:** See slide II-26.
7. How is the power spectrum of the disturbed signal usually estimated? What common methods exist for the estimation of the noise power spectrum? How do the methods differ?
8. What is musical noise?
9. How is it avoided?

2 Answers

1. Design of a filter that separates a desired signal optimally from additive noise.
2. Described signals are stationary random processes, knowledge of statistical properties up to second order.
3. The principle of orthogonality says that the linear estimation of the desired signal $s(n)$ with the observation signal $y(n)$ is exactly then optimal in the sense of MSE, if the error $e(n) = s(n) - \hat{s}(n)$ is orthogonal to every past entry $y(n-i)$, $\forall i \in [0, N-1]$.
4. Autocorrelation of undisturbed signal is not directly available. Equation system yields high complexity and potentially stability problems when inverting the matrix.
5. No autocorrelation/spectral density of the undisturbed signal is required. Simple division in the subband domain.
6. The maximum operator is applied because in speech pauses the **estimated** noise power spectral density can become bigger than the **estimate** of the power spectral density of the microphone signal. Thus, the filter coefficients would become negative and the signal would be amplified. To prevent this, the maximum operator is applied.
7. Magnitude squared spectrum for the disturbed signal. For noise: Derivation from disturbed estimate during speech pauses (voice activity detection required) or temporal minima tracking. The first approach is only able to update the estimate during speech pauses while the second approach is constantly in an adaptive state.
8. The short-term power of the input signal usually fluctuates faster than the noise **estimate** - especially during speech pauses. As a result the filter characteristic opens and closes in a randomized manner, which results in tonal residual noise (so-called musical noise). See slide II-23/24.
9. It is avoided by applying an overestimation factor to the noise estimate. See slide II-23.