Robust and **E**fficient **D**igital Signal Processing

Gerhard Schmidt

Digital Signal Processing and System Theory
Kiel University
Germany

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Prediction-based Filter Design

written by Gerhard Schmidt

In this chapter we will discuss how linear prediction can be used for designing filters with an arbitrary frequency response. The described design schemes can be used to implement real-time filter design applications that can work also on very simple hardware.

Contents:

1.1 Basics

In this chapter we will discuss how linear prediction can be used for designing filters with an arbitrary frequency response. Since linear predictors are used in a variety of applications (e.g. speech coding) various implementations exist, that solve the so-called normal equations in a robust and efficient manner. These schemes can be reused to implement real-time filter design applications that can work also on very simple hardware.

1.2 Application Examples

Prediction in general means to forecast signal samples that are not yet available (forward prediction) or to reestablish already forgotten samples (backward prediction). With this capability predictors play an important role in signal processing wherever it is desirable, for instance, to reduce the amount of data to be transmitted or stored. Examples for the use of predictors are encoders for speech or video signals.

However, linear prediction can also be used for several other applications:

Remark:

Before we start with the derivation of the filter design itself, the following applications should motivate the design process.

- **Loudspeaker equalization**

  To improve the playback quality of loudspeakers equalization filters might be placed before the DA converters of playback devices (see Fig. 1.1). These filters are designed such that the frequency response of the system consisting of the loudspeaker itself and the equalization filter should be close to a predefined curve.

  If more than one loudspeaker should be equalized often additional restrictions such as linear phase behaviour (constant group delay) are desired. Fig. 1.2 shows an example of such a desired frequency response together with a non-equalized loudspeaker and its equalized counterpart.

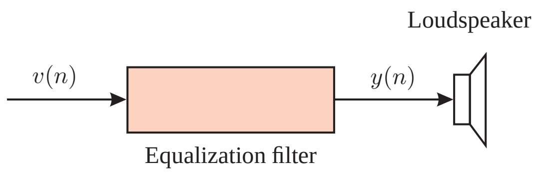

Figure 1.1: Basic structure of loudspeaker equalization schemes.

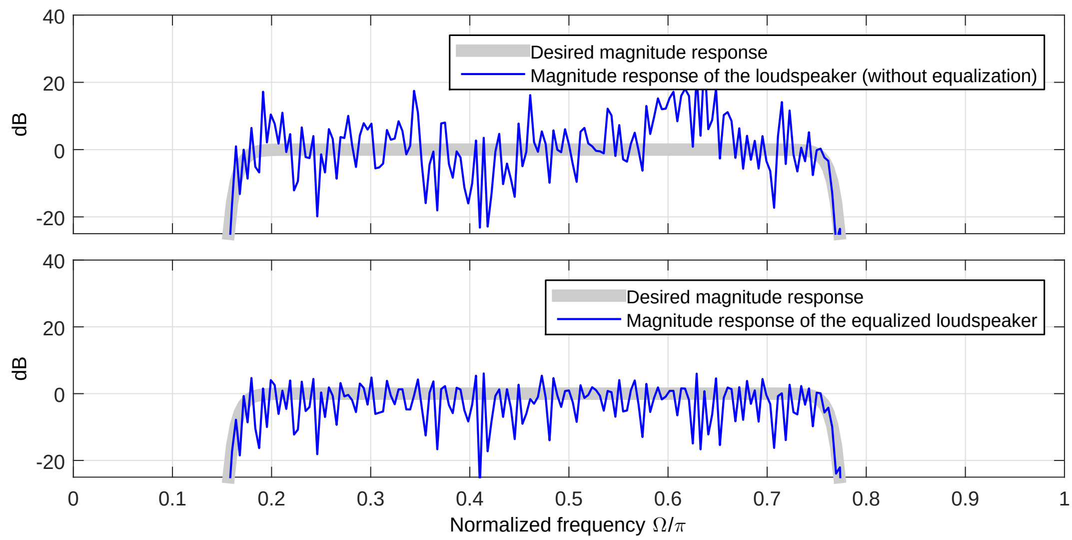

Figure 1.2: Magnitude responses of the non-equalized and the equalized loudspeaker.

- **Low-delay noise suppression**

  Whenever a desired signal is superimposed by noise signal enhancement techniques can be applied (see Fig. 1.3). Usually, statistically optimized, time-variant filters such as so-called *Wiener filters* [HS 2004] are utilized here.

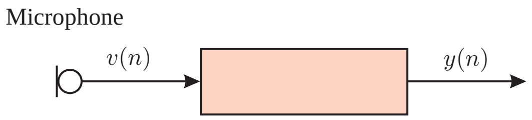

Figure 1.3: Low-delay noise suppression.

  Those approaches are usually realized in the short-term Fourier domain. However, if the delay that is inserted by the Fourier transforms is too large, time-domain approaches with low-order minimum-phase filters might be an alternative solution. The design of these filters can be prediction-based [LV 2005, LV 2006]. Fig. 1.4 shows an example of a noisy speech signal (filter input) and the corresponding noise-reduced signal (filter output).

- **Signal generation**

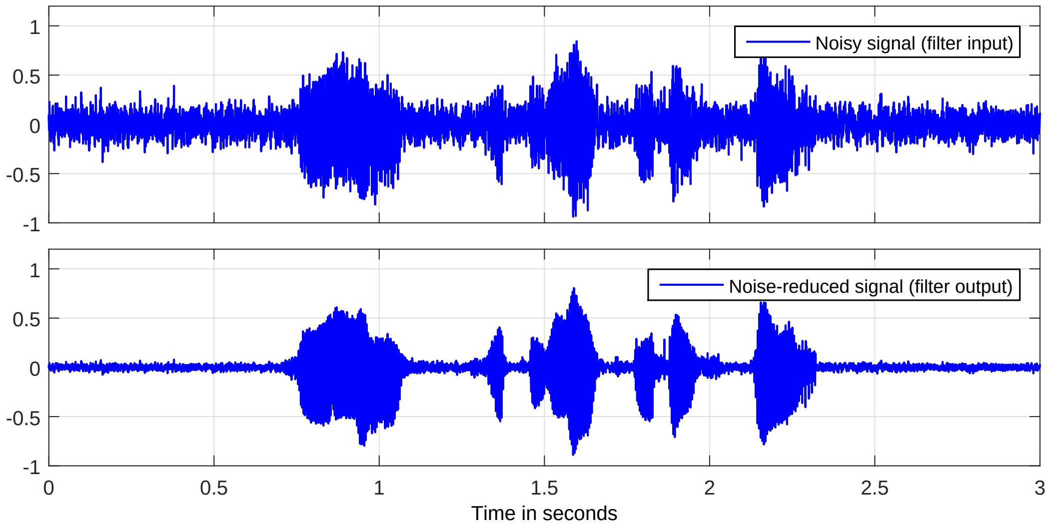

Figure 1.4: Signal before and after noise suppression.

As a last application so-called *general purpose noise or signal generators* can be mentioned. They are build usually by a white noise generator (either with Gaussian or unit amplitude distribution) and a succeeding shaping filter for adjusting the power spectral density (PSD) of the output filter (see Fig. 1.5).

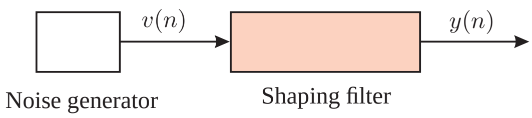

Figure 1.5: Signal generation.

Since the input PSD is constant (white noise) the shaping filter must be designed such that its frequency response (respectively the squared magnitude of it) is the same as the desired PSD. Fig. 1.6 shows an example of in input and output PSDs (in blue) together with the desired PSD (in grey).

1.3 References

[HS 2004] E. Hänsler, G. Schmidt: *Acoustic Echo and Noise Control*, Wiley, 2004.

[LV 2005] H. Löllmann, P. Vary: *Low Delay Filter for Adaptive Noise Reduction*, Proc. IWAENC '05, Eindhoven, The Neth

[LV 2006] H. Löllmann, P. Vary: *A Filter Structure for Low Delay Noise Suppression*, Proc. ITG-Fachtagung '06, Kiel, Ger

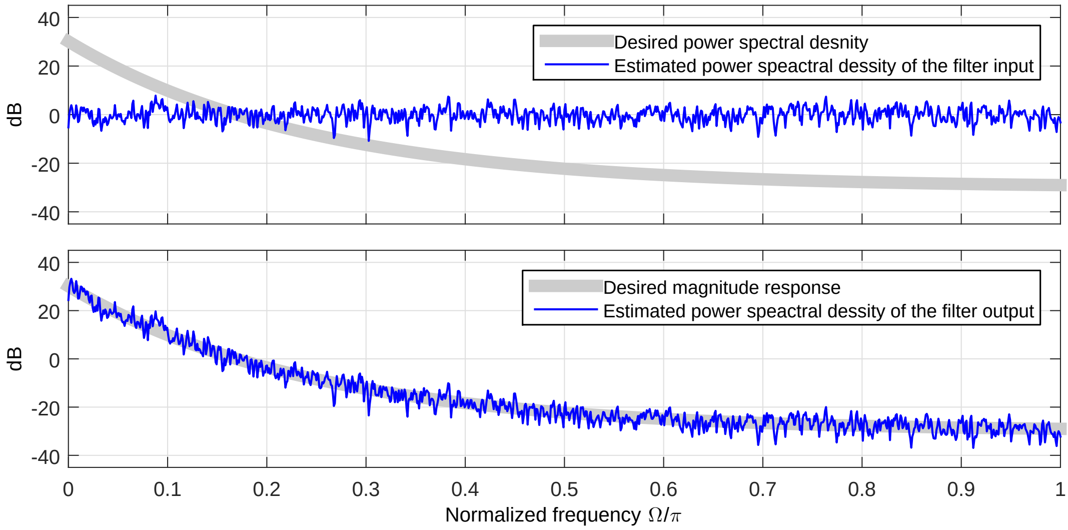

Figure 1.6: Power spectral density before and after the shaping filter.

1.4 Authors of this Chapter

Gerhard Schmidt Text GUS