

Lukas Schirmer*, Sören Lange* and Moritz Boueke, Keanu Kudicke, Henrik Wolfram, Michael Höft, Gerhard Schmidt

Predictor-based Equalization of Magnetoelectric Sensor with Integrated Coil

<https://doi.org/10.1515/cdbme-2026-0218>

Abstract: Magnetoelectric (ME) sensors provide high sensitivity at resonance, but exhibit limited usable bandwidth. This work proposes a prediction-based equalization approach to extend the usable bandwidth using broadband noise excitation and linear prediction. A preprocessing pipeline and spectral regularization ensure stable filter design. The resulting equalizer flattens the sensor response around resonance and significantly increases the effective bandwidth, at the expense of reduced peak sensitivity and signal-to-noise ratio (SNR) near resonance.

Keywords: magnetoelectric sensors, equalization, linear prediction, bandwidth extension, noise-based calibration, signal-to-noise ratio

1 Introduction

Magnetic sensors are employed in a wide range of applications, including automotive sensing, geological exploration, localization, and healthcare [2]. Various sensing principles are currently in use and under investigation, each offering specific advantages and constraints depending on the application. In recent years, magnetoelectric (ME) sensors have demonstrated significant potential, particularly in biomedical applications such as motion tracking [1]. Their advantages include cost efficiency, the capability for unshielded operation, miniaturizability, and the possibility of integration with complementary sensor systems.

The sensor system investigated in this work is based on a $1\text{ mm} \times 3\text{ mm}$ cantilever fabricated using a microelectromechanical systems (MEMS) process. It operates using direct detection readout and targets the first resonance frequency at approximately 6.7 kHz, where high sensitivity is achieved without requiring an external magnetic bias field. The printed cir-

cuit board (PCB), which hosts the peripheral electronics such as a low-noise charge preamplifier, also incorporates a multilayer planar coil system. This integrated coil structure enables the generation of a homogeneous magnetic field within the sensing region [2].

Building upon the proof-of-concept presented in [2], this work proposes a prediction-based approach for direct equalization of the sensor response using noise excitation generated by the PCB coil. Proper equalization enables an effective extension of the usable sensor bandwidth, which is particularly beneficial for multichannel applications, for instance in frequency-division multiplexing schemes [3].

To this end, a preprocessing pipeline consisting of cascaded high-pass and low-pass filters is introduced to condition the measured sensor signal. By exploiting prior knowledge of the resonance frequency, the signal can be efficiently downsampled, allowing the processing to focus on the relevant frequency band. Based on this preprocessed signal, the equalizer is derived using a linear prediction approach, enabling a joint compensation of magnitude and phase characteristics in a robust and computationally efficient manner. Experimental results demonstrate the feasibility and effectiveness of the proposed method.

The remainder of this paper is organized as follows: Section 2 presents the experimental setup, the preprocessing pipeline and the prediction-based equalization method. Section 3 discusses the experimental results, followed by conclusions and an outlook in Section 4.

2 Materials and Methods

The measurements were conducted using a bandlimited white noise signal to excite the sensor via the integrated PCB coil system. The Audio Toolbox in MATLAB [4] was used for signal generation and simultaneous acquisition of the sensor response. Data acquisition was performed using a RME Fireface UFX+ [5] at a sampling rate of 192 kHz.

The excitation signal was applied to the integrated coil structure on the sensor PCB, while the sensor output was recorded for further processing. All components were powered by a multichannel laboratory power supply (GPP-4323, Good Will Instrument Co., Ltd., Taipei, Taiwan [6]).

Lukas Schirmer*, Sören Lange*, Department of Electrical and Information Engineering, Kiel University, Kiel, Germany, E-mail: lusc@tf.uni-kiel.de, sola@tf.uni-kiel.de

Moritz Boueke, Keanu Kudicke, Henrik Wolfram, Michael Höft, Gerhard Schmidt, Department of Electrical and Information Engineering, Kiel University, Kiel, Germany, E-mail: mobo@tf.uni-kiel.de, keku@tf.uni-kiel.de, hewo@tf.uni-kiel.de, mh@tf.uni-kiel.de, gus@tf.uni-kiel.de

2.1 Processing Pipeline

As a basis for the equalizer design, the integrated coil system is excited using bandlimited white noise signals. Based on prior knowledge of the sensor characteristics, the first and second bending modes are located at approximately 6.7 kHz and 45 kHz, respectively. In this work, we focus on the first resonance and therefore use white noise bandlimited to a frequency range between 1 kHz and 10 kHz.

The sensor signal is initially acquired at a high sampling rate to capture the relevant frequency components. However, large portions of the spectrum are not relevant for the equalizer design. Therefore, a preprocessing pipeline consisting of cascaded filtering and downsampling is applied to isolate the frequency region of interest and reduce computational complexity.

First, a Butterworth high-pass filter of order 6 with a cutoff frequency of 3 kHz is applied to suppress low-frequency disturbances such as power line interference. Subsequently, a Butterworth low-pass filter of order 12 with a cutoff frequency of 10 kHz is used to attenuate high-frequency noise and suppress the second resonance mode. After filtering, the signal is downsampled by a factor of 8, resulting in an effective sampling rate of 24 kHz.

In a second stage, a Chebyshev Type I high-pass filter of order 12 with a cutoff frequency of 6.5 kHz is applied to further suppress residual low-frequency components. Consequently, a second downsampling step by a factor of 8 can be applied without loss of relevant information. This results in a signal that is predominantly confined to a bandwidth of approximately 1.5 kHz around the resonance frequency. This is possible because bandpass undersampling shifts the relevant portion of the spectrum into the baseband, where it occupies approximately the range from 0 to 1.5 kHz. The final signal is denoted as $\tilde{y}(n)$. Fig. 1 illustrates the spectral response during preprocessing. The estimated power spectral density (PSD) implicitly contains information about the sensor transfer function.

Since the goal of the proposed method is to derive an inverse filter for equalization, the problem can be interpreted as an inverse problem in the frequency domain. The PSD of the preprocessed signal is estimated in the frequency domain. For subsequent processing, it is normalized to its maximum value, yielding $S_{\tilde{y}\tilde{y},\text{norm}}(\mu)$. This normalization controls the dynamic range and improves numerical stability in the following steps. To further stabilize the inverse problem, a spectral floor is introduced:

$$S_{\tilde{y}\tilde{y},\text{mod}}(\mu) = \begin{cases} S_{\tilde{y}\tilde{y},\text{norm}}(\mu), & \text{if } S_{\tilde{y}\tilde{y},\text{norm}}(\mu) \geq \epsilon^2 \\ \epsilon^2, & \text{else.} \end{cases} \quad (1)$$

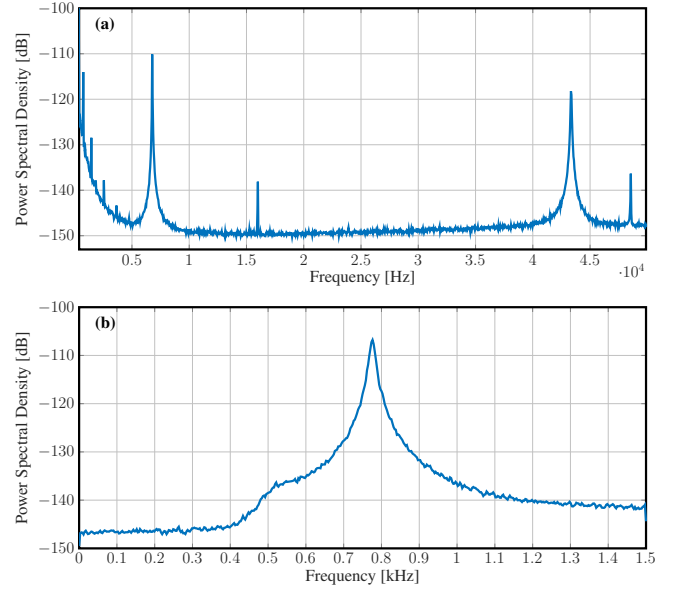


Fig. 1: (a) Power spectral density of the raw sensor response. (b) Power spectral density after preprocessing, showing the isolation of the resonance region.

This prevents excessive amplification of frequency components with low signal energy and improves the conditioning of the resulting autocorrelation matrix. Since the equalizer design corresponds to an inversion of the sensor response, small spectral values would lead to excessive amplification. This operation effectively limits the inversion to frequency components with sufficient signal energy. Finally, the autocorrelation estimation is obtained via inverse Fourier transform. The resulting autocorrelation sequence serves as the basis for the subsequent linear prediction-based equalizer design.

2.2 Prediction-Based Equalization Filter

Preprocessing restricts the signal to the frequency range of interest while reducing the sampling rate. Based on this signal, a linear prediction filter (LPF) approach is applied to derive an equalization filter.

Linear prediction is commonly used to model a signal as an autoregressive (AR) process and to whiten its spectrum. In this work, the same principle is exploited to implicitly approximate the inverse response of the sensor.

The prediction error is defined as

$$e(n) = \tilde{y}(n) - \hat{y}(n), \quad (2)$$

where $\hat{y}(n)$ denotes the linear prediction of the current sample based on past samples:

$$\hat{y}(n) = \sum_{i=1}^{N_{EQ}} h_i \tilde{y}(n-i). \quad (3)$$

This structure is visualized in Fig. 2.

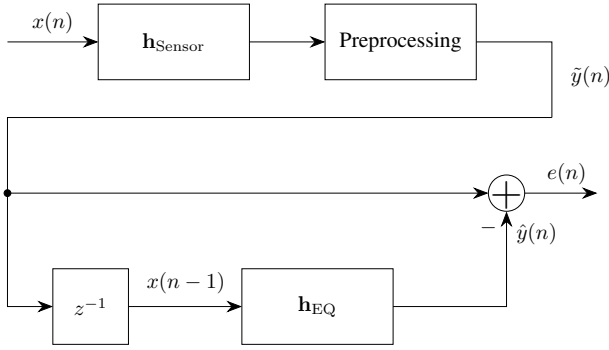


Fig. 2: Schematic representation of the equalization signal chain.

The optimal predictor coefficients are obtained by minimizing the mean squared error

$$E \{ e^2(n) \} \rightarrow \min. \quad (4)$$

This leads to the Yule–Walker equations, which can be written in matrix form as

$$\mathbf{R}_{\tilde{y}\tilde{y}} \mathbf{h} = \mathbf{r}_{\tilde{y}\tilde{y}}(1), \quad (5)$$

where $\mathbf{R}_{\tilde{y}\tilde{y}}$ is the Toeplitz autocorrelation matrix and $\mathbf{r}_{\tilde{y}\tilde{y}}(1)$ is the autocorrelation vector with one-sample lag [7]. The optimal coefficients are therefore given by

$$\mathbf{h} = \mathbf{R}_{\tilde{y}\tilde{y}}^{-1} \mathbf{r}_{\tilde{y}\tilde{y}}(1). \quad (6)$$

In practice, the system of equations is efficiently solved using the Levinson–Durbin algorithm [7]. The resulting prediction error filter is constructed as

$$\hat{\mathbf{h}} = [1, -\mathbf{h}^T]^T. \quad (7)$$

This filter corresponds to the inverse of the underlying autoregressive model and therefore acts as an equalizer for the sensor response.

The LPF is minimum-phase, i.e., all its zeros lie inside the unit circle,

$$|z_k| < 1 \quad \forall k \in \mathbb{R}, \quad (8)$$

which ensures a stable and causal implementation [8]. In addition, a minimum-phase system exhibits the smallest possible phase lag (or group delay) for a given magnitude response and possesses a stable causal inverse, which is advantageous for compensation and equalization tasks. This approach provides a numerically robust alternative to direct frequency-domain equalization.

Before analyzing the resulting system behavior, the design parameters of the equalizer are briefly discussed. The filter

length was set to $N_{\text{EQ}} = 200$, providing a sufficient model order to capture the sensor dynamics around the resonance. Spectral estimation was performed using an FFT length of $N_{\text{FFT}} = 2^{10}$, providing sufficient frequency resolution to accurately capture the resonance characteristics and ensure a reliable autocorrelation-based filter design. As shown in (6), the filter coefficients are obtained by inverting the autocorrelation matrix. Consequently, the conditioning of this matrix directly affects the numerical stability of the solution and the robustness of the resulting equalizer.

3 Results and Discussion

To evaluate the resulting system behavior, the effective transfer characteristics are analyzed. For this purpose, the PSD of the sensor response before and after equalization is compared. The comparison of the PSDs before and after equalization re-

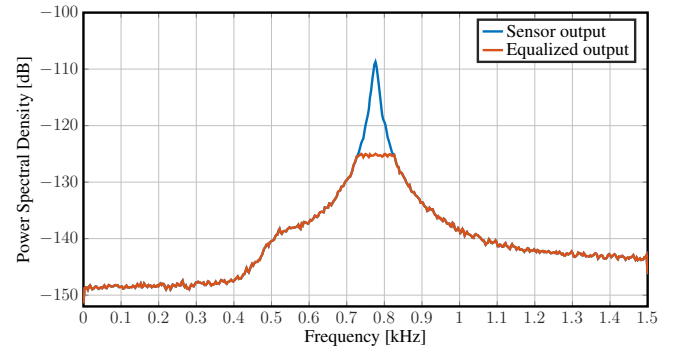


Fig. 3: Sensor output before and after equalization, demonstrating the reduction of the resonance peak and the increased effective bandwidth.

veals a significant reduction in sensitivity at the resonance frequency. At the same time, a considerably more uniform spectral distribution is achieved over an extended frequency range, increasing the effective bandwidth from approximately 10 Hz to about 150 Hz around the resonance. This improvement in usable bandwidth is accompanied by a reduction in signal-to-noise ratio (SNR), as the equalization suppresses the resonance amplification that originally provided the highest sensitivity. This demonstrates that the dominant resonance behavior of the sensor is mitigated, resulting in a more evenly distributed sensitivity. This also increases the relative influence of noise outside the original sensitivity peak, representing an inherent trade-off of the equalization process. Consequently, the proposed method enables a substantial extension of the usable bandwidth while maintaining stable system behavior due to the regularized inversion.

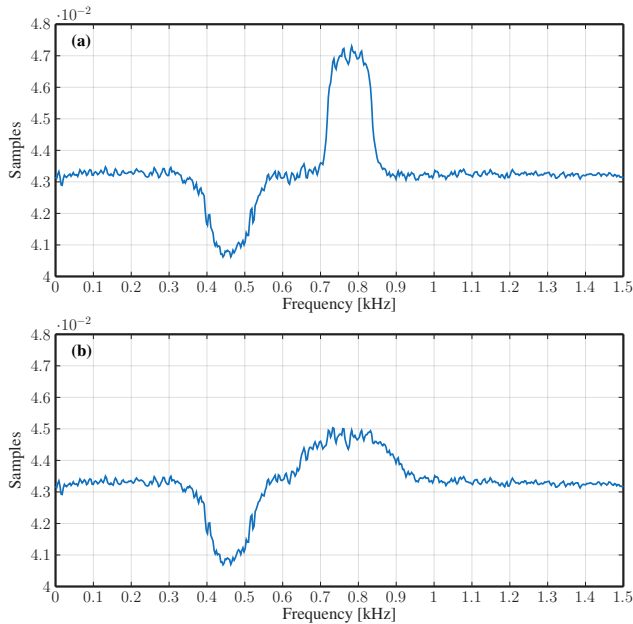


Fig. 4: (a) Group delay of the sensor response. (b) Comparison of the group delay before and after equalization, showing a significant reduction of dispersive effects and a more constant temporal response.

Furthermore, the group delay analysis indicates a reduction of frequency-dependent delay variations, resulting in a better temporal response, as seen in Fig 4. A fully linear group delay cannot be achieved, which can be attributed to the regularization of the inverse problem as well as the limited model order of the predictor.

4 Conclusion and Outlook

In this work, a prediction-based equalization approach for magnetolectric sensor systems has been presented. By exploiting noise excitation and linear prediction, an implicit inverse model of the sensor response is obtained. The proposed preprocessing pipeline efficiently reduces the signal bandwidth while preserving the relevant spectral information around the resonance frequency.

The equalization method results in a significant flattening of the sensor response and a substantial extension of the usable bandwidth. The resonance-dominated behavior is mitigated, leading to a more uniformly distributed sensitivity over the frequency range of interest.

Furthermore, the group delay analysis indicates a reduction of frequency-dependent delay variations, resulting in a better phase response. This indicates an improvement in both amplitude and phase characteristics of the system.

Due to the regularization of the inverse problem and the fi-

nite model order, a complete equalization cannot be achieved. Nevertheless, the results confirm that the approach provides a robust and computationally efficient solution for bandwidth enhancement of resonant sensor systems.

Future work will focus on achieving a more linear group delay and investigating extensions toward linear-phase equalization, while preserving the advantageous minimum-phase characteristics of the current filter structure. In this context, cascaded filter structures based on time-reversed versions of the equalizer coefficients, i.e., $h_{EQ}(n) \rightarrow h_{EQ}(N - 1 - n)$, may be investigated to further improve the phase behavior and reduce residual delay variations.

In addition, the approach will be further studied using sinusoidal excitation for targeted frequency-selective system characterization and extended to multi-channel as well as adaptive scenarios.

References

- [1] J. Hoffmann, H. Wolfram, E. Engelhardt, M. Boueke, T. Schmidt, J. Welzel, M. Höft, W. Maetzler, and G. Schmidt, "A magnetolectric distance estimation system for relative human motion tracking," *Sensors*, vol. 25, no. 2, p. 495, Jan. 2025. [Online]. Available: <https://www.mdpi.com/1424-8220/25/2/495>
- [2] H. Wolfram, J. Hoffmann, R. Burgardt, E. Elzenheimer, G. Schmidt, and M. Höft, "Pcb coil enables in situ calibration of magnetolectric sensor systems," *Current Directions in Biomedical Engineering*, vol. 9, no. 1, pp. 567–570, 2023. [Online]. Available: <https://doi.org/10.1515/cdbme-2023-1142>
- [3] J. Hoffmann, E. Engelhardt, M. Boueke, J. Welzel, C. Hansen, W. Maetzler, and G. Schmidt, "Pilot study: Step width estimation with body-worn magnetolectric sensors," *Sensors*, vol. 25, no. 11, 2025. [Online]. Available: <https://www.mdpi.com/1424-8220/25/11/3390>
- [4] I. The MathWorks, Audio Toolbox, Natick, Massachusetts, United State, 2025. [Online]. Available: <https://www.mathworks.com/help/audio/>
- [5] RME Audio, "Fireface UFX+," 2025. [Online]. Available: <https://rme-audio.de/fireface-ufx.html>
- [6] Good Will Instrument Co., Ltd., "GPP-series multi-output programmable D.C. power supply: GPP-4323," 2020. [Online]. Available: <https://www.gwinstek.com/global/products/downloadSeriesDownNew/14230/1737>
- [7] Mader, Andreas, Henning Puder, and Gerhard Uwe Schmidt. "Step-size control for acoustic echo cancellation filters—an overview." *Signal Processing* 80.9 (2000): 1697-1719.
- [8] Mertins, Alfred. "Parameter-und Signalschätzung." *Signaltheorie: Grundlagen der Signalbeschreibung, Filterbänke, Wavelets, Zeit-Frequenz-Analyse, Parameter-und Signalschätzung*. Wiesbaden: Springer Fachmedien Wiesbaden, 2023.