# Time reversal and Inverse filter for Blind Deconvolution of Ultrasonic Communication Through Solid Channel

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***Abstract*— Ultrasonic waves provide an effective means of transmitting information through solid media, such as metal pipes and bars. However, the complex geometry of these materials causes reverberations that degrade the quality of communication. In this paper, we propose denoising approaches using time reversal and inverse filtering methods for blind deconvolution. These methods enable significant improvements in signal-to-noise ratio (SNR), with the inverse filter demonstrating superior performance in both simulated and real-world scenarios. Our results show that this approach provides a robust solution for improving ultrasonic communication in complex solid channels.**

**Keywords—Ultrasonic Communication System, Blind Deconvolution, Time Reversal Filter, and Reverse Filter**

## I. Introduction

Ultrasonic waves serve as robust carriers for transmitting information through solid communication channels, such as metal pipes, rectangular bars, or plates. However, the geometric intricacies inherent in these channels give rise to complex reverberations, including surface waves, Lamb waves, longitudinal waves, reflections, refractions, dispersions, mode conversions, and scattering phenomena. These reverberations pose significant challenges and substantially lead to signal degradation and reduced communication reliability[1][2]. To address these challenges, signal processing techniques play a pivotal role. Denoising algorithms are employed to eliminate unwanted noise from the received signal, thereby enhancing clarity and reliability[3][4][5]. Mitigating intersymbol interference (ISI) is crucial for maintaining the integrity of transmitted data, especially in environments prone to channel distortions. Additionally, enhancing the signal-to-noise ratio (SNR) is essential for improving communication robustness, particularly in noisy environments. Existing methods, such as pusle shape filter and wavelet transform denoising techniques[5][6], fall short of providing the necessary signal clarity, especially in the face of complex multipath effects.

Blind deconvolution, which aims to recover the original signal without prior knowledge of the channel characteristics or distortion effects, is a promising approach to address these challenges [7]. In this paper, our approach employs two filtering techniques—time reversal filtering and inverse filtering—for blind deconvolution. By using these filters, we can significantly reduce the effects of multipath propagation and improve signal recovery, significantly improving signal clarity and SNR. The application of these techniques to both simulated and real-world data demonstrates the effectiveness of our approach.

## II. Time Reversal Filter and Inverse Filter

Time Reversal (TR) filtering has been extensively studied in underwater acoustic communication due to its ability to support high data rates [8][9]. The TR filter effectively treats the solid communication channel as a matched filter, allowing it to refocus the transmitted wave at the receiver. This technique capitalizes on the reciprocal nature of wave propagation in linear and time-invariant media. The TR filter is defined as

$$p(t) = h(-t)$$

where $h(t)$ represents the measured channel impulse response (CIR), and $h(-t)$ is the time-reversed CIR.

The output of the TR filter can be expressed as:

$$\widehat{m}(t) = \big(m(t) * h(t) + n(t)\big) * h(-t)$$

Simplifying this expression yields:

$$\widehat{m}(t) = \big(h(t) * h(-t)\big) * m(t) + h(-t) * n(t)$$

In this equation, $h(t) * h(-t)$represents the autocorrelation of the CIR, which effectively sharpens the received signal. As the complexity of the communication environment increases, so does the duration of the CIR, leading to an increase in the peak amplitude of the autocorrelation, which enhances the refocusing of the signal at the receiver. However, a key limitation of TR filtering is the presence of temporal sidelobes in the refocused signal, which introduce residual inter-symbol interference (ISI). This occurs due to imperfect refocusing, especially when the channel characteristics are complex. Additionally, achieving optimal performance requires accurate estimation of the CIR, as any errors in this estimation degrade the effectiveness of the filter. In our study, we utilize an arbitrary function generator to produce an impulse for the transmitter, and an oscilloscope to capture the CIR during the experiment. For the simulation, we use a half-sine wave as the excitation impulse, and the received signal serves as the CIR. The TR filter remains a compelling and straightforward method for signal recovery, thanks to its ease of implementation and effectiveness in various communication environments.

The frequency inverse filter is a widely used deconvolution method in digital communication, particularly for compensating channel distortions and reducing inter-symbol interference (ISI) to recover the transmitted signal. This filter works by normalizing the channel's frequency response to unity, effectively counteracting the distortions introduced by the communication channel.

The frequency response of the channel can be expressed as:

$$H(f) = FFT\big(h(t)\big) = |H(f)|e^{j\varphi(f)}, -\frac{f_s}{2} < f < \frac{f_s}{2}$$

where $|H(f)|$ is the magnitude response, $\phi$(f) is the phase response, and $f_s$ is the sampling frequency.

To perform inverse filtering, we define the inverse filter in the frequency domain as:

$$W(f) = \frac{1}{|H(f)|}e^{-j\varphi(f)}, -\frac{f_s}{2} < f < \frac{f_s}{2}$$

This shows that, ideally, the inverse filter completely eliminates the channel effects, leaving only the original signal and the filtered noise.

However, the challenge with frequency inverse filtering is that the reciprocal of the channel's frequency response, $1/\,|H(f)|$, can be unstable. If the magnitude response $|H(f)|$ is very small or zero at certain frequencies, the inverse filter produces very large or undefined values, leading to poor performance or even failure in signal recovery.

To mitigate this issue, the Wiener filter [11] is often used. It improves stability by balancing signal recovery with noise reduction. The Wiener filter is defined as:

$$\widehat{H}(f) = \frac{\sigma_S^2 H^*(f)}{\sigma_S^2|H^2(f)| + \sigma_n^2}$$

where $\sigma_S^2$ is the variance of the input signal, $\sigma_n^2$ is the variance of the noise, and $H^*(f)$ is the complex conjugate of $H(f)$. This filter provides a more stable solution by weighting the contribution of each frequency component based on the signal and noise characteristics, reducing the impact of frequencies with small or zero amplitudes.

Additionally, traditional deconvolution methods like the inverse filter struggle to handle Gaussian white noise, which is typically present in communication channels. The Wiener filter helps address this limitation by incorporating noise statistics into the filter design, allowing for more robust signal recovery.

## III. Simulation and Experimental Setup

In the simulation, we used the Finite Element Analysis (FEA) method, implemented through the ABAQUS software tool, to model the bit-wavelet spreading and multipath effects caused by the communication channel. The channel was represented by an aluminum flat bar with dimensions of 100 × 10 × 1 cm, serving as the medium for signal propagation. Fig. 1 illustrates the geometric configuration of the aluminum flat bar (ARB channel), including the positions of the transmitter and receiver. This setup allowed us to accurately simulate how ultrasonic waves travel through the solid channel, capturing the resulting reflections, mode conversions, and other reverberation effects that contribute to the complexity of the multipath phenomenon.

In the experimental setup, we used an aluminum rectangular bar (ARB) with dimensions of 100 × 6.5 × 2 cm as the communication channel. The transmitter and receiver were positioned at opposite ends of the ARB to facilitate signal transmission and reception. The geometric layout of the experimental channel is depicted in Figure 2. For signal transmission and detection, we employed piezoelectric lead zirconate titanate (PZT) transducers, which have a center frequency of 2.5 MHz. These PZT sensors were selected for their high sensitivity and compatibility with the frequency range required for ultrasonic communication in solid channels.

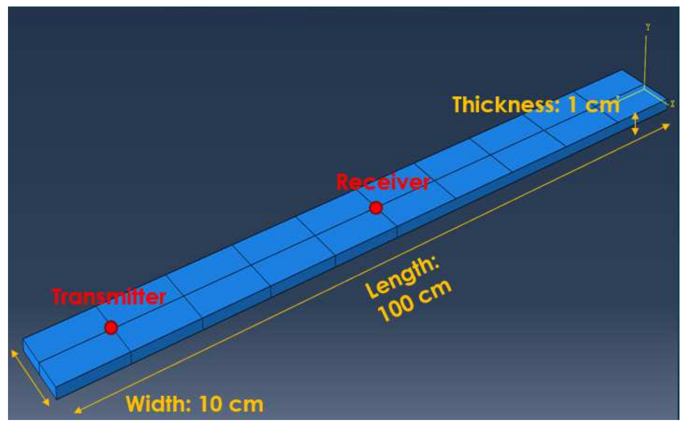


Fig 1. Simulation Channel


Fig. 2. Experimental Channel.

## IV. Test Results

In this section, we compare and analyze the performance of two deconvolution filters applied to ultrasonic communication, using both simulation and experimental data. A random binary sequence was transmitted through both the simulated and experimental channels to evaluate filter performance.

Fig. 3 shows the simulation results. In Fig. 3a, the original transmitted binary information is displayed, with red lines indicating the positions of binary '1's and green lines indicating the positions of binary '0's. Fig. 3b illustrates the received signal after propagating through the simulated communication channel. Due to the multipath effects and reverberations in the solid channel, the received waveform is highly distorted, consisting of several scattered wave packets that correspond to different propagation paths. This distortion resulted in a signal-to-noise ratio (SNR) of 5 dB in the simulation.

Fig. 4 presents the results for the experimental channel, showing both the transmitted and received signals. The received signal in this case had an SNR of 7 dB. In both the simulation and experimental scenarios, the received waveforms were too distorted to directly retrieve the binary information ('1' or '0') from the raw signals, as the bit positions were no longer distinguishable due to the overlap of scattered wave packets.

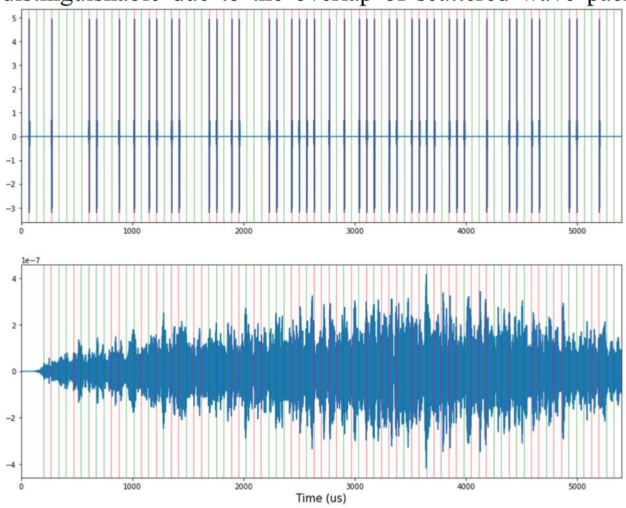


Fig. 3. Multiple binary bits test of simulation data. (a) transmitted digital information (b). received waveform.

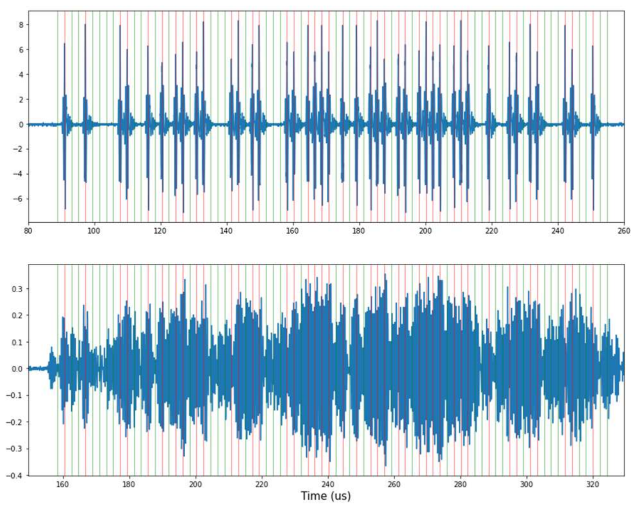


Fig. 4. Multiple binary bits test of experimental data. (a) transmitted digital information (b). received waveform.

Table 1 presents the SNR improvements achieved using the time reversal and inverse filters in both the simulation and experimental setups.

In the simulation, applying the time reversal filter improved the SNR by 14 dB, raising it from 5 dB to 19 dB. The inverse filter, while less effective in the simulation, provided a 5 dB improvement, resulting in a final SNR of 10 dB. These results indicate that both filters are effective, with time reversal performing particularly well under simulated conditions.

In the experimental setup, the time reversal filter improved the SNR by 8 dB, increasing it from 7 dB to 15 dB. However, the inverse filter significantly outperformed the time reversal filter, providing a 16 dB improvement and achieving a final SNR of 23 dB. These findings confirm the superior performance of the inverse filter in real-world conditions, leading to substantial improvements in signal quality.

TABLE I. SNR IMPROVEMENT.

| Method | Simulation SNR Improvement | Experimental SNR Improvement |
|---|---|---|
| Time Reversal | +14 dB | +8dB |
| Inverse Filtering | +5 dB | +16 dB |

In this paper, we proposed the use of time reversal and inverse filtering for blind deconvolution in ultrasonic communication through solid channels. Our approach significantly improves signal recovery by reducing the impact of multipath effects and reverberations. The experimental results, which showed an SNR improvement of up to 23 dB, demonstrate the potential of these techniques to enhance communication reliability in solid channels. With the ability to achieve data transmission rates of up to 1 Mbps, this method offers a practical solution for real-world ultrasonic communication applications.

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