

Comparative Study of Signal Processing Algorithms for Noise Reduction and Sonar Signal Reconstruction

Kavya Krishna P J¹, Anulekha K A², Niharika V Anil³, Aleena Merin Anto⁴, Abhilash O S⁵

Electronics and Communication Engineering

Christ College of Engineering

Thrissur, India

kavyakrishnapj@gmail.com¹, abhilashiit@gmail.com⁵

Abstract—Sonar signal processing plays an important role in extracting information from the unprocessed acoustic data. The processed data ensures accurate target identification. This paper represents a comparative analysis on different signal processing algorithms. The algorithms implemented on the sonar data include Non-negative Matrix Factorization (NMF), Least Mean-Squares (LMS) with variable step size, Hybrid Kalman and Wiener. The different signal parameters such as Signal-to-Noise Ratio (SNR), processing time, reconstruction error, convergence speed, noise robustness, and computational complexity for each algorithm are computed. Based on this, scope of each algorithm on different applications are analyzed.

Keywords— Sonar Signal Processing, Non-negative Matrix Factorization (NMF), Least Mean-Squares (LMS) with variable step size, Hybrid Kalman Filter, Wiener Filter

I. INTRODUCTION

A decisive role is played by sonar signal processing in applications including target detection, mapping, defense, communication, and environmental studies. Basic filters were involved in traditional methods for sonar signal processing. With the advancement in signal processing algorithms, more effective feature extraction with improved noise reduction from noisy environments was made possible.

Frequency domain BSS, deep learning, and other methods are used by the modern technique to enhance signal quality and reduce noise interference. For instance, convolution based blind source separation algorithm had a promising impact on active sonar signal separation. Reverberation was suppressed by the algorithm, and detection accuracy was improved by time and frequency analysis [1]. Additionally, the amplitude correlation of neighboring frequency points is organized, and segregated signals are arranged in the frequency domain by frequency domain blind source separation (BSS) [2].

An adaptive matched filter algorithm was developed that is used in conditions of reduced signal-to-noise ratio, incorporating features of the conventional matched filter and correlation denoising. Echo detection was significantly improved by the algorithm, and a -20 dB echo signal can be detected after beamforming. In comparison with the conventional matched filter, performance was noticeably improved [3].

The disadvantages of the traditional adaptive algorithm are overcome by the use of the LMS algorithm with a variable step size, and quick convergence is provided. It effectively suppresses impulse noise and achieves minimal steady-state error [4].

Poor convergence was exhibited by conventional NMF methods due to reverberation characteristics not being accounted for. On the other hand, a modified sparse updating mechanism and a pre-trained reverberation basis matrix are incorporated into an improved NMF-based reverberation reduction technique [5]. In addition, the limitations caused by conventional NMF algorithms on continuous wave reverberations were overcome by two preprocessing techniques, such as dechirping transformation and modulation operation, and by applying the NMF method to Linear Frequency Modulation reverberation [6].

To obtain optimal parameters for various signal-to-noise ratios (SNR) in advance, models are trained, and parameters are also adjusted based on the third-dimensional normalized amplitude of the input data, a variable parameter DBSCAN algorithm had been developed [7]. Furthermore, emerging methods such as FPGA (Field-Programmable Gate Array), where the algorithm is broken down based on the features and the right structure is chosen to lower system volume and power consumption, are used [8].

A comparative analysis is represented in this paper that highlights the suitability of algorithms in various applications. Furthermore, the result in identification of a more efficient sonar system, which is suitable for real-life application.

II. BLOCK DIAGRAM

The general block diagram of sonar signal processing is mainly involved with three components: input, signal processor, and output. The acquired raw sonar data is represented by the input. The core of the system is the signal processor block, in which signal processing algorithms are applied to the input data. The processed signal, which contains information and helps in target identification, is represented by the output.