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Low SNR communications using a double Wiener filter in an upwelling environment

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This paper presents a double Wiener filter approach for low SNR communications. Low power superimposed training bitstreams are transmitted and recorded on a four-hydrophone pyramidal array. The method uses a minimum mean square error (MSE) Wiener filter for channel equalization. To explore temporal diversity, a fast Hadamard transform estimates the channel impulse responses for synchronization of the filtered bitstreams. Coherent averaging of time-aligned signals results in an error-corrected bitstream. A second Wiener filter is then applied over the averaged sequence to eliminate residual intersymbol interference. After removing the probe interference, the original message is recovered. To prove the concept, a small subset of data collected during the experiment BioCom'19 was used. The short-range experiment was performed in an upwelling shallow-water site, in Cabo Frio Island bay (Brazil), a challenging environment for communications due to the rapid ocean temperature stratification. Despite BER fluctuation, the proposed scheme proved robust, dealing with these short time scale signal fluctuations and high noise levels that hamper efforts to recover the message. The four channels were processed independently. Compared to a previous method based on a single Wiener, the double Wiener filter approach achieved better performance, providing an average MSE gain up to 2.8 dB.

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1. INTRODUCTION

Research on low signal-to-noise ratio (SNR) underwater acoustic communications has been encouraged, in recent years, due to their numerous applications.[1] The command and control of battery-operated autonomous underwater vehicles (AUV) and the telemetry of sensors used for environmental monitoring have motivated low power demanding communication signals to extend their operating life. In a hostile scenario, these low SNR signals may also keep underwater vehicles and sensors undetected while communicating. Furthermore, several countries have strict policies in place to regulate the power of acoustic transmissions, and therefore, to mitigate impacts on marine life.[1] However, establishing low SNR communications in complex channels such as upwelling shallow water is a challenging task.[2] The rapid signal fluctuation in time and space, a long multipath delay spread, and the high noise levels due to human and biological activities are factors that impact signal demodulation, motivating new signal processing approaches.

Most common state-of-the-art coherent-modulated communication systems are based on nonlinear adaptive filters such as decision feedback equalizers (DFE) along with spatial diversity combining.[3] However, DFE is computationally intensive and has presented poor results in low SNR environments due to uncertainties on channel estimation and synchronization.[4] Furthermore, spatial diversity requires several hydrophones, not suitable for AUV and other single sensor applications.

Computationally simpler than DFE, the optimum linear Wiener filter[5] has been widely studied in the literature. Minimizing the mean square error (MSE) between the transmitted and the estimated signal, the filter has been used for noise reduction, channel equalization and signal estimation.[6–8] Thus, a previous paper has presented a superimposed training method for low probability of detection underwater acoustic communications based on a single-stage Wiener filter.[9]

This paper proposes a double Wiener filter scheme, for low SNR communications. To explore temporal diversity, broadband low power superimposed training bitstreams are transmitted. The signals are Wiener filtered for channel equalization. Then, a fast Hadamard transform (FHT)[10] estimates the channel impulse responses (CIR) for time synchronization of each filtered sequence. A coherent averaging of several time-aligned signals results in an error-corrected averaged signal. A second Wiener filter is then applied over the time-averaged sequence to mitigate residual intersymbol interference. After removing the intentional training sequence interference, the original message is recovered.

To demonstrate the double stage Wiener filter approach, this paper relies on a small subset of data from the BioCom'19 experiment.[11] The short-range experiment was performed in an upwelling shallow-water site, in Cabo Frio island bay (Brazil).[12] The water column temperature at the receiver location was monitored to observe the upwelling occurrence. Communication bitstreams were transmitted and recorded by four hydrophones on a pyramidal array. The estimated CIR were compared for different temperatures at the receiver depth to observe the degradation of the communication system along time. To evaluate the performance in terms of bit error rate (BER) and MSE, the four channels were processed independently. Moreover, the BER vs. SNR and the MSE vs. time were plotted to compare the single and double Wiener approaches.

2. DOUBLE WIENER FILTER APPROACH

The Wiener filter[5] is an optimal linear filter under the minimum mean square error (MSE) criterion. Suitable for linear time-invariant (LTI) systems, the Wiener filter produces an estimate of the transmitted signal $g(k)$ filtering a distorted and noisy received signal $y(k)$ (Fig. 1). In most communication systems, there is no previous knowledge about the emmited signal. Thus, to design the filter and to optimize its tap weights, a training sequence $x(k)$, known to the receiver, must be transmitted.

In discrete-time domain, the estimated error signal $e(k)$ is given by Eq. (1)

$$
\begin{aligned}
e(k) &= x(k) - g(k) \\
&= x(k) - \sum_{m=0}^{M-1} w(m) y(k-m)
\end{aligned}
\tag{1}
$$

where k is the discrete-time index, M is the order of the filter, and $w(k)$ are the Wiener filter coefficients.

Figure 2 presents the block diagram of the proposed double Wiener filter communication system. Most common approaches use training sequences periodically inserted into the data stream to perform channel estimation. However, the presented system relies on a superimposed training method.[9] In this work, Z consecutive broadband 2-PSK bitstreams $y(t)$ of period τ are recorded in each channel. The bitstream is composed of a long m-sequence (2047 bits) overlaid to the message. The message stream is formed by 4 data packets of 511 bits and three empty bits. M-sequences are usually used as training signals, due to their impulse-like autocorrelation. Thus, each data packet also includes a short m-sequence of 31 bits for hard synchronization.

Assuming that the input signal $y(k)$ and the training sequence $x(k)$ are both zero-mean stationary random processes, the MSE $\xi(k)$ can be written as Eq. (2)

$$
\begin{aligned}
\xi(k) &= E[e^2(k)] \\
&= E[(x(k) - \mathbf{w}^{\mathbf{T}}\mathbf{y})^2] \\
&= \sigma_x^2 - 2\mathbf{w}^{\mathbf{T}}\mathbf{r}_{\mathbf{xy}} + \mathbf{w}^{\mathbf{T}}\mathbf{R}_{\mathbf{yy}}\mathbf{w}
\end{aligned}
\tag{2}
$$

where $E[.]$ is the expectation operator, $\mathbf{w}^{\mathbf{T}}$ is a transpose of the $M \times 1$ vector of the Wiener filter coefficients, $\mathbf{y}$ is the $M \times 1$ vector of the received signal, σ_x^2 is the variance of $x(k)$, $\mathbf{R}_{\mathbf{yy}}$ is an $M \times M$ Hermitian Toeplitz matrix of autocorrelation of the input signal, and $\mathbf{r}_{\mathbf{xy}}$ is the cross-correlation vector of the input and training signals.

In Eq. (2), if the matrix $\mathbf{R}_{\mathbf{yy}}$ is invertible, the solution for the optimum Wiener filter tap weights $w(m)$, known as the Wiener-Hopf equation, is given by Eq. (3)

$$
\mathbf{w} = \mathbf{R}_{\mathbf{yy}}^{-1}\mathbf{r}_{\mathbf{xy}}
\tag{3}
$$

In a low SNR environment, due to the severe noise corruption of the transmitted signal, channel estimation is imprecise causing synchronization problems, and therefore, reducing the system performance. Furthermore, the upwelling channel is fast time varying. Assuming that the system is LTI for the time-slot of each received bitstream, the Wiener filter coefficients $w(m)$ are estimated to minimize the MSE between each filter output $g_z(t + z\tau)$ and the training signal $x(k)$, reducing the additive noise, compensating the amplitude/phase distortions, and mitigating intersymbol interference on data packets.

A fast Hadamard transform[10] of the filtered sequences $g_z(t + z\tau)$ estimates the channel for each time slot $(t + z\tau)$. Based on the CIR $h_z(t + z\tau)$ peaks, and considering the strongest peak of $h_{1Ref}(t)$ as the

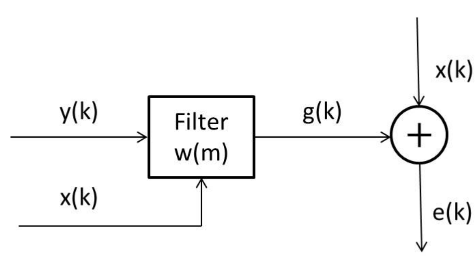

Figure 1: Diagram of the Wiener filter for linear MMSE estimation.

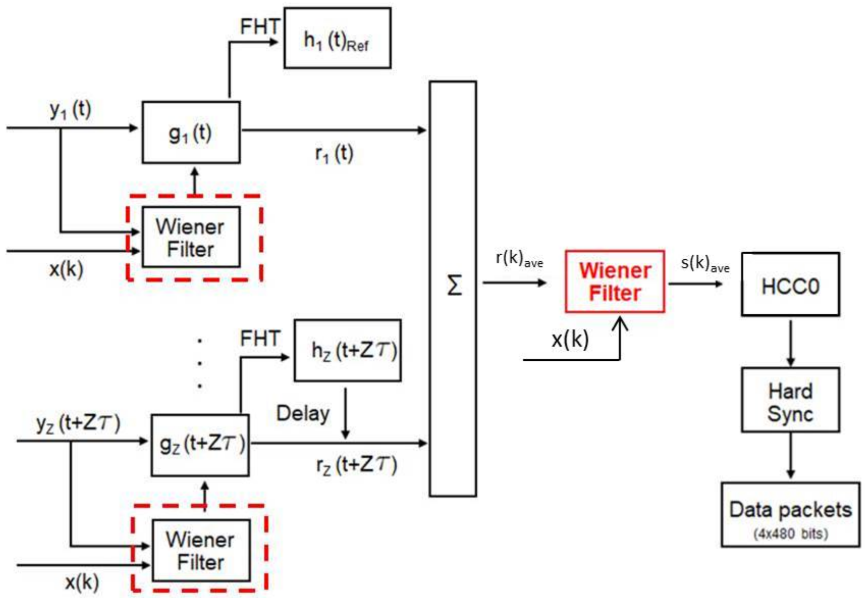

Figure 2: Diagram of the double Wiener filter communication system shows the following steps: channel equalization (single Wiener filter), soft synchronization, temporal coherent averaging of Z received bitstreams, the double Wiener filter over the averaged bitstream, probe interference removal (HCC0), hard synchronization, and data retrieval.

reference, the signals are synchronized in time and coherently averaged. Averaging of the single filtered sequences $r_z(t + z\tau)$ performs error correction and provides a higher SNR bitstream estimation compared to the low power received signals.

Calculated in a similar way to the first filter, but over the averaged sequence $r(k)_{ave}$, the second Wiener filter coefficients $W(m)$ compensate the residual intersymbol interference Eq. (4)

$$\mathbf{W} = \mathbf{R_{rr}^{-1} r_{xr}} \tag{4}$$

To remove the m-sequence interference from the double Wiener filtered bitstream $s(k)_{ave}$, a technique named "Hyperslice Cancellation by Coordinate Zeroing (HCC0)[13] is used. Based on the FHT of the short m-sequences of 31 bits, the packets are hard synchronized and the message retrieved.

3. BIOCOM'19 EXPERIMENT

The BioCom'19 experiment took place in a shallow-water site, in the Cabo Frio island bay (Brazil) from Jan. 14-18, 2019. The region is a challenging underwater communications environment due to the upwelling oceanographic phenomena that influence the ocean temperature stratification, and therefore, the acoustic propagation. Low power superimposed training bitstreams were continuously transmitted during the experiment. The m-sequence overlaid to the message is an orthogonal code and provides an impulse-like autocorrelation, used to estimate the channel conditions under several temperature regimes. The signals were set up with a central frequency of 7.5 kHz, bandwidth of 3 kHz, and were emitted from an omnidirectional acoustic source placed at mid-water in a 4 m deep water column. To explore channel temporal diversity, each bitstream was transmitted 55 times per minute, repeated every five minutes. The signals were recorded on a four-hydrophone array, placed 1.6 km away from the source. The hydrophones were mounted on a pyramidal frame of 1m-long edge posed at the bottom in an 8 m water depth. Furthermore, to monitor the upwelling during acoustic transmissions, a time-series of temperature profiles were continuously recorded along the water column at the receiver location.

A. CIR VARIABILITY DURING UPWELLING

Coastal upwelling off the Cabo Frio island is a complex oceanographic phenomenon.[12] Triggered by the NE/E wind blowing regime, the process induces the movement of cold water towards the sea surface, severely modifying the ocean temperature stratification. Sound speed in the ocean is strongly dependent on the temperature which also modifies the index of refraction of the different layers. Therefore, the physics of the acoustic propagation is directly affected by the changes in the ocean temperature. In shallow water, the multiple interactions of the sound wave with the sea surface and the bottom impose an additional challenge to the communication system. Multiple late arrivals create intersymbol interference, degrading the communication system performance.

Figure 3 (bottom plot) shows the temperature evolution in time measured by a thermistor placed at the same depth of the hydrophone #1, approximately 1.5 m above the bottom. To observe the channel degradation as upwelling occurs, Fig. 3 (top plot) presents three different CIR structures estimated from signals recorded by hydrophone #1 (top of the array) in a restricted time frame from Jan. 17, 4 p.m. to 8:30 p.m.

During Jan. 17, from 4 p.m. to 7 p.m., the temperature at the receiver remained approximately stable. However, the structures of the two CIR shown in Fig. 3 (top plot, left and center) indicate the occurrence of the upwelling. As cold water seeps into the bay through the depression located at mid-range between the source and the receiver, the sound speed profiles (SSP) changed from a warm isothermal to a downward

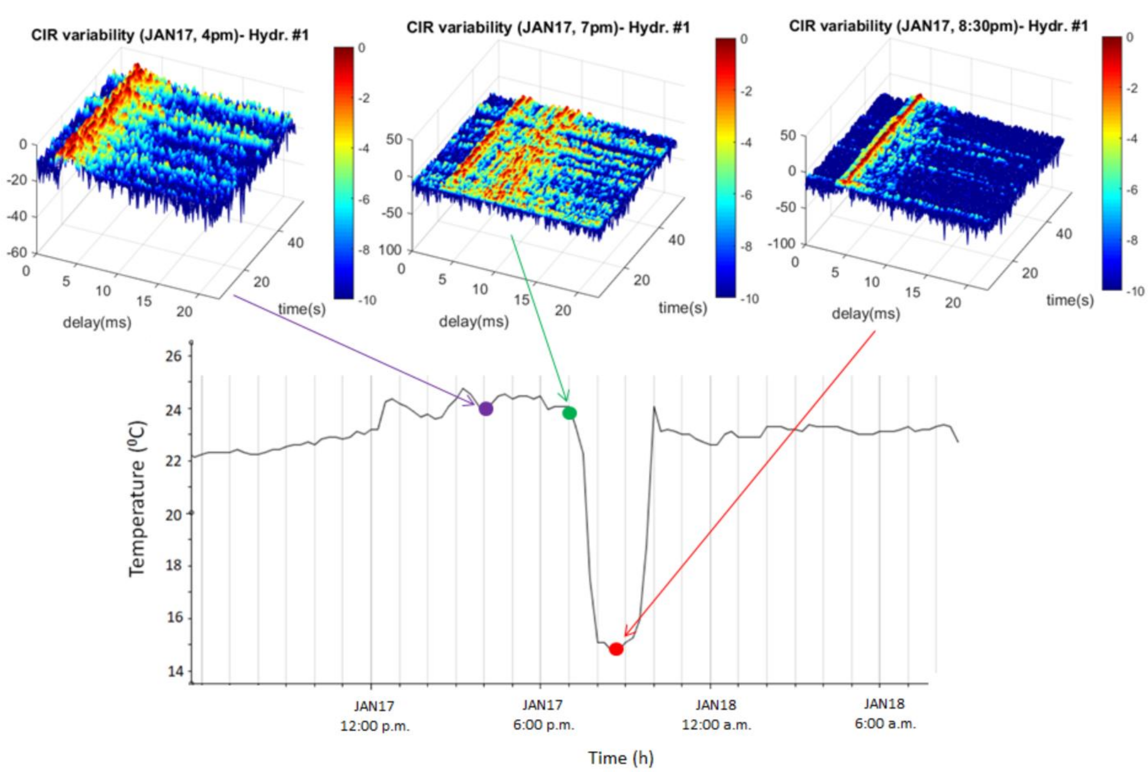

Figure 3: Top plot: CIR variability in a restricted upwelling time frame from Jan. 17, 4 p.m. to Jan. 17, 8:30 p.m. Bottom plot: temperature evolution in time, measured by a thermistor located close to hydrophone #1, approximately 1.5m above the bottom.

refracting condition. At 4 p.m., the CIR presents stronger arrivals and a short multipath spread of approximately 5 ms. However, at 7 p.m., despite the temperature at the receiver was still around 24ºC, the cold water had already mixed with warm layers along the propagation track. As a consequence, the CIR at 7 p.m. presents weaker arrivals, spreading over more than 10 ms. Comparing both CIR plots (4 p.m. and 7 p.m.), one can expect a severe degradation of the communication system due to the longer multipath spread and the reduction of the available acoustic energy at the receivers.

As the cold water spread throughout the channel and moved towards the sea surface, a drastic short-term drop of temperature of about 10ºC was observed at the receiver location from 7 p.m. to 8:30 p.m. Observing the respective CIR shown in Fig. 3 (top plot, center and right), one can conclude that the channel evolved from a downward refracting to a cold isothermal SSP, a more stable channel condition. Thus, at 8:30 p.m., the arrivals are strong and the CIR presents a short multipath spread of approximately 2 ms, indicating an improvement of the system performance.

4. COMMUNICATION PERFORMANCE RESULTS

To observe and compare the performance of the single and the double Wiener filter approaches in this challenging environment, a small subset of data from the shallow water BioCom'19 experiment was used. From Jan. 17, 4 p.m. to Jan. 18, 1 a.m., once every hour, the four channels of the pyramidal array were analyzed independently. Doppler is generally an issue in coherent communication systems. However, in this work, as the source and receiver were steady in the water, Doppler effects were not relevant. The in-band SNR (dB), for each channel, was estimated according to $SNR = 10log_{10}[(S - N)/N]$, where S is the mean of the signal plus noise power of 20 sequences, and N is the mean of the noise power of a sequence of the same length of S, estimated from a period after transmissions. The BER were estimated averaging 20 bitstreams, from the 55 in each file, equivalent to an effective bit rate of 22 bps. Increasing the number of sequences in the averaging process improves the error correction and tends to provide a better estimation of the transmitted signal, masking the efficiency of the proposed double Wiener filter scheme.

Figure 4 shows the SNR vs. BER estimated from all four channels, independently. Despite the power of the transmitted signals being kept constant, SNR levels varied from -3.9 to 7.3 dB. This large SNR variation is related to both the channel degradation caused by the upwelling that reduced the acoustic energy at the

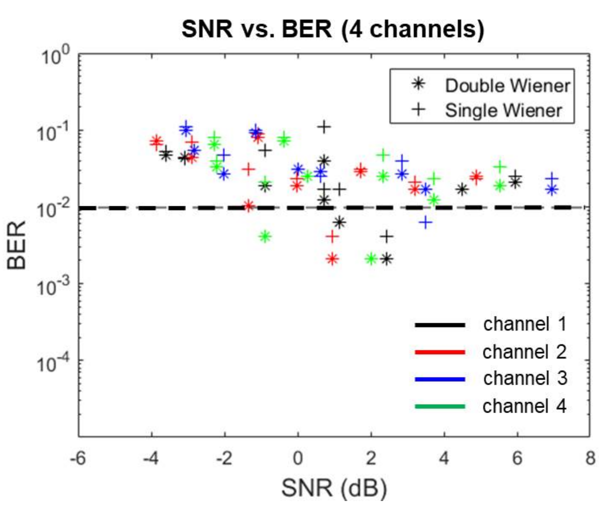

Figure 4: SNR vs. BER estimated for the four channels, independently, using both single and double Wiener filter approaches.

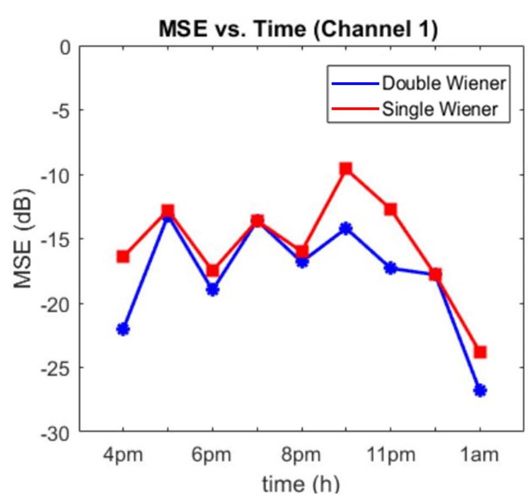

Figure 5: MSE vs. time estimated for channel #1 during the upwelling window, to compare the performance of the single and double Wiener filter.

Table 1: Average MSE difference (MSE$_{double}$−MSE$_{single}$) for the 4 channels (dB).

Channel	Average MSE difference (dB)
#1	-2.78
#2	-1.81
#3	-0.67
#4	-2.89

receivers, and to the time-varying noise levels in the region, related to man-made and biological factors. The BER is related to the SNR for both filters: the higher the SNR, the lower the BER. However, the double Wiener filter improved the single Wiener filter results, reducing the BER towards the 10^{-2} threshold in most cases, in all four channels.

To express this improvement in the system performance during the upwelling window, Fig. 5 shows the MSE estimated from channel 1, using both filters. Furthermore, in Table 1, one can observe the average MSE difference between the double and single Wiener filter, in dB, for all four channels. A factor that may have contributed to the achieved BER and MSE results is the severe multipath observed by each hydrophone. Even being separated from each other by just a few wavelengths, the hydrophones presented different multipath structures and noise levels. The hydrophone #3, positioned over the bottom presented a lower averaged MSE difference compared to the hydrophone #1 at the top of the array. But in all channels, the double Wiener filter outperformed the single Wiener filter.

5. CONCLUSION

In this paper, we presented a double Wiener filter approach for low SNR communications in a shallow water upwelling environment. The Wiener filter is an optimum linear filter, in the sense that it minimizes the mean square error. The proposed approach applies a Wiener filter over each received bitstream. Then, a second Wiener filter is estimated over the coherent averaged sequence to remove any residual intersymbol interference.

Based on a subset of data from BioCom'19 experiment, a time-series of water temperature close to the bottom was plotted. To observe the ocean upwelling impacts over the communication system performance, the CIR for three temperature profiles were plotted and compared. Results showed a large SNR variation (-3.9 to 7.3 dB) probably related to the ocean temperature stratification and to the time-varying noise in the region. To evaluate the performance, the four channels were processed independently. Despite BER fluctuation in time, the system proved robust, dealing with these short time scale signal fluctuations and high noise levels that hamper efforts to recover the message. Compared to a previous method based on a single Wiener filter, the double Wiener filter approach achieved better performance in all four channels, providing an average mean square error gain up to 2.8 dB.

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REFERENCES

[1] R. Diamant and L. Lampe, "Low Probability of Detection for Underwater Acoustic Communication: A Review," IEEE Access **6**, 19099–19112 (2018).

[2] D. Kilfoyle and A. Baggeroer, "The state of the art in underwater acoustic telemetry," IEEE J. Ocean. Eng. **25**(1), 4–27 (2000).

[3] M. Stojanovic, J. A. Catipovic, and J. G. Proakis, "Phase-coherent digital communications for underwater acoustic channels," IEEE J. Ocean. Eng. **19**(1), 100–111 (1994).

[4] T. Yang and W. B. Yang, "Low probability of detection underwater acoustic communications for mobile platforms," in *OCEANS 2008*, IEEE, Quebec City, QC, Canada (2008), pp. 1–6.

[5] B. Farhang-Boroujeny, *Adaptive filters: theory and applications*, 2nd ed. (Wiley, 2013), pp. 48–89.

[6] C. Ribeiro, M. de Campos, and P. Diniz, "A new approach for channel equalization using Wiener filtering," in *GLOBECOM'01. IEEE Global Telecommunications Conference*, IEEE, San Antonio, TX, USA (2001), pp. 290–294.

[7] R. Carvajal, K. Mahata, and J. C. Aguero, "Low Complexity Wiener Filtering in CDMA Systems Using a Class of Pseudo-Noise Spreading Codes," IEEE Communications Letters **16**(9), 1357–1360 (2012).

[8] J. Benesty, J. Chen, and Y. Huang, "Study of the widely linear Wiener filter for noise reduction," in *2010 IEEE International Conference on Acoustics, Speech and Signal Processing*, IEEE, Dallas, TX, USA (2010), pp. 205–208.

[9] F. B. Louza and H. A. DeFerrari, "Superimposed training low probability of detection underwater communications," J. Acoust. Soc. Am. **148**(3), EL273–EL278 (2020).

[10] A. Lempel and M. Cohn, "On fast m-sequence transform," IEEE Trans. Inf. Theory **23**(1), 135–137 (1977).

[11] F. B. Louza, J. Osowsky, F. C. Xavier, E. E. Vale, L. P. Maia, R. P. Vio, M. V. S. Simões, V. Barroso, and S. M. Jesus, "Communications and biological monitoring experiment in an upwelling environment at Cabo Frio island bay," in *OCEANS 2019 - Marseille* (2019), pp. 1–7.

[12] L. Calado, O. Rodriguez, G. Codato, and F. Xavier, "Upwelling regime off the Cabo Frio region in Brazil and impact on acoustic propagation," J. Acoust. Soc. Am. **143**(3), 174 (2018).

[13] H. S. Chang, "Detection of weak, broadband signals under doppler-scaled, multipath propagation," Ph.D. dissertation, Univ. Michigan, Michigan, MI, 1992.