Physics-Based Signal Processing Approaches for Underwater Acoustic Sensing

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Physics-based approaches can greatly improve the classification and localization of underwater sources in complex propagation environments.

Introduction

The undersea world is an important and fascinating domain, but it is also a very challenging environment for sensing. The dominant modality for underwater sensing is acoustics because electromagnetic and optical waves attenuate quickly in seawater. Fortunately, many sources of interest to humans, such as biologics (whales, dolphins, fishes) and man-made vehicles (submarines, ships), have distinct acoustic signatures that can be detected passively or actively. Sound radiated in the lower part of the acoustic band (i.e., 100s of hertz or below) can travel over fairly long distances, and these signals can be detected at significant ranges. However, the propagation over these distances can change the acoustic signature appreciably due to the physics of the underwater propagation, which introduces phenomena such as refraction and reflected multipaths.

Several of the most promising techniques in underwater signal processing used in oceanography are based on exploiting knowledge of the expected structure of the acoustic signature (in both the frequency and time domains). This structure is a result of the radiated source pattern (for passive detection) or the target reflection pattern (for active detection) and the modulation imposed by propagation through the acoustic channel. If the structure is known, a time-frequency filter can be designed to extract sources of interest. Furthermore, with an appropriate array aperture, a filter or beamformer can be constructed based on the expected spatial structure.

Here examples of physics-based spectral, temporal, and spatial filters are presented and demonstrated for biological and man-made sources in both shallow and deepwater environments.

Time and Frequency Structure

Acoustic signals can be represented in both the time and frequency domains. In the time domain, changes in acoustic amplitude correspond to the time-varying strength of the acoustic signal as measured at the receiver. As an example, consider **Figure 1, *left***, that shows the sound amplitude measured with a microphone of a voice recording from the song "That's Amore" over a five-second period. The amplitude clearly varies as the singer articulates, but the tonal components or the frequency content is also varying (i.e., the notes in the song). To extract the frequency signature, a short-term Fourier transform can be applied with a sliding window. The window needs to be small enough so that the assumption of stationarity is valid, that is, the frequency does not vary appreciably in that short window (Van Trees and Bell, 1968). This time-frequency output, defined as a spectrogram, is seen in **Figure 1, *right***. It can be seen that the frequency content is time varying, pertaining to the changing harmonic structure of the song.

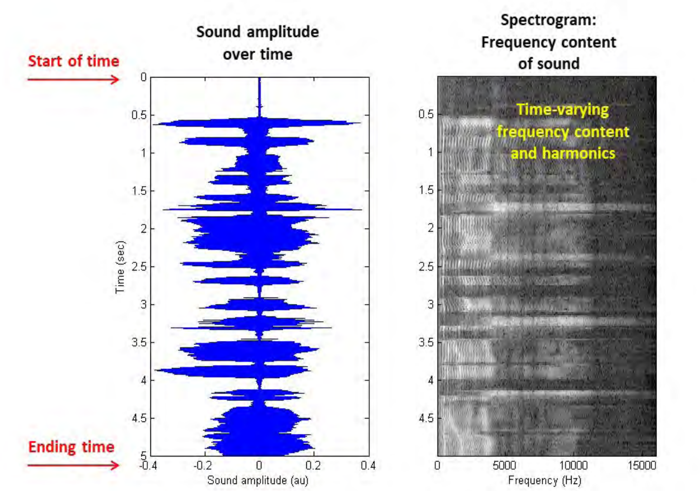

Figure 1. ***Left****: time-varying amplitude of a recorded sound.* ***Right****: frequency content of the same sound obtained by a taking a short-time Fourier transform with a sliding (and overlapping) window.*

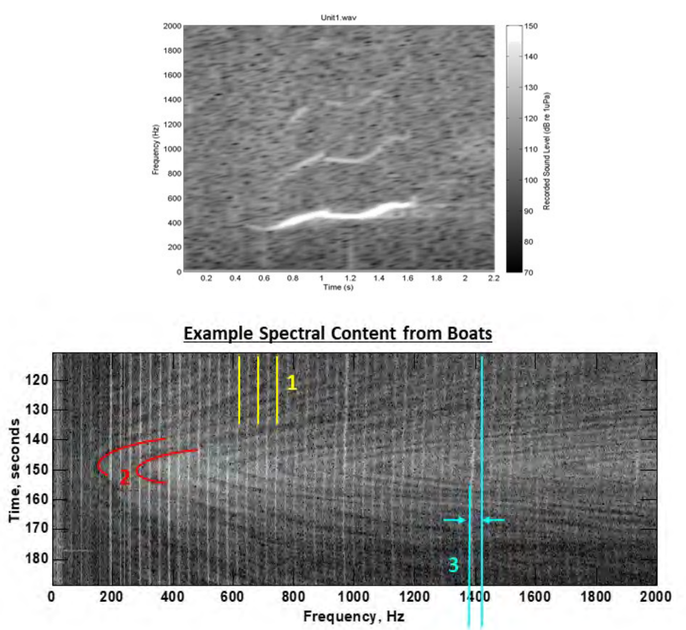

Figure 2. Spectrograms of sound recorded underwater showing the time-frequency structure of the received signal. ***Top****: spectrogram of a humpback whale sound recorded in the Hawaiian islands (Ou et al., 2013).* ***Bottom****: time-frequency signature of a surface vessel recorded in the Willamette River, OR. Note the structure introduced by the propagation physics (the "bathtub"). A time-frequency filter was designed by tracking the harmonics* ***(1, yellow****), the propagation modulation (****2, red****), and the Doppler shifts (****3, blue****) (Ogden et al., 2011).*

Both time domain and frequency domain representations are useful, depending on the nature of the acoustic signal. Often, the ability to characterize the source of a signal is dependent on a joint time-frequency signature. As an example, **Figure 2, *top***, shows the time-frequency signature of a humpback whale measured on a hydrophone in an underwater channel adjacent to the Hawaiian Islands (Ou et al., 2013). As seen in **Figure 2, *top***, the whale "song" consists of roughly four frequency components that change slowly over time, resulting in a chirp-like signal. The ability to distinguish biological signals from other noise (e.g., anthropogenic signals) depends on the ability to design joint time-frequency filters that are matched to the signal of interest. In Ou et al. (2013), this filter was implemented utilizing an image-processing technique that was then applied to the acoustic spectrogram.

The image in **Figure 2, *bottom***, is the signature of a surface vessel (ship). It was measured with a hydrophone deployed in the Willamette River in the state of Oregon after propagation and clear time-frequency structure can be observed. This structure is a result of both the radiation structure of the source (vessel) and the propagation modulation. The time-frequency filter is discussed later in this article.

Spatial Beamforming

Beamforming is a signal-processing technique that can be used to identify the direction of incoming (acoustic) energy. This is accomplished by using knowledge of the spatial structure of the incoming wave front as measured across the acoustic sensors deployed in an array. Although the processing can be quite complex, as described in the article by Wage in this issue of *Acoustics Today*, it is based on the same principle used in hearing to sense the direction of sound relative to the head. The delay in arrival of a sound from one ear to another is used by the brain to determine the likely direction of the source (i.e., "left" versus "right"). Determination of the location or direction of a source is defined as localization.

Consider, for example, the simple case of a line array where the elements are uniformly spaced (a uniform line array [ULA]). These types of arrays can be deployed horizontally as a horizontal line array (HLA), often towed from a surface platform (such as a ship), or as a vertical line array (VLA) tethered to the ocean bottom or another surface. Beamforming is the processing that would be used to localize the source of the energy.

To implement a beamformer and compute the likely arrival direction, the acoustic field measured on the array hydrophones is correlated with a vector of complex weights (the "weight vector") that represent the anticipated amplitude and phase structure of an incoming wave. Multiple weight vectors are computed, each one corresponding to a different possible arrival direction or steering direction (Van Trees and Bell, 1968). Under ideal circumstances, the peak of the

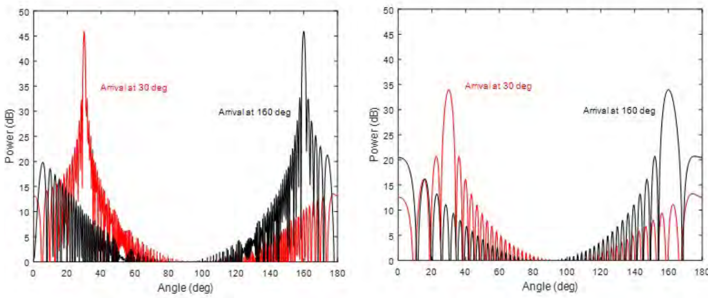

Figure 3. *Beamformer output for a plane wave incident on a uniform line array (ULA) at 30° (**red**) and 160° (**black**). Note that the beamformer output is maximized at the angle of arrival and decreases in other directions. The array used to compute the results on the **left** is 5 times as long as the array used for the results on the **right**.*

beamformer output will occur when the steering angle assumed for the weight vector is the same as the actual incoming wave direction.

As an example, consider the beamformer output seen in **Figure 3**, which shows simulated output from a ULA with interelement spacing of λ/2 (where λ is the acoustic wavelength) in response to a far-field point source at a specified angle. **Figure 3, *red curve***, shows the output of the beamformer for a source arriving at 30° relative to the array axis (0° and 180° corresponding to forward and rear endfire, respectively), and **Figure 3, *black curve***, shows a sound source at 160° relative to the HLA. If both sources were present simultaneously, the output would be the linear combination of the two curves.

The ability of an array to resolve the direction of arrival depends on the length of the array relative to the frequency of the sound source. For low-frequency underwater sources, such as shipping noise or seismic sources, very large arrays, potentially kilometers in length, are required. For higher frequency sources such as biologics (whales and dolphins), much shorter arrays can be used. In **Figure 3**, the output on the ***right*** is computed for an array that is one-fifth the length of the array used for the output on the ***left***. Note that the beam widths widen (i.e., loss of resolving power) and the output power drops.

Frequency also affects the sensing and localization performance by impacting the distance sound can travel underwater before being attenuated so greatly that it is undetectable. In general, low-frequency sound travels much further than the higher frequencies, making it a better choice for longer range-sensing applications.

Complex Propagation and Matched Field Processing

Unlike free space environments, in underwater channels, acoustic energy often does not travel only via a direct path from the sound source to the acoustic receiver. Instead, the sound reflects off the boundaries (the air-sea interface and the ocean-seabed interface) and is channeled to the receiver in what is called multipath propagation. **Figure 4** shows graphically how the air-sea and sea-sediment layer(s) can produce multiple propagation paths that can be received on an array, particularly in shallow water environments. These paths will potentially arrive at different angles and different times at the receiver array due to the variable path length distance. In addition to the channel boundaries, variations in the physical properties of the water column itself (as well as properties of the seabed) can also greatly impact the nature of the sound propagation. These variations are typically described in terms of the sound speed profile, which quantifies the change in acoustic wave speed as a function of depth and density. Sound speed profiles can change dramatically over time, including both diurnal cycles and seasonal variations, and can also vary along the wave path (Urick, 1983).

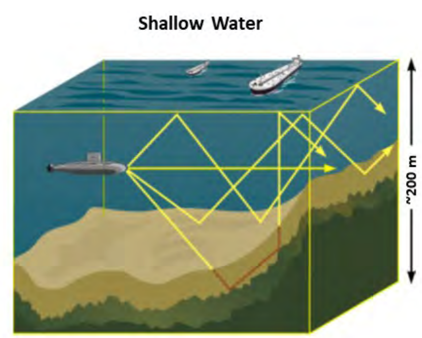

Figure 4. *Graphical representation of the multipath propagation for an acoustic source (submarine) in an underwater channel. The sounds are reflected from boundaries (such as the air-sea interface or sediment layers) and is refracted by gradients in the sound speed profile (not shown).*

An example of the signal measured in a multipath environment is shown in **Figure 2, *bottom***. These data were obtained in the Willamette River with a hydrophone in the presence of a passing ship. The ship propeller produces multiple spectral harmonics that depend on the propeller specifics (i.e., the number of spokes, rotation rate), the multipath propagation, and the Doppler introduced by the motion of the boat relative to the receiver. **Figure 2, *curved lines***, shows the "bathtub" pattern that is produced by a ship approaching the hydrophone, nearing the closest point of arrival (or CPA), and then moving away from the receiver. In this case, the correct classification of the vessel is obtained by employing a multifrequency filter that is designed to detect (1) the ship harmonics, (2) the propagation-induced bathtub structure, and (3) the Doppler shift introduced by the vessel velocity

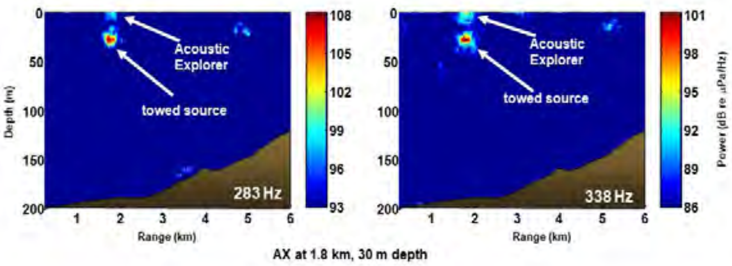

*Figure 5. Matched field processing ambiguity surfaces computed for data from the Santa Barbara (CA) Channel Experiment (SBCX) at 283 Hz (**left**) and 338 Hz (**right**). The surfaces represent ambiguity surfaces in range and depth and show the towed source and the tow vessel (*Acoustic Explorer *[AX]) localized at the correct location (Zurk et al., 2003).*

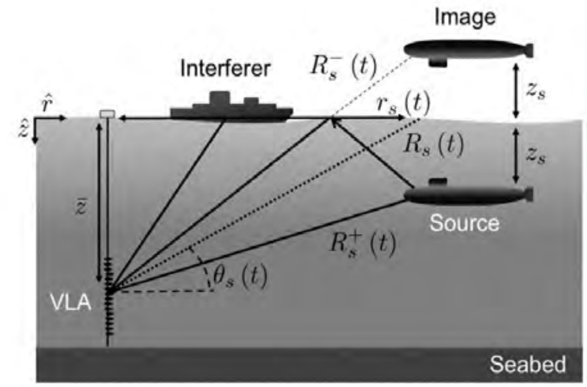

Figure 6. Graphic showing the propagation paths as predicted with image theory for a submerged source (submarine) and a surface interferer as the signal received on a deeply deployed vertical line array (VLA). While the submerged source has two paths that interfere coherently and cause an interference pattern, the surface source has only a direct path.

(Ogden et al., 2011). The filter was implemented with a time-varying Kalman filter that was based on the radiation and propagation physics.

With an array aperture, the changing arrival angle of the received can be measured and correlated with a model of the underwater propagation structure. This approach is called matched field processing (MFP) and involves the use of physics-based understanding to optimize the beamforming process (Baggeroer et al., 1991). In MFP, the weight vector is computed using an underwater propagation model, such as normal mode theory or ray-based propagation. These calculations require knowledge of the physical properties of the underwater channel (i.e., the depth-dependent sound speed, water depth, and sediment composition) to compute a frequency-dependent space-time filter.

Application of a MFP filter achieves not only classification of the source but also localization with a greater resolution than inherently possible with the array aperture. An example of MFP output is shown as range-depth surfaces, defined as ambiguity surfaces, in **Figure 5** for two different tonal frequencies using data measured on a VLA deployed in the Santa Barbara (California) Channel Experiment (SBCX; Zurk et. al., 2003). The interpretation of the plots is that the “hot spots” (or areas with colors tending toward the red part of the color spectrum) in the diagram represent spatial regions that have a high probability of containing an acoustic source. In the SBCX, an acoustic projector was towed from a research vessel (the *Acoustic Explorer*), and the sound from both the projector and the tow ship itself are clearly identifiable in **Figure 5**.

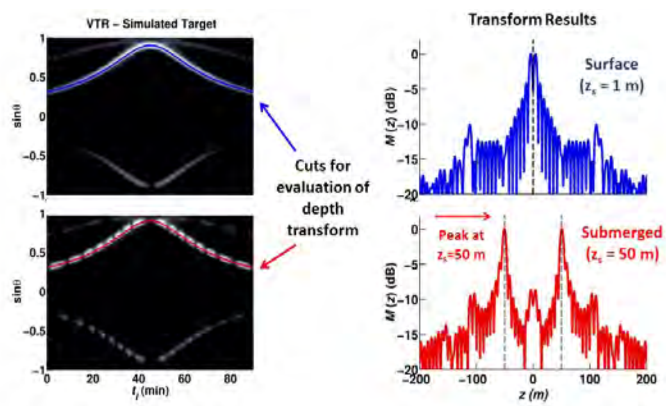

*Figure 7. Simulated results for a deepwater robust acoustic path (RAP) geometry with either a surface source (**top, blue**) or source submerged at 50 m in depth (**bottom, red**). **Left**: vertical time record (VTR), with the elevation angle varying over time as the sources move. **Right**: harmonic content calculated at time-varying angles. The peaks on the **right** appear at the correct depth for each of the sources (McCargar and Zurk, 2013).*

Depth Estimation in Deepwater

The MFP approach generated considerable interest and exploration in the research community, but performance in actual practice suffered because the accuracy of the output depended critically on having detailed knowledge of the underwater channel, which was generally not available. More recently, approaches that are robust to environmental mismatch are being considered. One example is an approach applicable to geometries with a shallow source and a VLA deployed below the critical depth in deepwater environments. This propagation scenario is called robust acoustic path (RAP) environments (Urick, 1983). In this geometry, the multipath is much simplified and primarily consisted of a direct and surface-bounce signal with little refraction from the sound column. Under these circumstances, no knowledge of the ocean environment is required and a simple image theory calculation can be used, as seen in **Figure 6**.

From a mathematical standpoint, the coherent addition of the two multipath signals at the receiver as the source transits in range can be represented as a time-varying signal. The frequency of the interference between the direct and surface bounce can be directly related to the source depth, and a modified Fourier transform can be applied to the output of a VLA over time (this is defined as a vertical time record [VTR]). As an example, a simulation of the VTR for a source radiating energy at 150 Hz in a 5,000-meter channel is shown in **Figure 7, *left***, for a surface (***top***) and submerged (***bottom***) source. Note that the incident angle measured on the VLA changes over time as the source transits in the vicinity of the array. **Figure 7, *right***, shows the transform output as a function of source depth, and the peaks correctly appear at the source depth (McCargar and Zurk, 2013), with the peaks for the surface source appearing at 1 m in depth (***top, blue***) and at 50 m for the submerged source (***bottom, red***).

Conclusions

Acoustic sensing in underwater channels is quite challenging due to the complicated structure of many sources (both passive radiation and active reflected) and the propagation physics. Several techniques have been devised to exploit this temporal, spectral, and spatial structure, and application of these algorithms can lead to enhanced detection, classification, and localization of underwater sources. In this article, a short introduction to physics-based signal processing has been provided with examples from the research literature applied to biological and man-made sources in both shallow and deepwater environments.

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BioSketch

Lisa Zurk is the executive director of the Applied Physics Laboratory at the University of Washington (UW), Seattle, and a professor in the Electrical Engineering Department at UW. Previously, she served as a program manager at the Defense Advanced Projects Research Agency, as a professor of electrical and computer engineering at Portland State University, OR, and as an associate group leader at the MIT Lincoln Laboratory. She is the author of over 50 technical papers and has received multiple recognitions, including the Presidential Early Career Award for Scientists and Engineers, the National Science Foundation CAREER Award, a Fulbright scholarship, and the Office of Naval Research Early Faculty Award.