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Source characterization using recordings made in a reverberant underwater channel

Kay L. Gemba *, Eva-Marie Nosal

Department of Ocean and Resources Engineering, School of Ocean and Earth Science and Technology, University of Hawai'i at Mānoa, Honolulu, HI 96822, United States

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The ability to accurately characterize an underwater sound source is an important prerequisite for many applications including detection, classification, monitoring and mitigation. Unfortunately, anechoic underwater recording environments required to make ideal recordings are generally not available. This paper presents a practical approach to source characterization when working in an imperfect recording environment; the source spectrum is obtained by equalizing the recording with the inverse of the channel's impulse response (IR). An experiment was conducted in a diving well (depth of 5.18 m) using a logarithmic chirp to obtain the IR. IR length is estimated using methods borrowed from room acoustics and inversion of non-minimum phase IR is accomplished separately in the time and frequency domain to allow for a direct comparison. Results indicate that the energy of controlled sources can be recovered with root-mean-square error of −70 dB (10–70 kHz band). Two equations, one coherent and the other incoherent, are presented to calculate source spectral levels of an unknown source in a reverberant environment. This paper introduces a practical procedure outlining steps to obtain an anechoic estimate of an unknown source using equipment generally available in an acoustic laboratory.

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1. Introduction

Underwater source characterization is important for numerous applications. For example, passive acoustic detection and classification can be improved by knowledge of the sound characteristics of the object of interest. With knowledge of the source, array configuration and specifications can be optimized for monitoring. As another example, environmental compliance laws regulate an environment by putting limits on emitted acoustic energy, so that a sound source needs to be well understood before being used in the environment. Unfortunately, anechoic underwater recording environments required to make ideal recordings are generally not available or are cost-prohibitive.

An anechoic recording contains the direct arrival of acoustic energy from a source to the hydrophone with minimal noise or wall reflections. Sound levels estimated from recordings made in a reverberant environment (such as a test tank or pool) generally overestimate source levels due to additional wall reflections and noise. It was found [4,9] that the acoustic power of a source can be separated from reverberant energies by measuring the spectral pressure at one or more random locations in a reverberant enclosure (yielding spatial mean spectral levels). Recordings must be conducted in the far field of the source, e.g., the hydrophone is placed within the homogeneous and isotropic reverberant field. An estimate of the source is obtained by adjusting recorded levels with calculated reverberant energies. The reported error for a 100 Hz broadband white noise source [4] is ~1.5 dB and expected vs. calculated spectral levels for pure sinusoids differ by 0.1–5.8 dB. This approach provides an economic way [9] to estimate source power but is inherently limited to an incoherent estimate. To our knowledge, no other approaches exist for characterization of sound sources in underwater reverberant environments. Here, we follow a different ansatz using methods borrowed from room acoustics to estimate and invert the recording channel.

The recorded signal is the convolution of the source signal with the impulse response (IR) of the channel, hence, in principle, convolving the recorded signal with the inverse of the IR equalizes the channel (see Section 2), yielding an anechoic estimate of the source signal (Ref. [27] serves as an excellent introduction to deconvolution). The acoustic IR can be estimated with an excitation signal and by convolving an inverse filter with the received signal [24,6]. Theoretically, using an impulsive excitation signal is the preferred way to estimate the IR since an impulse freezes the system under investigation in time. In practice, when the test device is not purely electrical but has an acoustic path in the

* Corresponding author.
E-mail address: gemba@hawaii.edu (K.L. Gemba).

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measurement chain, this procedure has to be adjusted because the transmitting transducer cannot realize an impulse. The excitation signal is selected and pre-colored to maximize signal to noise ratio (SNR) and the recorded signal reflects the states of the system over the playback duration. Popular signals include periodic signals such as maximum length sequences (MLS) and non-periodic signals such as linear or logarithmic sweeps. Once the IR is deconvolved and its length is estimated (see Section 3), it can be inverted.

Three primary methods have been investigated in room acoustic literature to coherently invert an acoustic IR: homomorphic deconvolution [31,21,25], single channel least squares (SCLS, a time domain method) [35,12,18], and inversion in the frequency domain [10]. In principle, homomorphic deconvolution is attractive because deconvolution of minimum phase signals in the time domain is division in the frequency domain and subtraction in the cepstrum domain [23]. However, non-minimum phase signals have cepstral overlap and the direct arrival cannot be easily separated from early reflections. It was found [21] that an IR has minimum phase only if the wall reflectivity coefficient is small enough (below approximately 0.4), otherwise its inverse will be acasual or unstable. The problem in room and underwater acoustics is the same: the IR is of non-minimum phase if partial energies (in the time domain) are not strictly decreasing. This is clearly the case for late reflections from a high impedance boundary (such as water–air). In addition, spectral zeros of the IR result in narrow band noise amplification and direct inversion is not desirable.

SCLS can address this problem and has been found to be more practical than homomorphic deconvolution [18]. The inverse of a mixed-phase IR in the least-squares sense can be significantly improved using a processing delay [21,17,3] to render it causal and improve stability. Even though only approximate equalization can be achieved [16], SCLS is robust to measurement noise and only partially equalizes deep spectral nulls [20], hence reducing narrow band noise amplification after equalization. In addition, it can easily evolve into a multi-channel method [16]. Ill-conditioned inverse problems can also efficiently be solved and regularized in the frequency domain [33,10]. The inverse filter is loaded with a small frequency dependent constant to improve the inversion. In this paper, we use and compare SCLS and frequency domain inversion.

In a preliminary experiment [8], a linear sweep was used to estimate the IR of an underwater reverberant recording channel. The inverted IR was used to remove reverberation effects to approximate source spectral levels (SSL) of a recorded SCUBA diver over an appropriate band. A follow up experiment was conducted to investigate and quantify dereverberation performance using control sources; these results are presented in the following sections.

This paper presents a practical procedure for underwater acoustic experimentation to recover an anechoic estimate of a source recorded in a reverberant environment. It is structured as follows: First, the problem is formulated in Section 2 and Section 3 describes the proposed experimental procedure which is validated by an experiment (Section 4). Methodology of data analysis is presented in Section 5 followed by results in Section 6. The paper concludes with a discussion in Section 7.

2. Mathematical formulation

Fig. 1(a) shows a diagram of the inverse problem in an underwater recording environment. The recording process can be modeled as the convolution of individual IRs. Here, the input signal of the source $d_i(t)$ is recorded in a reverberant channel $g(t)$ with a hydrophone $r_2(t)$. The hydrophone is connected to an analog to digital converter (ADC, denoted by $r_1(t)$) and the recorded output signal $d_o(t)$ is stored on a hard drive:

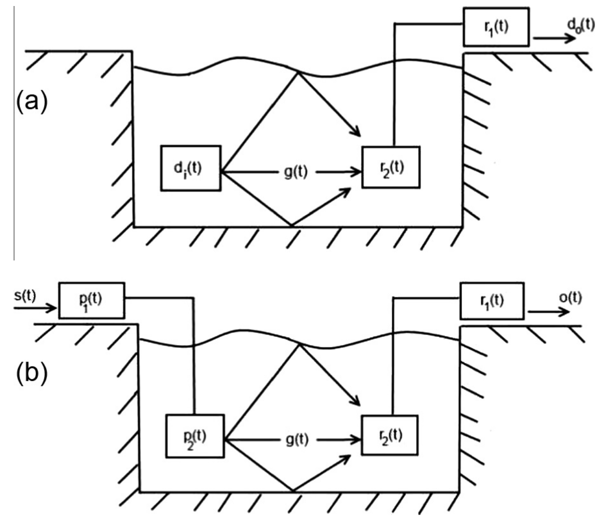

Fig. 1. Pool diagram showing schematics of (a) the inverse problem with an unknown source and (b) the forward problem with a known source.

$$d_o(t) = r_1(t) * r_2(t) * g(t) * d_i(t). \tag{1}$$

The problem of interest here is to estimate the input signal which is not immediately possible since both the source and the IR of the channel are unknown. To estimate the IR, the source signal is replaced by a known signal, shown in Fig. 1(b). For the forward problem, the source signal $s(t)$ is fed through a playback and pre-amp device $p_1(t)$ which is connected to a transmitting transducer $p_2(t)$. It is assumed that both the unknown source and the transmitting transducer have similar directionality and are of similar shape. The channel and the recording equipment is the same as in the inverse problem and the recorded signal is denoted by $o(t)$. For convenience in the rest of this paper, the total IR combining the playback and recording devices with the channel is abbreviated by the filter $h(t)$ (Eq. (2)).

$$h(t) = r_1(t) * r_2(t) * g(t) * p_2(t) * p_1(t) \tag{2}$$
$$o(t) = h(t) * s(t) \tag{3}$$

Our first task is to identify the IR of the system $h(t)$ which is convolved with the input $s(t)$ to the system to produce output $o(t)$ (Eq. (3)). Since the pool remains unchanged except for random fluctuations due to pool pumps and outside disturbances (such as wind), we assume that the resulting channel is an ergodic stochastic system. If we further assume that the in-phase and quadrature components of both amplitude and phase each have Gaussian distributions, the sinusoidal pressure in the channel follows a Rayleigh distribution [13] which is a function of absorption coefficient α_i, combined surface area (A_i) of the walls and water surface, and distance (r) from the source to the hydrophone. The 68% range of the sinusoidal sound pressure level (SPL) distribution (corresponding to approximately one standard deviation (SD), denoted by σ) was derived in [5] and is a linear approximation between σ and r for a poorly reverberant enclosure (this is where the constant in Eq. (4) comes from). Here, the original equation is slightly modified to average over six non-uniform absorption coefficients, corresponding to the boundaries of a rectangular enclosure:

$$\pm\sigma \approx 40r\left(1 - \sum_{i=1}^{6}\frac{\alpha_i}{6}\right)^{\frac{1}{2}}\left(\sum_{i=1}^{6}\alpha_i A_i\right)^{-\frac{1}{2}} \mathrm{dB}. \tag{4}$$

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