Emulation System for Underwater Acoustic Channel

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Abstract
Mathematical models describing acoustic underwater channels, cannot accurately predict the channel’s behavior. To find a channel’s response to a specific signal, sea trials need to be conducted. This paper presents a tool to Emulate Underwater Acoustic Channel (EUAC) by evaluating its time varying impulse response. The tool allows estimation of the channel’s output for any given signal communication scheme without the need for a sea trial, thus saving time and resources. The initial procedure requires a set of sea trials. In each experiment specific signals with narrow auto-correlation property are transmitted, and then their responses are recorded. This allows impulse response, Doppler shift, and phase shift estimation of the actual channel. The results of the experiments can then be used to create a database of various EUAC. This database may be useful in evaluating the performance of various communication schemes without sea trials.

1. Introduction
This paper describes a tool for measuring and emulating the underwater acoustic channel, which can be used to build a database of emulated channels. The Emulated Channel Response (ECR) to a given signal is found to be typically highly correlated (more than 80%) with the response of the actual channel.

A waveform suitable for measuring the channel impulse response is one whose auto-correlation function is as close as possible to an impulse. This implies the use of a test signal with a wide as possible bandwidth. To increase the energy of the test waveform over that of a very short simple pulse, a waveform with a high time-bandwidth product is used.

To EUAC for a given signal, it is assumed that the channel is a Linear Time Invariant (LTI) system. Therefore, all non-linear and time varying features of the channel must be evaluated separately and corrected before evaluating the impulse response. These features are added later to the emulated signal.

This paper presents a scheme for channel emulation composed of two stages: (i) Impulse response and channel characteristics evaluation, emulation process and database handling. (ii) Choosing and checking an emulated channel from a database. Then Passing a given signal using the chosen emulated channel and noise addition in any desired Signal to Noise Ratio (SNR).

2. Characteristics of underwater acoustic channels
The underwater acoustic channel is a time-varying frequency selective spatially uncorrelated channel with additive colored Gaussian noise. It is characterized by frequency dependent and range dependent absorption, which together with the multipath phenomenon results in fading. In the following sections some characteristics of the underwater channel are described.

2.1. Doppler shift
A relative movement of the receiver and transmitter or a moving medium (in the case of a non-negligible current) can change the frequency of the sound waves propagating through the channel. This apparent change in the signal’s carrier frequency and time domain is known as Doppler shift.

Assuming the relative speed of the source and the observer (v) is much smaller than the speed of sound (c), the observed frequency of the sound wave is given by[1]:

\[ f' = \frac{v + c}{c} f \] (1)

where, \( f \) is the transmitted frequency. The Doppler shift effect causes the transmitted signal’s length (in the time domain) to shrink or expand. The received signal duration is:

\[ T_s' = T_s \frac{f}{f - f'} \] (2)

where \( T_s \) is the transmitted signal duration.

2.2. Multipath
The multipath is mainly caused by reflections from the sea floor and surface, the number of signal bounces determines the multipath spread. Furthermore, the channel consists of volume reflectors such as plankton and fish. Assuming that the transmitter-receiver’s range is large enough, the signal propagates from the transmitter to the
receiver via various paths. The delay of each path depends on its geometry. The channel’s impulse response is modeled as:

\[ A(\tau, t) = \sum_{k=0}^{L} A_k(t) \delta(\tau - kT_c) \quad (3) \]

where \( A_k(t) \) represent the power loss and phase shift of each path, \( L \) is the number of effective paths and \( T_c \) is the minimum path delay. The cross power density function of the channel is defined as:

\[ \frac{1}{2} E[A^*(\tau, t)A(\eta, t + \Delta t)] = Q(\tau, \Delta t)\delta(\eta - \tau) \quad (4) \]

where the Multipath Intensity Profile (MIP) is \( Q(\tau, \Delta t = 0) \), and the delay spread \( T_m \) is defined as the time gap where \( Q(\tau, \Delta t = 0) \neq 0 \). The MIP of the underwater channel is modeled as [2]:

\[ Q(\tau) = \frac{\sigma_e^2 e^{- \tau / T_m}}{T_m} \quad (5) \]

The MIP function fulfills the condition \( Q(nT_c + \epsilon) = Q(nT_c) \) with \( 0 < \epsilon < T_c \), hence the \( A_k(t) \) components are modeled as gaussian random processes with variance of \( \sigma^2_k = T_cQ(nT_c) \). The delay spread is related to the coherence bandwidth using: \( \Delta f_c \propto \frac{1}{T_m} \). When \( \Delta f_c \) is small in comparison to the bandwidth of the transmitted signal, the channel is said to be frequency-selective.

### 2.3. Doppler Spread

The Doppler spread \( B_w \), expresses the spectral width spreading of the received signal. The coherence time of the channel relates to the Doppler spread \( \Delta T_c = \frac{1}{\Delta f_c} \). In shallow water the reflections from the water surface are the primary reason for the time-variance of the channel. Wave movements are the primary cause of the spreading of the reflections from the water surface and therefore causing the Doppler spread. The value of the Doppler spread depends on the waves height and frequency, wind speed, number of reflections from sea surface and floor, and the nominal impact angle.

### 2.4. Channel noise

The channel noise is assumed to be an additive colored gaussian ambient noise, with frequency response of:

\[ |P(f)| = k_0 f^{-2} \quad (6) \]

where \( k_0 \) is an empiric constant, which is frequency band and sea state dependent. In low frequency (below 1KHz) the dominant noise component is of distant shipping noise, in mid band - wind related noise and in high band mostly thermal noise[3]. One may whiten the noise at the receiver and transmitter.

### 2.5. Transmission loss

The power of an acoustic wave passing through the channel is reduced due to absorption loss and scattering loss. The loss can be modeled as:

\[ TL(r, f) = k \times \log_{10}(r) + \alpha(f)r[\text{dB}] \quad (7) \]

where \( k \) ranges from 10 to 20, \( r[\text{m}] \) is the transmission range, and \( \alpha \) is absorption parameter that depends on the transmission wave’s carrier frequency[1].

### 3. Evaluating channel characteristics

The channel impulse response can be estimated by transmitting a signal that has a narrow auto-correlation, which is as close as possible to an impulse: \( R[s(t)] = \int s(\tau)s(\tau + \tau)d\tau \sim \delta(\tau) \) where \( s(t) \) is the transmitted signal. Thus, assuming LTI channel

\[ s(t) \otimes h(t) \otimes s(t) = R[s(t)] \otimes h(t) \approx h(t) \quad (8) \]

Where \( h(t) \) is the channel impulse response and the \( \otimes \) operation denotes the convolution operation.

#### 3.1. Transmitted signals block

A signal’s auto-correlation (or matched filter) main lobe width is determined by[4]:

\[ \text{MF}_{\text{width}} = 1/\text{BW} \quad (9) \]

where BW denotes the bandwidth of the signal. Hence, the transmitted signals should be wide band signals. The transmitted signals should also have immunity for frequency selective channels in order to ensure (8). Such signals are the Direct Sequence Spread Spectrum (DSSS) signals[5]:

\[ s(t) = \sqrt{2P}d_{n,N+1}, \quad nT_s + \ell T_c \leq t \leq nT_s + (\ell + 1)T_c \quad (10) \]

where \( P \) is the signal’s power, \( d_n \) is the pseudo-random sequence consisting of \( N \) chips and \( T_c \) is the chip’s duration. To evaluate the time varying impulse response, a block of \( M \) DSSS signals is transmitted. The EUAC will therefore be suitable for signals with bandwidth smaller than \( \frac{1}{T_c} \).

#### 3.1.1. Synchronization signals

The DSSS signal’s matched filter output is vulnerable for Doppler shift. Therefore, a synchronization signal needs to be transmitted prior to the DSSS block. One may do so by using a wide band signals immuned for Doppler shift. Such signals are the Chirp signals. By transmitting an ‘up’ chirp and ‘down’ chirp (i.e a chirp that has a frequency slope that is positive and negative respectively), the Doppler shift can be estimated.
3.1.2. DSSS synchronization

The time shift in the synchronization is handled via chip synchronization of the DSSS signal. The synchronization placement is found where the output of the despreading process (i.e., multiplication with the spreader DSSS sequence) is a narrow band signal (detected using spectral analysis operations).

Since the received symbol is Doppler shifted, it is necessary to estimate the Doppler shift value (see section 3.2) prior to the de-spreading operation in order to compensate the samples set’s Doppler shift.

A single carrier signal (CW) at carrier frequency equals to that of the DSSS signal is transmitted after the DSSS block to evaluate the Coherence Time (CT) of the channel.

3.2. Doppler shift estimation

The Doppler shift estimation is performed at the synchronization signals and at the DSSS block.

3.2.1. Doppler shift estimation for chirp signals

The received signal is passed through two matched filters: one for the ‘up chirp’ and one for the ‘down chirp’. Each matched filter output consists of one major peak for each chirp received. It can be shown that the Doppler shift is related to the difference in the position of the peaks in the time domain by:

\[ \Delta f = \frac{\Delta T \times BW}{T_s} \]  

(11)

where \( \Delta T \) is the difference between the peaks position for the ‘up’ and ‘down chirp’ matched filter outputs respectively. To minimize timing errors the \( \Delta T \) value is measured using central mass evaluation of the matched filters output.

3.2.2. Doppler shift estimation for DSSS signals

Using spectral analysis on chip synchronized de-spreaded DSSS symbol yields the Doppler shift estimate.

\[ \Delta f = f' - f \]  

(12)

To mitigate estimation errors the evaluated Doppler shift vector is smoothed by a best fitted suitable polynomial. In Figure 1 a vector of evaluated Doppler shift and its polynomial fitting is presented.

3.3. Coherence time estimation

Better results for the channel estimation are expected using signals the duration of which is shorter than the CT of the channel. Assuming that the CT value does not change significantly during the transmission, it is possible to evaluate it using the CW signal.

The normalized matched filter between vectors \( x, y \) is defined as:

\[ \text{NMF} = \frac{\sum x \cdot y - N \bar{x} \cdot \bar{y}}{\sqrt{\left( \sum x^2 - N \bar{x}^2 \right) \left( \sum y^2 - N \bar{y}^2 \right)}} \]  

(13)

where \( N \) is the symbol duration [samples], and \( \bar{x} \) is the mean value of \( x \). Dividing the CW signal to sub-signals and using the first sub-signal as reference, a vector of normalized matched filter peak outputs is obtained. The CT is taken to be the duration that the normalized matched filter of the sub-signals with the reference signal is above a certain threshold.

The DSSS symbols are then divided into sub-symbols with duration shorter than \( \Delta t_c \).

3.4. Impulse Response Matrix (IRM) evaluation

Each DSSS symbol is chip synchronized after Doppler shift correction using interpolation techniques. A cross-correlation operation on each of the received DSSS symbols using the transmitted symbol is performed (divided into sub-signals as in section 3.3). Assuming an accurate compensation of the Doppler shift and assuming a signals duration shorter than the CT, the received output is the channel’s evaluated impulse response for the current symbol. A matrix of such sequential impulse responses represents the time varying impulse response of the channel. An example for such a matrix obtained from a shallow water acoustic channel in the Mediterranean Sea is presented in Figure 2.

The horizontal axis represents the time of a single impulse response. The vertical axis represents the time varying of the channel. From the figure one can observe 3 main paths that fade and strengthen during the matrix duration. A single impulse response from the matrix presented is shown in Figure 3.

Figure 1: Modeled Doppler shift vector
The impulse response matrix is shifted to the carrier frequency in which it was measured. The given signal that is emulated should also be at the same frequency range, otherwise the emulated channel response is not valid. A 2-D convolution operation is performed between the given signal and the emulated IRM. Then, the signal is re-sampled according to the modeled Doppler shift. The output of these operations yields the ECR of the signal.

4.1. Expanding the IRM

In case where the signal is longer than the evaluated IRM duration, the impulse response matrix needs to be expanded. One method to do so is via periodic expansion of IRM. The cycle of the periodic expansion is extracted from the maximum of 2-D cross-correlation of the IRM. Results of more than 90% cross-correlation were received in sea trials.

A better way is to model the IRM as a Markov process. In this model each row is a function of the last state and a set of current state parameters. Accurate results are expected for current state values of wind speed, wave cycle, noise level, wave height, multipath number, and nominal impact angle at the receiver.

4.2. Expanding the Doppler shift vector

The modeled Doppler shift vector needs to be expanded as well in order to fit the given signal, since the polynomial fitting is not suitable for periodic expansion. It is assumed that the Doppler shift is composed from a DC component (the relative speed between the transmitter and receiver) and an AC component (which is dependent on the wave cycle). Assuming the cycle is a function of two frequencies - swell wave and second degree waves, a two frequency sinusoidal wave is evaluated from the Doppler shift vector samples. This is done using the Pisarenko Harmonic Decomposition method [6].

4.3. Transmission loss evaluation

The transmission loss of the received signal is evaluated by measuring the source level (SL) of the transmitted signal (using a monitoring receiver at the transmitter), and measuring the received level (RL) at the receiver. The emulated signal’s power is multiplied by the evaluated transmission loss.

4.4. Ambient noise addition

There are two alternative ways to add noise to the channel to emulate various SNR’s:

1. Modeling the noise (see section 2.4) and setting the noise level according to the desired SNR.
2. Measure actual noise from the channel, and periodically expanding it in case the signal duration is larger than of the measured noise.

5. Results

The emulation system credibility can be evaluated by comparing a signal received from actual channel, and the corresponding ECR signal. In Figure 4 the given signal is the same signal used to create the IRM and Doppler shift vector. The comparison is performed between the outputs of each system.

In Figure 5 comparison results for a shallow water channel in the time domain are presented. One can see that a cross-correlation coefficient of approximately 80% is achieved. These values were typical for sea trials performed on a variety of acoustic underwater channels.

6. Conclusions

An underwater acoustic emulation system was presented. The system consists of time varying IRM evaluation, channel affects compensation and modeling, and expansion of the evaluated impulse response and Doppler shift.
An algorithm for emulating given signals was presented, achieving 80% credibility comparing to real measured underwater channel.

By building a database that consists of several IRM and Doppler shift vectors, a useful tool for extensive evaluation of underwater communication algorithms is available.

7. Acknowledgments

Helpful discussions with G. Caspari, M. Doron, K. Buchris, R. Melitz and U. Nahir are gratefully acknowledged.

8. References