Signal Processing for Underwater Acoustic Communications

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ABSTRACT

The performance and complexity of signal processing systems for underwater acoustic communications has dramatically increased over the last two decades. With its origins in noncoherent modulation and detection for communication at rates under 100 b/s, phase-coherent digital communication systems employing multichannel adaptive equalization with explicit symbol-timing and phase tracking are being deployed in commercial and military systems, enabling rates in excess of 10 kb/s. Research systems have been shown to further dramatically increase performance through the use of spatial multiplexing. Iterative equalization and decoding has also proven to be an enabling technology for dramatically enhancing the robustness of such systems. This article provides a brief overview of signal processing methods and advances in underwater acoustic communications, discussing both single-carrier and emerging multicarrier methods, along with iterative decoding and spatial multiplexing methods.

INTRODUCTION

The last two decades have brought about great advances in the research, development, and deployment of underwater acoustic digital communications systems. An area that was once of interest primarily for military and deep sea research applications has ballooned into a rich field with great potential, made possible in large part by a 10,000-fold increase in achievable data rates over the last few decades. In environments that once admitted lumbering single-digit to 100 b/s links, commercially available systems running in small-form-factor submersible buoys can achieve data rates in excess of 10 kb/s [1, 2].

Such untethered systems are of increasing interest in a wide variety of applications, such as commercial fishing and oil exploration, where remotely controlled vehicles and equipment are used to probe, sense, and actuate apparatus from a surface vehicle or station. With increasing interest in environmental sensing and wildlife monitoring and tracking, as well as continued exploration of the potential for research, commercial, and scientific applications in the oceans and shorelines of the world, the ready availability of high-rate digital acoustic communications systems has become a catalyst for an explosion of applications. These applications range from command and control links to submarines and autonomous underwater vehicles in research, military, and search-and-rescue contexts, to remote operation and control of sensing equipment in deep sea fishing, off-shore oil exploration, and environmental monitoring.

This article is meant as an introduction to the area of underwater acoustic communications for the greater signal processing and communications research communities. While these communities have shown intense interest in both wireline and wireless communications over the last several decades, relatively little attention has been paid in this research literature to the underwater acoustic realm. A great deal of research studying underwater communications has been published in the oceanic engineering and underwater acoustics literature.

In a short article such as this, it is neither possible nor our intention to cite all relevant literature on the topic. This mission of this article is simply to provide an introduction to the tremendously rich and exciting field of underwater acoustic communications and to note that, as data rates increase in mobile wireless and cellular networks, many of the challenges and solutions currently considered unique to the underwater acoustic environment may become highly relevant to wireless communications in general. We urge interested readers to look not only to the references listed here, but also to the references therein and apologize in advance to our many colleagues whose work we have not been able to mention.

In radio frequency (RF) communications, information is transmitted in the form of electromagnetic waves. The information bearing signals are typically composed of one or a number of sinusoidal components that have been modulated in amplitude and phase, resulting in either a single-carrier or multicarrier modulated signal. Electromagnetic waves do not propagate over long distances through the ocean, however. The salinity of sea water induces conductivity, which

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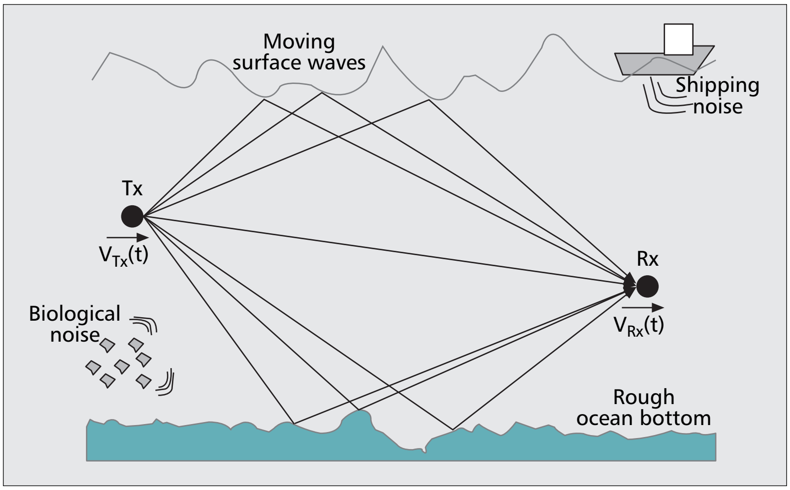
results in rapid attenuation of electro-magnetic signals, especially at higher frequencies. The signals used to carry information through an underwater communication channel are acoustic (pressure) waves, which can propagate over long (and in some cases, very long) distances. As discussed in the article by Stojanovic and Preisig in this issue, the underwater acoustic channel presents a number of unique challenges for the design of high-data-rate digital communication systems.

The ocean environment characteristics that drive the complexity of underwater acoustic communications systems include:

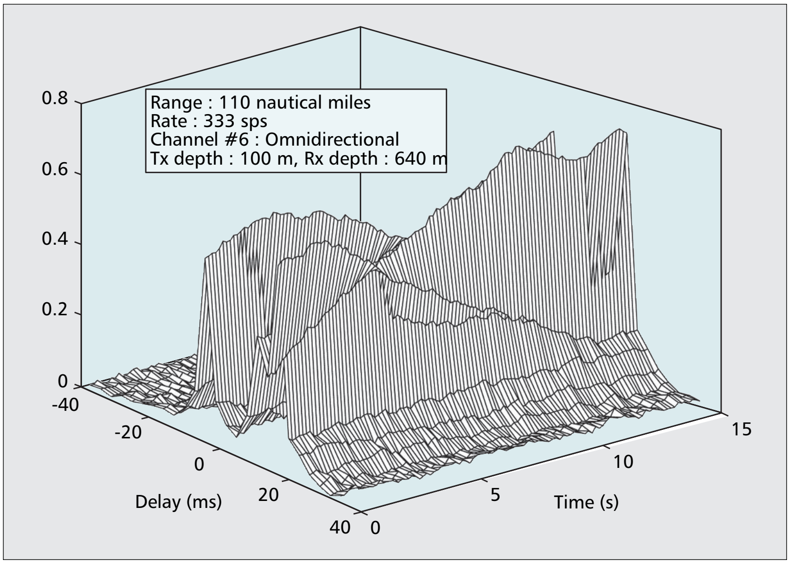
- The large delay spreads (travel time spreads) induced by severe multipath propagation
- The presence of Doppler spread, due to source/receiver motion as well as motion of the water column (waves) that may not be well represented by a simple Doppler shift
- The frequency dependence of propagation loss, and the comparatively low velocity of acoustic propagation (compared with RF propagation) [1, 2]

While frequency-dependent propagation losses yield relatively small available signal bandwidth, potentially large delay spreads lead to strong frequency selectivity that may be highly time-varying. Additionally, narrowband assumptions typically used in RF links rarely apply, as the signal bandwidth is not a negligible fraction of its center frequency of modulation.

As shown in Fig. 1 for a shallow water environment, in addition to the direct path of propagation, the signal propagates via multiple reflections from the surface and bottom. Additional spreading may occur due to the relatively rough scattering surfaces of each. Variations in sound speed across depth lead to ducting, or (time- and spatially-varying) waveguide behavior, creating additional sources of path-dependent propagation. In deep water, such ray bending and surface interactions are the primary sources of multipath. Figure 2 shows an ensemble of channel responses obtained in deep water with a multipath spread on the order of 10 ms. The observed time variation of the channel response is caused by both environmental fluctuations and source and receiver motion. One major difference between research in underwater acoustic communications and RF wireless communications lies in the vast variability of the nature of the underwater acoustic medium. As such, good synthetic channel models simply do not exist. While research can be guided by numerical simulation and experimentation, the only true test of such technologies involves at-sea transmission and reception. Due to the relatively large resources required for such testing, many measurements are often made in concert with a given test to enable further numerical experimentation and post-processing. For example, many environmental measurements are made and correlated with recorded measurements, along with additional measurements of the acoustic ambient noise, such that experiments might be "replayed" at various signal-to-noise ratios in the laboratory. While there are no "typical" underwater acoustic channels, the highly reverberant and nonstationary nature of this example might be viewed as "typical" sample functions over a broad ensemble of random environmental conditions.



■ **Figure 1.** *The underwater acoustic communication channel experiences time-varying multipath due to multiple reflections off the moving surface waves and rough ocean bottom. Relative motion of the transmitter and receiver induces Doppler spread. Noise is introduced by wind, shipping traffic, and various forms of ocean life.*



■ **Figure 2.** *Ensemble of impulse responses over a long range and in deep water. (Courtesy [1]).*

The effect of time-varying multipath propagation is intersymbol interference in the digital communication system that extends over several tens to several hundreds of symbol periods, rendering many of the methods used in RF wireless systems practically useless from a computational complexity perspective. For example, minimum bit error rate receivers, given by a maximum likelihood receiver, have a computational complexity that is exponential in the delay spread of the channel. As mentioned previously, the acoustic propagation velocity varies with depth and location, but is nominally c = 1500 m/s. The comparatively slow propagation velocity (vis-à-vis RF links) dramatically reduces the efficacy of

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any modulation technique based on receiver feedback of channel state information. Another issue that arises from the relatively low propagation velocity is the potential for the generation of severe Doppler distortion in systems with source and receiver platform motion. For example, for source/receiver relative velocity v (say, a few meters per second), the Mach number $M = v/c$ can often be several orders of magnitude greater than that experienced in mobile RF wireless systems. As a result, many practical acoustic communication systems have been designed with explicit Doppler estimation, tracking, and correction, which often involves not only correcting for frequency and phase offsets in the received signal, but may also involve dynamic resampling of the waveform to account for temporal dilation or contraction [1–3].

The remainder of this article is structured as follows. The next section discusses challenges related to single-carrier acoustic communication links, including synchronization and timing, modulation, detection, and equalization, and some of the methods that have been developed to address these challenges. Both single- and multichannel systems are considered. We then describe the use of error control coding in such systems, and the development of iterative equalization and decoding methods. In the following section some of the recent emphasis on multicarrier modulation techniques is discussed along with the associated challenges of working with such modulation strategies in time-varying environments. Next the use of spatial multiplexing and space-time codes for multi-element transmission systems is discussed. The final section describes areas of ongoing research and key challenges for current and future work.

Single-Carrier Systems

Historically, communication over the underwater acoustic channel was limited to noncoherent techniques such as frequency shift keying (FSK). The rapid phase variation that results from the combination of transmitter/receiver movement and slow sound propagation in the underwater acoustic channel made carrier phase tracking appear too difficult to allow for coherent detection. Noncoherent approaches, of course, provide reduced bandwidth efficiency relative to coherent schemes, which is of particular detriment in the bandwidth-limited underwater acoustic channel. Hence, coherent signaling and detection schemes employing pulse amplitude modulation have more recently been explored.

Development of signal-processing-based methods for compensation of the multipath propagation and Doppler fluctuation effects of the underwater acoustic channel can be particularly challenging. While the complex amplitudes and delays of the multipath components vary with time, the carrier phase along each path can vary rapidly as well, which poses a significant hurdle for coherent detection-based methods. While the complex coefficients of an adaptive digital equalizer can adapt to changes in both the multipath components' amplitudes and carrier phase, when the rate of phase variation is large, the tracking ability of such an adaptive equalizer can be compromised. This results in coefficient rotation and additional adaptation noise. As a result, the number of independent observations over the coherence interval of the underwater acoustic channel may be less than (and often significantly less than) the number of parameters needed in the equalizer to mitigate the channel effects. By taking into account knowledge of the physical phenomena that give rise to coefficient rotation and explicitly tracking (and compensating for) the carrier phase, the net effect is a dramatic improvement on the tracking ability of such an adaptive equalizer [4].

For a variety of reasons, including the applications of interest, the relatively slow propagation velocity and the highly variable nature of the underwater acoustic environment, signal transmission typically occurs in packets (blocks) of data, along with which a large time-bandwidth channel probe is generally transmitted at the beginning for symbol synchronization and coarse channel estimation. The channel probe can be filtered at the receiver to provide frame synchronization and initialization of any channel estimation or equalization parameters, and subsequent training data can be used to further track the channel or adapt the equalizer and synchronization parameters.

For a phase-coherent pulse amplitude modulation (PAM) signal, a complex baseband representation of the transmitted signal can be given by

$$x(t) = \sum_n d_n g(t - nT),$$

where d_n denotes the PAM data symbols and $g(t)$ denotes the transmit pulse. Assuming coarse (frame level) synchronization has been achieved, the received signal can be written as

$$r(t) = \sum_n d_n h(t - nT - \tau)e^{j\phi} + w(t)$$

for t in an observation window over which the channel can be assumed to be constant. In this model, $h(t)$ denotes the channel impulse response (including transmit and receive filtering), T denotes the symbol period, ϕ denotes the carrier phase, $\tau = d/c$ is the propagation delay over path distance d, and $w(t)$ denotes additive noise.

A critically important development for underwater acoustic communications was the demonstration of the feasibility of high-data-rate coherent modulation and detection in underwater acoustic communications [4]. Noting that effective synchronization is inhibited by time-varying intersymbol interference (ISI) and that effective equalization of such ISI relies on successful synchronization, a receiver was developed that jointly addresses synchronization and equalization. The receiver employed an adaptive decision feedback equalizer (DFE) with embedded carrier recovery, similar to a single-channel version of the receiver shown in Fig. 3. The equalizer coefficients and carrier recovery parameters can be jointly estimated according to a minimum mean square error (MMSE) criterion. To account for time variation in the channel, a least mean squares (LMS) approach can be employed to update the equalizer coefficients, and a second order phase locked loop (PLL) can be used to track the synchronization parameters. Recursive least squares (RLS)-based methods can improve convergence at the expense of added computational complexity relative to that of LMS.

Within a DFE-based receiver, improved phase tracking has emerged [3] that specifically addresses large Doppler shifts introduced by motion of the transmitter and receiver. Training data can be used to construct a gross Doppler estimate that can be used to resample the data. The Doppler-corrected received signals can then be processed with a DFE with an embedded PLL. Potential instability in the PLL can be overcome by performing phase adjustment prior to the feed forward filter of the DFE.

Multichannel receivers are often employed in underwater acoustic systems to improve performance through processing and diversity gains [5]. In Gaussian noise, the optimal combiner consists of the inverse noise covariance matrix followed by a bank of matched filters, each tuned to the channel seen by the respective receiver, whose outputs are summed. When the channel (including phase) is fully known, the output forms a sufficient statistic on which single channel equalization can be performed. The effective single channel spectrum is simply the sum of the individual channel spectra; hence, multichannel combining significantly reduces the likelihood of observing a null in the channel spectrum. In the rapidly time-varying underwater acoustic channel, however, equalization and synchronization must be performed adaptively.

One of the potential drawbacks of multichannel combining and equalization is the computational complexity required to operate a bank of long adaptive filters, as well as the potential for numerical instability when computationally efficient algorithms are employed. Significant complexity reduction can be achieved by implementing preprocessing to reduce the number of effective channels and hence the number of adaptive equalizers. In [6] adaptive beamforming is used to develop an L-channel receiver that reduces computational complexity by employing an $L \times P$, $P < L$ adaptive beamformer/combiner followed by a P-channel DFE similar to that proposed in [5]. Combining multiple received signals prior to equalization has the added benefit of reducing noise enhancement by reducing the appearance of spectral nulls.

While the receivers described above employ adaptive equalizers that bypass explicit channel estimation, the underwater acoustic channel may also be modeled, and this model can be estimated and tracked directly for use in equalization. An MMSE (or other criterion) equalizer can be calculated directly from the known, possibly time-varying, channel. The extended delay spread and rapid fluctuation of the underwater acoustic channel make channel estimation particularly challenging. With the slow propagation of sound in water and the wide bandwidth of the transmitted signals in comparison to the inverse delay spread, multiple arrivals generated via reflections from the surface, bottom, and other scatterers can be resolved in time, and a sparse channel response results. The effectiveness of channel estimation efforts can be improved by exploiting such sparsity, thereby reducing the number of parameters that must be tracked, and consequently improving convergence and reducing complexity. Sparsity and time variation of the channel can be jointly addressed [7] by estimating the delay-Doppler-spread function, which models channel variations in the Fourier domain and can be assumed to remain constant over a longer window than the channel response. The matching pursuit algorithm and its variants can be used to generate sparse estimates of the delay-Doppler-spread function from which channel estimates can be directly determined. The effects of channel estimation error on the performance of many forms of equalization can be analyzed as a function of the acoustic channel scattering function [8]. Such results allow for performance prediction of various equalization methods under a given set of environmental conditions. In addition, the insight gained can be used to develop a channel-estimate-based DFE that appears more robust to errors in channel estimation.

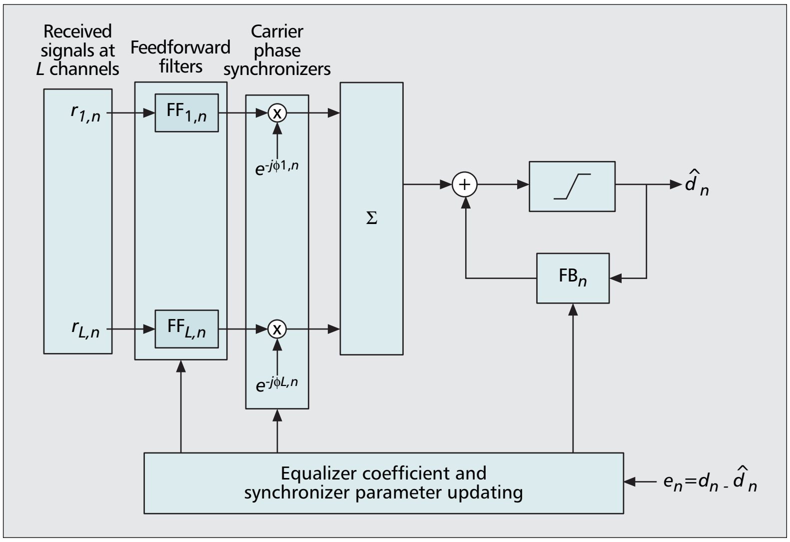

Figure 3. *Multichannel DFE receiver for underwater acoustic communications. Received signals are processed by a bank of adaptive linear filters that jointly perform matched filtering and feed-forward equalization. Adaptive phase synchronization is then performed on each branch before the signals are combined and passed to a single DFE feedback and decision loop.*

Iterative Equalization Methods

To increase the fidelity of such underwater acoustic communication links, a controlled amount of redundancy is added for error correction coding (ECC). Block codes, convolutional codes, and LDPC codes have all been used in various contexts. While including ECC helps to reduce overall bit error rate (BER), ECC is considerably less effective when used subsequent to but isolated from equalization in the presence of severe ISI. Fortunately, the block processing nature of the transmitted data allows the use of the vast array of iterative decoding and equalization algorithms that have been developed since the advent of turbo codes and turbo equalization

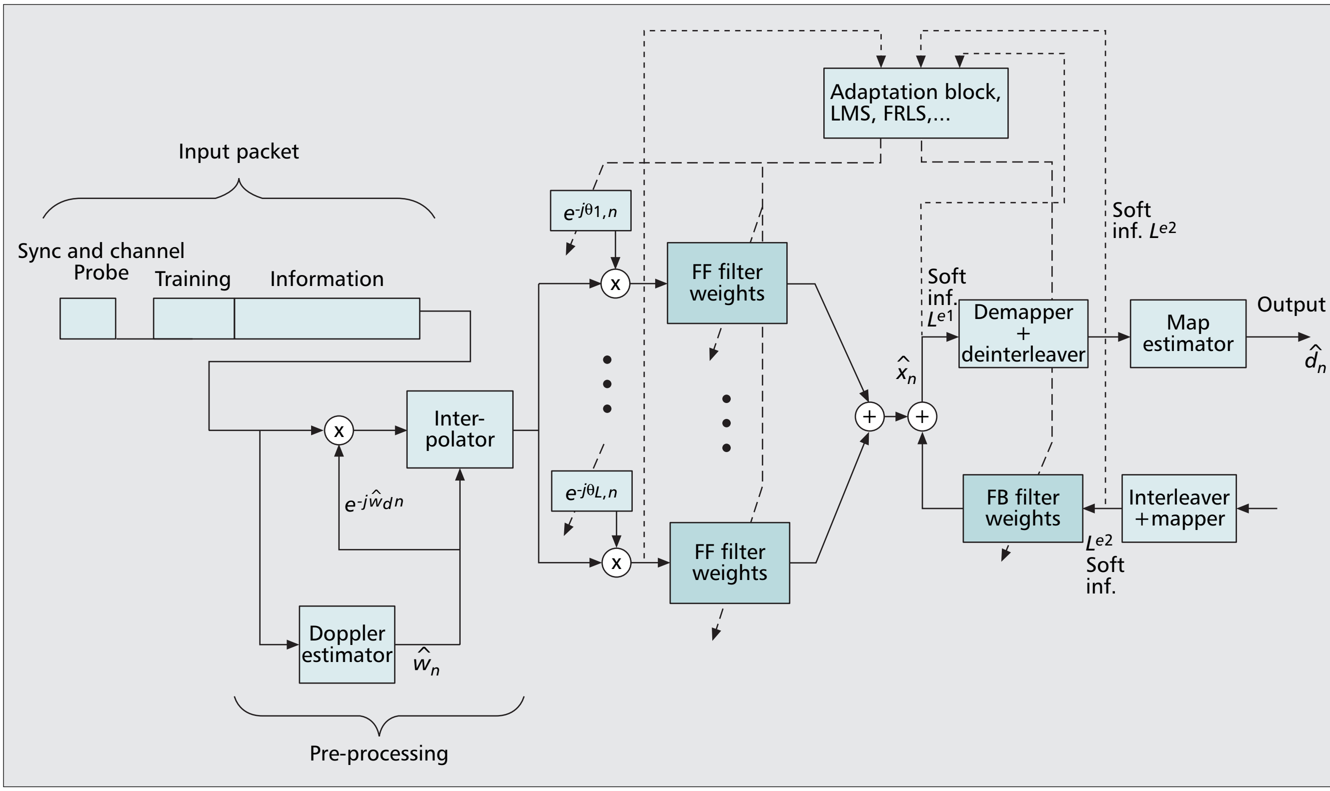

Figure 4. *Block diagram of an underwater acoustic communications link using turbo equalization with adaptive phase tracking.*

[9] methods over the last decade. Turbo equalization has been shown to provide significant performance gains, even for severe ISI channels, through iterative soft-input/soft-output equalization and decoding. However, due to the long delay spreads involved, MAP-based turbo equalization is simply impractical. Similarly, MMSE-based methods requiring channel knowledge at the receiver also have a computational complexity that is often beyond the available resources. This computational burden is further amplified by the need to perform the equalization and decoding steps multiple times over the received data block. A number of low-complexity methods for turbo equalization have been developed over the last decade, and many of these methods have been applied with initial success in underwater acoustic environments including those exploiting multichannel spatial and temporal diversity combining methods [10, 11].

Multicarrier Modulation Techniques

One approach to overcoming the long delay spread inherent in underwater acoustic channels is through the use of multicarrier methods, such as orthogonal frequency-division multiplexing (OFDM), which transform the frequency-selective channel into several narrower flat fading channels. Under this model, equalization can be performed by multiplying each flat fading channel output by a single complex tap value, thereby significantly reducing complexity by eliminating the need for long equalization filters to combat ISI. The principle challenge in using multicarrier modulation on the underwater acoustic channel is the effect of time variation in the channel. The single-tap equalization enjoyed by multicarrier methods relies on orthogonality among the carriers. In the presence of Doppler spread, orthogonality no longer holds, and intercarrier interference (ICI) results.

One proposed method for addressing this challenge is to use a sparse Fourier basis expansion to model the channel in the frequency domain and allow for a small amount of ICI [12]. Joint channel estimation and data detection can then be performed using a turbo-style receiver. A banded OFDM approach [13] can also be used to further reduce the complexity of channel estimation at the receiver.

Space-Time Modulation Techniques

As has been the trend in RF wireless systems, to achieve higher spectral efficiency over the limited available bandwidth, spatial multiplexing techniques have emerged, making use of the spatial diversity offered in many acoustic transducer arrays used for signal transmission. When used in concert with receive-hydrophone arrays, as with RF wireless systems, the theoretical link capacity can increase with the number of effective simultaneous independent channels available, although such explicit analysis for the underwater acoustic environment remains an open problem. Many of the same multiplexing and diversity trade-offs discussed in RF wireless multiple-input-multiple-output (MIMO) links are also beginning to be explored in the context

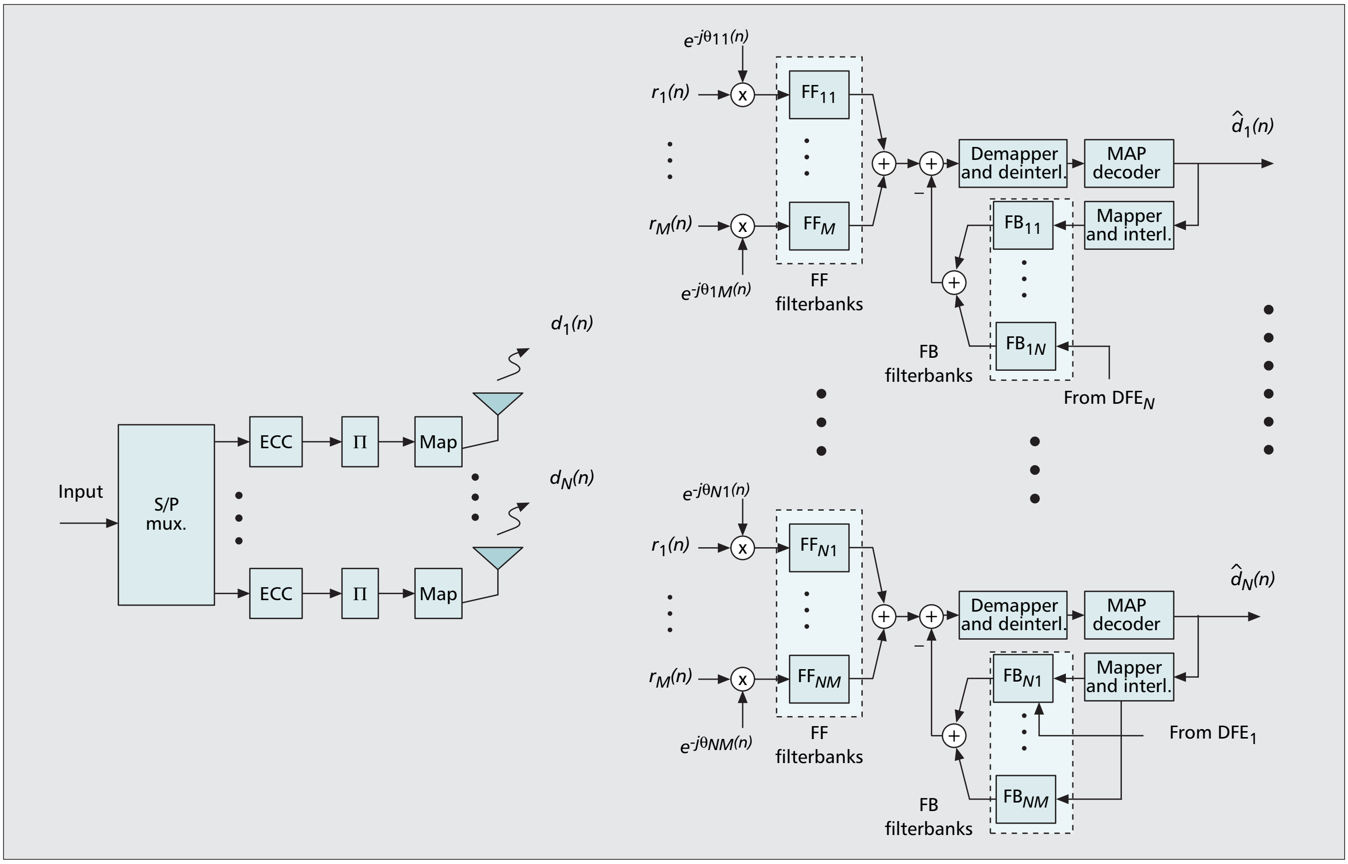

Figure 5. *An example of a MIMO transmit and receive system for operation in the underwater acoustic channel.*

of underwater acoustic communications [14]. When used with space-time coding (coding across both spatial and temporal axes), reliability can be further enhanced through the use of iterative equalization and decoding methods, similar to those used in the single-input/single-output (SISO) (i.e., turbo equalization) case.

For such channels with large delay spreads, MIMO transmission and decoding methods cannot typically rely on the type of MAP-based optimal detection and decoding algorithms used in many RF wireless MIMO systems. However, adaptive DFE-based receivers with turbo equalization can provide systems with moderate complexity that can achieve communication rates that would have been unachievable through their SISO counterparts. Such adaptive equalizers not only need to mitigate ISI, due to the large delay spread in the channel, but also must accommodate co-channel interference (CCI) across the various source/receiver channel pairs.

An example of such a MIMO system employing spatial multiplexing with ECC and a multichannel iterative DFE in the receiver is shown in Fig. 5. Here, the data is first spatially multiplexed across the transmit streams, one for each transducer in the transmit array; then each transmit stream is independently coded and mapped onto the channel symbols prior to modulation and transmission. At the receiver, the signal is then processed using a multichannel DFE, making full use of the receiver array. For each transmit stream, the receive hydrophone array employs a series of feed-forward filterbanks with coherent phase tracking to formulate a coherent estimate of the corresponding transmit stream, incorporating a DFE to eliminate both ISI and CCI using estimates from all of the DFEs as feedback. To simultaneously improve achievable data rates and reduce receiver complexity, OFDM within the MIMO framework (so-called MIMO-OFDM) has also been explored in the underwater context, expanding on similar activities in the RF wireless literature. To enable such MIMO-OFDM operation, the channel must be assumed constant over an entire OFDM block, and null subcarriers can be used to estimate Doppler shifts [15].

Concluding Remarks

This article provides a brief introduction to some of the exciting work in signal processing for underwater acoustic communications. The systems employed to date leverage state-of-the-art signal processing methods, including multichannel equalizers, with explicit embedded phase tracking and symbol timing, Doppler tracking and signal resampling, and spatial multiplexing methods in MIMO-turbo equalization transmitter/receiver systems. However, as stated earlier, the underwater channel provides a number of unique attributes that continue to challenge present systems. For example, the large delay spreads in shallow water environments continue to challenge the computational resources required to operate an adaptive multichannel DFE in portable systems. Perhaps better models

While the slow propagation speeds prohibit the efficacy of full channel state information feedback, the question of how much feedback, if any, might offer additional computational savings or performance enhancement remains open.

might capture the salient characteristics of the acoustic channel with substantially fewer parameters than the explicit DFE or time-varying impulse response structures used presently. Researchers developing OFDM-based receivers continue to struggle with the need for long block lengths to maximize computational savings, at the expense of nonorthogonal carriers, due to temporal variability. Finally, while the slow propagation speeds prohibit the efficacy of full channel state information feedback, the question of how much feedback, if any, might offer additional computational savings or performance enhancement remains open.

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BIOGRAPHIES

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