Peer-Reviewed Technical Communication

An Overview of Underwater Time-Reversal Communication

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***Abstract*—Time reversal (TR) exploits spatial diversity to achieve spatial and temporal focusing in complex environments. Over the last decade, the TR concept has been applied successfully to phase-coherent acoustic communications in time-varying multipath ocean environments, as an alternative to conventional adaptive multichannel equalization. Temporal focusing (pulse compression) mitigates the intersymbol interference (ISI) and subsequent single-channel equalization removes the residual ISI, thus providing nearly optimal performance in theory. The spatial focusing capability facilitates multiuser or multiple-input–multiple-output (MIMO) communications without explicit use of time, frequency, or code division multiplexing, while an adaptive TR approach can reduce further the crosstalk among users or multiple transmitters. TR communications can be extended easily to time-varying channels using a block-based approach with channel updates. This paper provides an overview of TR communications in both shallow and deep water and recent advances including bidirectional equalization, multiuser communications with mobile users, and communication with a glider serving as a mobile gateway.**

***Index Terms*—Adaptive multichannel equalization, crosstalk, MIMO/multiuser communication, time reversal (TR), underwater acoustic (UWA) communication.**

I. Introduction

UNDERWATER acoustic (UWA) channels are characterized by large multipath spreads resulting in intersymbol interference (ISI) in the context of acoustic communications that degrades the quality of the received signal and requires compensation (i.e., channel equalization). The time-varying nature of the multipath also requires continuous tracking of the receiver parameters necessary for demodulation. The available bandwidth in UWA channels is limited due to severe frequency-dependent attenuation of the physical medium. In addition, the slow speed of sound propagation ($\sim$1.5 km/s) makes Doppler effects significant even for relatively slowly moving platforms. As a result, UWA channels are considered one of the most challenging environments in which to achieve bandwidth-efficient, reliable (robust) communication. Almost every aspect of acoustic telemetry including channel equalization up to the 1990s is covered extensively in the review paper by Kilfoyle and Baggeroer [1].

In the early 2000s, a time-reversal (TR)-based approach [2], [3] was proposed for phase-coherent UWA communications, a spinoff of a decade-long research on TR physics [4]. Since then TR communications has made significant progress and is accepted as an alternative to adaptive multichannel equalization developed in the early 1990s [5], along with more recent multicarrier orthogonal frequency-division multiplexing (OFDM) [6]. This paper aims to provide an overview of TR communications from basic physics, salient features of TR techniques, to the latest developments.

II. Time-Reversal Physics

TR is a process of transmitting the received signal on an array in a time-reversed order while the array is referred to as a time-reversal mirror (TRM). Due to spatial reciprocity and TR invariance of the linear acoustic wave equation, the retransmitted signal converges back to the position where the original signal was generated [frequently referred to as the probe source (PS)]. Since TR corresponds to phase conjugation (PC) in the frequency domain, TR and PC are used interchangeably. Excellent reviews on TRM can be found in [7] and [8].

Fig. 1 illustrates the components of an active TRM experiment conducted in the ocean. First, a PS indicated by one of the rectangles (e.g., middle) on the vertical receive array (VRA, left) in Fig. 1(a) sends out a pulse (e.g., 2-ms tone at 3.5 kHz) that is received at the source–receive array (SRA, right). Second, the dispersed signal with all of its multipath structure shown in Fig. 1(b) is time reversed and retransmitted by the SRA. This process is referred to as backpropagation with the SRA emulating a TRM. Finally, the signal multipath structure collapses to a spatial and temporal focus (almost the original PS pulse length, 2 ms) at the original PS (40-m depth) that is colocated in range with the VRA as shown in Fig. 1(c).

The resulting temporal and spatial focusing without *a priori* knowledge of the environment or array geometry is the major property of TR physics. Implicitly assumed is that the ocean environment is stationary (frozen) during the round trip, although this requirement can be relaxed in some cases due to the robustness of TR allowing for some channel variations [9]. The TR focal process can be interpreted in terms of the mode theory [10].

III. Application to Communication

The temporal and spatial focusing capability of TR immediately offers potential application to communications, espe-

Manuscript received September 08, 2014; revised April 06, 2015 and June 19, 2015; accepted July 22, 2015. This work was supported by the U.S. Office of Naval Research.

Associate Editor: J. Gomes.

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Digital Object Identifier 10.1109/JOE.2015.2461712

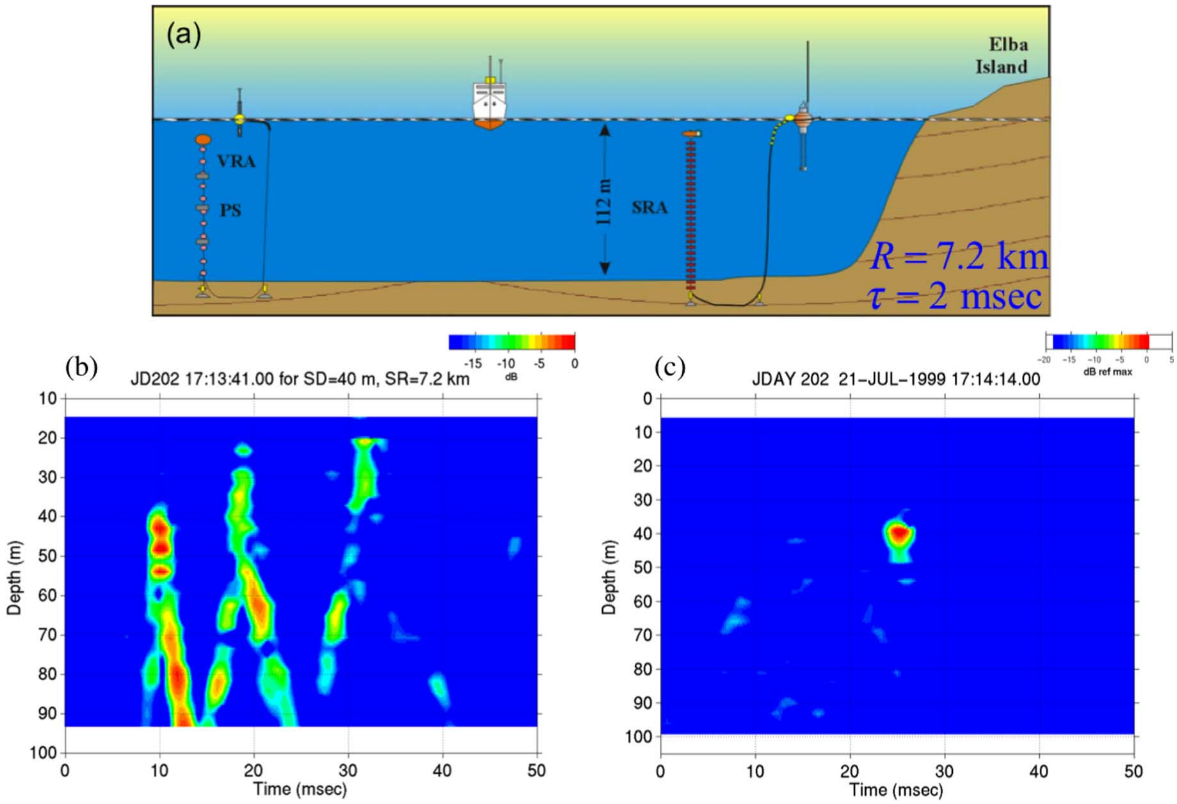

Fig. 1. (a) Schematic of active TRM experiments in the Mediterranean Sea. (b) The probe source data (2-ms pure tone at 3.5 kHz) at the vertical SRA which is time reversed and retransmitted, resulting in the focus measured on a VRA collocated with the PS shown in (c).

cially in an environment with significant multipath. The temporal compression mitigates ISI resulting from multipath propagation, while the spatial focusing achieves a high signal-to-noise ratio (SNR) at the intended receiver (PS) with a low probability of interception elsewhere.

When a known signal $s(t)$ (e.g., shaping filter) is transmitted from a PS, the (noiseless) received signal on the ith element of the TRM in Fig. 1(b) is $r_i(t) = s(t) * h_i(t)$ where $h_i(t)$ is the channel impulse response (CIR) and $*$ denotes convolution. The M-element TRM retransmits a time-reversed version of the received signal $r_i(-t)$ which propagates through the same channel $h_i(t)$. The signal received back at the PS position $s_{ps}(t)$ then can be written as

$$\begin{aligned} s_{ps}(t) &= \sum_{i=1}^{M} r_i(-t) * h_i(t) \\ &= s(-t) * \left[\sum_{i=1}^{M} h_i(t) * h_i(-t)\right] \end{aligned} \tag{1}$$

where the term in the bracket is referred to as the q-function, the summation of the autocorrelation of each CIR [11].

In an ideal case, the q-function denoting an overall TR system's impulse response approaches a Dirac delta function [i.e., $q(t) \approx \delta(t)$] and allows for the original signal to be recovered at the PS location in a time-reversed fashion $s(-t)$, while the shaping pulse $s(t)$ usually is symmetric for communications (e.g., a raised cosine filter [12]). Thus, the TR performance is fully characterized by the q-function which depends on the channel complexity (i.e., the number of multipaths), the number of array elements (M), and their distribution in space (spacing and aperture).

In digital communications, the process of undoing the multipath or channel distortion is called channel equalization and has been studied extensively in the literature [12]. The two-way TR process essentially is performing self-equalization as characterized by the q-function, albeit not perfect, since $q(t) \neq \delta(t)$ in practice.

In this original setup, a message is sent from TRM to PS where the TRM and PS correspond to a base station and a user in terrestrial communications, respectively, i.e., downlink with a multiple-input–single-output (MISO) system. The application of this active TR concept to underwater communications was first demonstrated in shallow water using a 29-element, 78-m aperture TRM operating in the midfrequency band (3–4 kHz) over 10-km ranges [2].

A. Active (Downlink) Versus Passive (Uplink)

TR involves backpropagation from TRM to PS in Fig. 1, which can be carried out numerically without actual retransmission using known or estimated CIRs at TRM, $h_i(t)$, as indicated in Fig. 2. In this case, $s(-t)$ in (1) is replaced by $s(t)$ which remains the same for a symmetric signal. On the other hand, the communication direction is opposite from PS (user) to TRM (base station), i.e., uplink with a single-input–multiple-output (SIMO) system. This one-way communication is referred to as passive TR or PC (PPC) as opposed to the original two-way active TR communication. In addition, the theoretical performance of the two approaches is shown to be identical in [13].

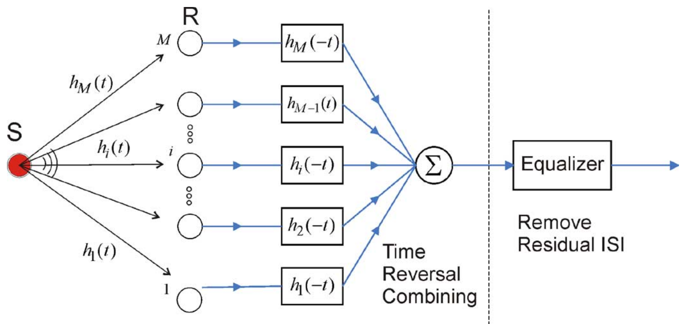

Fig. 2. Block diagram for a passive TR (uplink). An active TR (downlink) is equivalent to the passive TR with the communication link being in opposite directions. The vertical line indicates temporal and spatial focusing achieved at the PS location via the TR process. Subsequently, TR can be followed by single-channel equalization to further remove residual ISI, offering nearly optimal performance (see Section IV-A).

In summary, passive (uplink) TR is equivalent to active (downlink) TR with the communications link being in opposite directions. First proposed by Dowling [14], PPC communication was demonstrated using a 14-element VRA in shallow water (5–20 kHz) [3]. Note that passive TR was referred to as virtual TR by Silva *et al.* [15].

In practice, active TR is both expensive and difficult, if not impossible, to implement since: 1) it requires a number of transducers (TRM/SRA) that can both receive and transmit signals; and 2) usually there is a considerable lapse of time (e.g., a few minutes) between the reception of the PS signal and retransmission of a communication message composed at TRM, allowing for the channel/environment to vary and potentially violate the stationarity assumption. This especially is true of high-frequency signals typical for acoustic telemetry (e.g., 10–20 kHz) where the channel coherence time can be less than a few hundred milliseconds. Not surprisingly, active TR communications was proven successful only up to the midfrequency band (e.g., 3–4 kHz) in benign shallow-water environments [2], [16]–[18]. Nevertheless, those earlier, limited demonstrations of active, downlink communications are quite valuable for confirming the application of TR concept to communications, albeit not practical.

A common source of confusion is to assume that TR-based communications only refers to the two-way, active, downlink, MISO communications since it originated from the TR physics described in Section II. However, the TR concept is equally applicable to the one-way, passive, uplink, SIMO communications in Fig. 2 with the same theoretical performance. As will be addressed in Section IV-D, an invaluable benefit of passive TR or PPC is its capability of handling time-varying channels effectively on the fly.

B. Time Reversal and Matched Field Processing

TR or PC is relevant to the recent trends in acoustic signal processing which have emphasized utilizing knowledge of the environment such as matched field processing (MFP) [19]. In MFP, we compare data received on an acoustic array with output from a propagation model for the purpose of locating a source and/or determining some properties of the propagation medium. Thus, the comparison is done with replicas derived from an environmental acoustic propagation model whose fidelity is inherently limited by the environmental input to the model.

Since no model or replica is involved, localization cannot be done with active PC or TR while the focus is assumed to occur at the PS location (spatial focusing). In conjunction with a model [generating $h_i(t)$ in Fig. 2], however, PPC corresponds to an unnormalized conventional MFP with a potential sidelobe structure (i.e., ambiguities) [4]. From a signal processing point of view, TR or PC is just a spatio–temporal matched filter (MF) that achieves a maximum SNR at the PS.

In the context of acoustic communications, TR or PC enjoys a significant advantage over MFP: 1) TR has a fully cooperative PS whose signal characteristics are completely known to the receiver, enabling fully coherent broadband MFP; and 2) spatial sidelobes are not a concern for point-to-point communications between a source and a receiver except for covert communications [20].

IV. Time-Reversal Communication

The earlier TR communications, both active [2], [17] and passive [3], [21], [22], were successful in demonstrating its potential along with its robustness and computational simplicity. As described in Section III, the performance of TR depends entirely on the behavior of the q-function and thus it is desirable to have a q-function that approaches a delta function to minimize the ISI. In practice, however, there is always some residual ISI. Moreover, a mismatch between the assumed and actual CIRs introduces additional distortion. In an effort to maintain the quality of the q-function, adaptive spatial combining with different weighting was proposed by Gomes *et al.* [23]. For phase tracking, a decision-feedback carrier-phase estimate based on maximum likelihood (ML) [12] can be applied before decoding.

A. TR With Equalization (TR-DFE)

Despite the proof of concept, it was clear that TR alone would be far from being optimal since $q(t) \neq \delta(t)$. Using theoretical performance bounds, Stojanovic [13] showed that the TR performance would saturate due to the residual ISI with an increasing SNR. Yang [21] compared the difference between PPC and the multichannel decision-feedback equalizer (M-DFE) using both simulations and experimental data, indicating the limitations of the TR approach. Further, Preisig [24] examined the variances (σ^2) of symbol estimates for three different approaches including a linear equalizer (LE) and showed that $\sigma^2_{\mathrm{DFE}} \leq \sigma^2_{\mathrm{LE}} \leq \sigma^2_{\mathrm{TR}}$. Consequently, a series of extensions or modifications has been proposed to improve the original TR approach.

First, the TR combining architecture was followed by an adaptive single-channel equalizer to further eliminate the residual ISI. The overall block diagram is illustrated in Fig. 3(a) in the discrete time domain with a feedforward finite impulse response (FIR) filter $a[n]$ and a feedback FIR filter $b[n]$. Following the initial investigation [16], active TR combined with a DFE (TR-DFE) was demonstrated using a 2-D billboard array (3–4 kHz) at a 2-km range in 50-m deep shallow water [18]. Using high-order constellations such as 32 quadrature

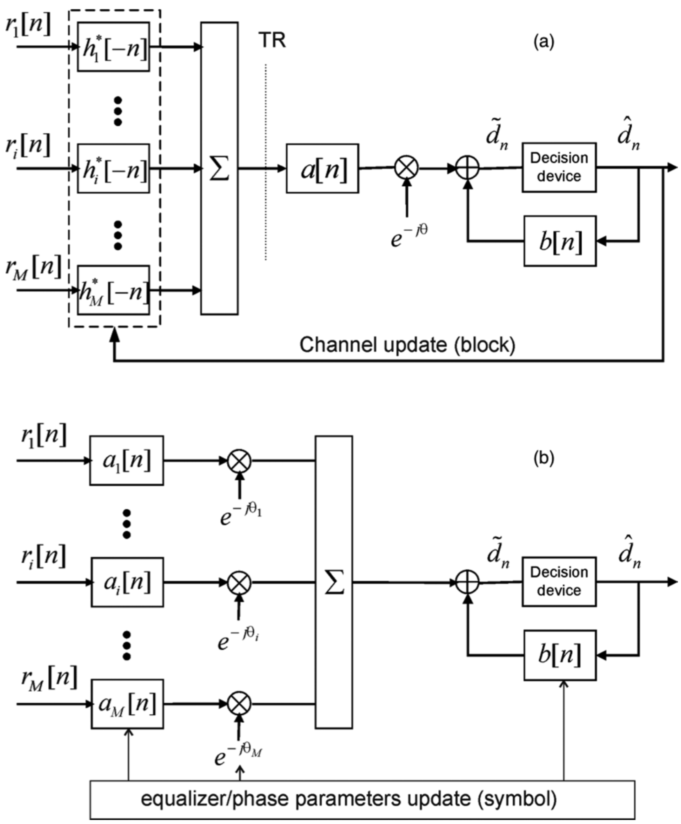

Fig. 3. Block diagram: (a) TR combining followed by a single-channel adaptive decision-feedback equalizer (TR-DFE) and (b) conventional adaptive multichannel DFE (M-DFE). $a_i[n]$ and $b[n]$ are feedforward and feedback filters, respectively, $h_i[n]$ is channel impulse response, and $\hat{d}_n$ is an estimated symbol. Note that (a) can be implemented on a block-by-block basis while the channels are updated using previously detected symbols (decision-directed mode). In (b), phase tracking is carried out individually on each channel (θ_i) whereas in (a) phase tracking is required only for a single channel (θ) after TR combining.

amplitude modulation (32-QAM), it was confirmed that indeed the performance of TR alone is saturated and TR-DFE can improve the performance significantly over TR as shown in Fig. 4 (i.e., 13-dB increase in terms of output SNR).

Similarly, passive TR-DFE was demonstrated successfully by Song *et al.* [25] as well as Yang [26] who called it a correlation-based DFE. Gomes *et al.* [23] also examined the performance of various TR methods using data collected in a continental shelf experiment off the west coast of Portugal. Now that TR-DFE is routinely employed for TR-based communications, simple TR is referred to as TR-DFE henceforth, unless specified otherwise.

Various potential approaches to underwater communications also were investigated by Stojanovic [13], including TR and channel equalization, who analyzed their theoretical performance bounds using a simple model channel under ideal conditions. It was indicated that TR requires a large number of array elements to compete with other approaches, which stems from misunderstanding of the spatial diversity exploited for TR or MFP described in Section III-B. Song and Kim [27] demonstrated that TR combined with equalization (TR-DFE) offers nearly optimal performance using a modest 4-element array with appropriate spatial diversity [28]. Further, Song *et al.* [29] compared the performance of TR alone and TR-DFE

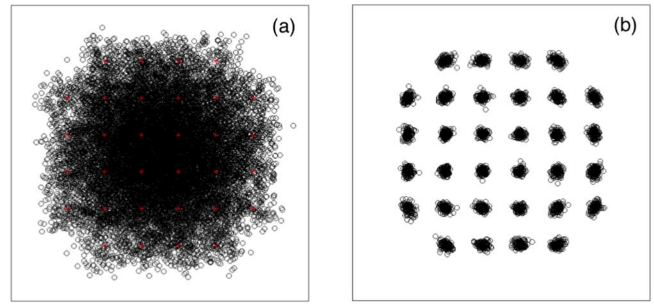

Fig. 4. Scatter plot for 32-QAM modulation: (a) TR and (b) TR-DFE. Reproduction of Fig. 9 in [18].

between data and theory, confirming that the theoretical performance [13] can provide a useful upper bound for predicting the performance of TR communications.

B. Multichannel Equalization (M-DFE)

As a baseline, a block diagram for standard multichannel DFE (M-DFE) is displayed in Fig. 3(b), involving M feedforward filters $a_i[n]$ and a single feedback filter $b[n]$. Coupled with a second-order phase-locked loop (PLL) for phase tracking of $\theta_i[n]$, M-DFE has been used for UWA communications over the last two decades, which in theory can provide an optimal performance similar to TR-DFE [13].

There are two ways to implement M-DFEs in time-varying channels. First, channel-estimate-based equalizers (CE-DFEs) are those that calculate the filter coefficients ($a_i[n]$ and $b[n]$) based on estimates of the time-varying CIRs and the statistics of the ambient noise field [24], [30]. Given the explicit knowledge of the channel required, TR-DFEs also can be viewed as CE-DFEs. Alternatively and more often, the equalizer coefficients are directly tracked using either the least mean square (LMS) or the recursive least squares (RLS) adaptive algorithms [12]. In this case, the CIRs are estimated rather implicitly as matched filters at the front end of the feedforward filters $a_i[n]$, whereas matched filtering is performed explicitly in TR-DFE.

While adaptive M-DFEs embedded with a PLL have been applied successfully to UWA channels [1], the major issue remains the computational complexity since the number of degrees of freedom (DOF) is proportional to the product of the equalizer length in taps times the number of array elements (M). Many acoustic channels are dispersive, and the number of taps needs to span the duration of the channel sampled at the symbol rate times a factor (usually two) for fractional-spaced equalizers (FSEs) [12]. At high symbol rates, the larger number of DOF coupled with insufficient data to support the adaptation leads to instabilities associated with updating adaptive filters. Consequently, reducing the receiver complexity for efficient implementation has been the challenge ever since its introduction.

C. Reduced-Complexity M-DFE: TR-DFE

Notably, a spatial precombiner (or optimal beamforming) was proposed which aimed to reduce the total number of input channels (M) to a smaller number $(P < M)$ for subsequent multichannel equalization [31]. A simple example of the spatial precombiner is a plane wave beamformer when the array elements are closely spaced (e.g., half the wavelength) to mitigate ISI and channel fading by passing a dominant path while rejecting undesirable multipath interference and noise [32]. This approach requires angular separation of multipath components and typically more array elements, which is not optimal since all multipath components are not fully exploited. Alternatively, eigenvector beamformers [33] use the eigenvector corresponding to the largest eigenvalue, rather than focusing on specific multipaths.

With the array elements well separated (e.g., three to four times the wavelength) to ensure spatial diversity, each beam output (P) does not necessarily correspond physically to the individual component of the multiple paths. Ideally, the parameters of the spatial precombiner and multichannel equalizers can be jointly optimized to minimize the mean squared error (MSE) of the symbol estimates [34]. Further, the possibility that the reduction in complexity might be achieved at no cost in performance was indicated in [34] since the multipath structure is not independent among the array elements.

Recall from Section III-B that passive TR or PPC is equivalent to broadband MFP which coherently combines the multiple paths, referred to as a generalized beamformer [19], and then is followed by single channel equalization (1-DFE) as shown in Figs. 2 and 3(a). Thus, TR-DFE can be viewed as a special case of reduced-complexity M-DFE with $P = 1$ [31]. While achieving optimal performance [27], TR-DFE also reduces the computational complexity as follows. First, TR-DFE involves a single TR-combined impulse response defined by $q[n]$ regardless of the number of channels or diversity M, rather than M individual CIRs, $h_i[n]$. Second, the number of taps or DOF required for TR-DFE is much smaller than the case with M-DFE due to the already compact structure (temporal compression) of $q[n]$. Third, the averaged single phase $\theta[n]$ needs to be tracked, in contrast with M individual phase fluctuations $\theta_i[n]$ in M-DFE. Note that the averaged phase tends to vary more slowly than the individual phases.

D. *TR-DFE for Time-Varying Channels*

While M-DFE implemented with an adaptive algorithm (RLS or LMS) is directly applicable to time-varying channels, TR communications are constrained by the time invariance of the channel $h_i[n]$ in both active and passive TR. In practice, underwater acoustic channels are characterized as dynamic, highly dispersive, and sparse. In rapidly time-varying channels or for long-duration data packets, the mismatch between assumed and actual channels will deteriorate the TR performance significantly. However, passive (one-way) TR communications of interest easily can adapt to the channel variability as described below.

A simple approach to avoid the mismatch is frequent insertion of channel probe signals at the expense of data throughput [3]. A more elaborate approach was proposed by Flynn *et al.* [22] which tracks the CIR continuously using previously detected symbols (i.e., decision-directed mode) prior to matched filtering without compromising the data rate, called decision-directed PPC (DDPPC). Note that both PPC and DDPPC are still pure TR approaches, unlike TR-DFEs involving post single-channel equalization.

Similarly, TR-DFE can be extended to time-varying channels [35], [36]. The basic idea is to apply the TR approach on a block-by-block basis such that within each block the channel remains almost time invariant and subsequently is updated using detected symbols (decision-directed mode), thus minimizing the mismatch between assumed and actual channels as indicated in Fig. 3(a). The block size depends on the channel coherence time. Normally, the low-complexity LMS algorithm is employed for faster execution of multichannel estimation whereas the RLS algorithm is applied to the subsequent single-channel DFE for faster convergence. Alternatively, the LMS channel estimation can be replaced by greedy algorithms such as matching pursuit (MP) [37] to exploit the sparsity of the UWA channels.

A potential benefit of the block-based TR approach is elimination of explicit phase tracking required for phase-coherent communications except during the initial training period as demonstrated in [38]. This is accomplished by a combination of a small block size and adaptive channel estimation on a symbol-by-symbol basis. The most recent channel estimates then are applied as matched filters to the immediately following block, leaving just the incremental phase evolution. Any residual phase averaged across the channels can be compensated for further by the subsequent adaptive equalizer [39].

E. *Mobile Source*

Much of the published work on underwater acoustic communications considers a stationary source and receiver in fluctuating ocean environments. In the presence of relative motion between them, broadband communication signals experience a Doppler compression or dilation, which depends on the propagation path (angle) and the relative radial speed between the source and the receiver. The differences in the compression or dilation among the paths will lead to a Doppler spread in multipath ocean environments [40], posing a challenging task coupled with the Doppler spread induced by time-varying channels discussed in Section IV-D.

Typically the geometry is constrained in underwater channels: 1) the motion is horizontal and uniform with a constant speed; and 2) the range separation is much greater than the water depth (i.e., far field). With this assumption, all significant paths will arrive at the receiver with a small angular spread relative to the horizontal, allowing the compression or dilation to be represented by a single (mean) Doppler parameter. The motion effect then can be compensated for by resampling the received signal. However, the residual Doppler for different paths collectively leads to a time-varying CIR, even in the absence of environmental fluctuations in the medium such as surface waves, internal waves, etc., at various time scales. It should be mentioned that at closer ranges (i.e., near field) where the Doppler spread can be significant due to wide arrival-angle separation [41], no TR-based approach is reported in the literature.

By restricting our interests to far-field scenarios, the resampling approach in conjunction with a block-based TR-DFE has been applied successfully to mobile underwater communications [42], [43], [25]. Recently, Song [40] confirmed that the impact of source motion can be minimal provided that the observation time window (block size) is constrained to about a wavelength in terms of a radial travel distance (i.e., quasi-stationary), and thus the resampling process is not required.

On the other hand, the relative motion between a source and receiver pair can be exploited by using synthetic aperture communication (SAC) where a virtual horizontal array provides diversity similar to the spatial diversity provided by a vertical array in a waveguide. In addition, there is a temporal diversity incurred due to the time interval between transmissions or receptions allowing for channel variations. Following the initial investigation using a simply on/off keying modulation [44], phase-coherent SAC has been demonstrated successfully for two different frequency bands (2–4 kHz and 8–20 kHz) and high-order constellations (e.g., 8PSK), and achieved a high data rate using two to five consecutive transmissions from a source moving at 4 kn over 3–6-km ranges in shallow water [45]. Similarly, gliders equipped with a single hydrophone can exploit the combined spatial and temporal diversity generated by a constant motion in deep water as will be discussed in Section VI-B.

V. MIMO/Multiuser Communication

As described in Section II and Fig. 1(c), the spatial focusing capability of TR can be exploited for multiuser or multiple-input–multiple-output (MIMO) communications without an explicit use of time, frequency, or code division multiplexing. Provided that multiple users are well separated in range and depth from each other compared to the TR focal size in a complex environment [10], each user either can receive (downlink) or send (uplink) independent messages simultaneously. The first active (downlink) MIMO/multiuser communications was demonstrated in the Mediterranean Sea using an array of multiple sources (3–4 kHz) and a receiver array separated by 8.6 km in a 105-m deep water [17]. Subsequently, the first passive (uplink) MIMO/multiuser communications were confirmed in the same area such that as many as six users/transmitters sent messages over a 4-km range in a 120-m deep water using QPSK modulation, achieving an aggregate data rate of 6 kb/s [i.e., a spectral efficiency of 6 b/(s·Hz)]. The same data rate also was achieved at 20-km range by three users/transmitters using 16-QAM [46]. As in the single-user case, we focus mainly on the passive (uplink) MIMO/multiuser communications.

Denote $h_i^j(t), i = 1, \ldots, M$ as CIRs from a user/transmitter j to the VRA as depicted in Fig. 2 for a single-user/transmitter case ($j = 1$). The conventional TR (CTR) approach extracts signals for each user j simply by matched filtering the received signals $r_i(t)$ with each set of the channel responses $h_i^j(t)$ and combining [46]. Thus, the performance of MIMO/multiuser TR communication depends on the generalized $q(t)$-function, a measure of the crosstalk between users (m and n)

$$q_{mn}(t) = \sum_{i=1}^{M} h_i^m(t) * h_i^n(-t). \tag{2}$$

When $m = n$, $q_{nn}(t)$ reduces to the original $q(t)$-function defined in (1) for the nth user, and a measure of temporal compression (focusing) is achieved in the absence of other users. An example of $q_{mn}(t)$ for a three-user case ($N = 3$) can be found in [47, Fig. 4].

A. Adaptive Time Reversal

Similar to the ISI case due to $q(t) \neq \delta(t)$, spatial focusing is not perfect and the crosstalk among users or cochannel interference (CCI) among transmitters cannot be completely eliminated using CTR [i.e., $q_{mn}(t) \neq 0$]. Kim and Shin [48] proposed an active adaptive time-reversal (ATR) approach to achieve simultaneous focusing with minimal interference. Subsequently, active ATR based on the minimum variance distortionless response (MVDR) array processor [49] has been extended to passive TR communications [50]. The ATR filter is derived in the frequency domain and then converted back into the time domain using an inverse fast Fourier transform (IFFT).

For simplicity, consider a two-user case ($N = 2$) which can be easily generalized to more users. Let us define a column vector $\mathbf{d}_j$ as the collection of channel frequency responses $H_i^j(f)$ from each user j such that

$$\mathbf{d}_j = \left[H_1^j(f) \cdots H_i^j(f) \cdots H_M^j(f)\right]^T \tag{3}$$

where T denotes the transpose operation. Denote a column vector $\mathbf{w}$ as a weight vector (spatial filter) applied to the received array data $\mathbf{r}$ consisting of signals from both users, $\mathbf{w}^\dagger \mathbf{r}$ where the superscript $\dagger$ denotes complex conjugate transpose. For CTR, $\mathbf{w}_j = \mathbf{d}_j$ for each user, ignoring the crosstalk or CCI resulting from spatial correlation between the two users, $\rho_{12} = \mathbf{d}_1^\dagger \mathbf{d}_2$. To suppress the crosstalk, the adaptive weight vector for user j is provided in [46], [51]

$$\mathbf{w}_j = \Lambda_j \mathbf{R}^{-1} \mathbf{d}_j, \qquad \text{where} \quad \mathbf{R} = \mathbf{d}_1 \mathbf{d}_1^\dagger + \mathbf{d}_2 \mathbf{d}_2^\dagger + \sigma^2 \mathbf{I}. \tag{4}$$

Note that $\mathbf{R}$ is a synthesized cross-spectral density matrix (CSDM) exploiting knowledge of the channel responses at the receiver array and Λ_j is a normalization constant such that $\Lambda_j^{-1} = \mathbf{d}_j^\dagger \mathbf{R}^{-1} \mathbf{d}_j$ (i.e., $\mathbf{w}^\dagger \mathbf{d} = 1$). σ^2 is a small diagonal loading factor included for matrix inversion since $N < M$ (i.e., pseudo uncorrelated noise power) and $\mathbf{I}$ is an identity matrix [49]. Since the rank of the matrix $\mathbf{R}$ is well defined (i.e., the number of users), a broad range of values can be used for diagonal loading [52].

Finally, the adaptive time-domain filter $w_i^j(-t)$ will replace the conventional time-domain filter $h_i^j(-t)$ in (2), resulting in a modified q-function

$$\tilde{q}_{mn}(t) = \sum_{i=1}^{M} h_i^m(t) * w_i^n(-t). \tag{5}$$

An example of $\tilde{q}_{mn}(t)$ versus $q_{mn}(t)$ is illustrated in [47, Figs. 4 and 5], confirming significant suppression of the crosstalk (off-diagonal panels).

B. Adaptive Time Reversal Versus Least Squares

Recently, ATR has been proven essentially equivalent to the LS solution of an overdetermined system [51]. The frequency-domain LS solution corresponds to the zero-forcing (ZF) detection technique in multicarrier MIMO-OFDM widely used in wireless channels [53]. Since the main objective of ATR in broadband, single-carrier communication systems is to suppress the CCI between transmitters or crosstalk between users [i.e., $\tilde{q}_{mn}(t) \approx 0$], the residual ISI still remains since $\tilde{q}_{nn}(t) \neq$

$\delta(t)$. Thus, ATR subsequently needs to be followed by a DFE (ATR-DFE) to enhance the performance, similar to its CTR counterpart (CTR-DFE or TR-DFE). In contrast, multicarrier MIMO-OFDM assumes that the channel frequency responses $H_i(f)$ are flat over the bandwidth of each subcarrier (i.e., no ISI) and no additional equalizer is applied.

For the three-user case over the 20-km range with 16-QAM in [50], it was found that as much as 6.5 dB per user can be improved by ATR-DFE [50]. Recently, using a larger bandwidth at higher frequency (11–19 kHz), an aggregate data rate of 60 kb/s (a spectral efficiency of 8) was demonstrated for three users/transmitters distributed over 4.2-m depth at 2.2-km range in a downslope shallow-water environment [52]. It is found that adding each user/transmitter degrades the performance by 3–4 dB on average with the benefit of a linear increase in data rate.

C. Interference Cancellation

Two other similar approaches subsequently have been proposed for CCI mitigation: 1) successive interference cancellation (SIC) [54]; and 2) parallel interference cancellation (PIC) [55], [56]. Unlike ATR, both approaches are applied to the single-channel time series after CTR multichannel combining [i.e., $q(t)$] to eliminate the CCI, but prior to the DFE, such that CTR-SIC-DFE and CTR-PIC-DFE, respectively. For time-varying channels, the two approaches can be implemented on a block-by-block basis along with channel reestimation [55]. Alternatively, Cho *et al.* [57] proposed a coupling of ATR with SIC and iterative processing (SIC-ATR-DFE) where the SIC is applied to each individual channel prior to the ATR.

D. Asynchronous Multiusers

A constraint in TR multiuser communications has been the requirement of some synchronization among the users. An initial estimate of the CIR from each user is assumed known to the receiver (or the base station) before decoding of data-bearing signals can commence, and are obtained through receptions of channel probes from each of the users in the absence of multiple-access interference (MAI). This MAI-free condition implies that the transmission of channel probes by users are organized into time slots by the base station through a reliable feedback channel (from the base station to the user). This scenario would require significant networking overhead and would be unfavorable in UWA channels, where the combination of small coherence times (typically much less than a second) and large propagation delays (typically much more than a second) discourages reliance on feedback and two-way communications. Consequently, the SIC-ATR-DFE approach, coupled with matching pursuit (MP) and iterative processing, has been extended to the case when users transmit asynchronously through a time-varying channel [58].

E. Multiuser Communication With Mobile Users

The multiple users/transmitters considered heretofore were stationary. Given the difficulties discussed in Section IV-E even for a single-user mobile case, a more general scenario with multiple users in motion would be quite challenging where the effect of different Dopplers must be taken into account. In such cases, the multiuser signals may be distributed across the Doppler dimension, and separation becomes a nontrivial task. When decoding any one user's Doppler corrected signal, the MAI removal process must incorporate the impact of Doppler correction on the MAI prior to cancellation.

Recently, Cho *et al.* [59] developed a receiver capable of separating overlapping packets from multiple, potentially mobile users transmitting to a shared base station. The receiver embedded an ATR processor within an iterative SIC framework, designed to decode each user in succession and form interference estimates to aid in the decoding of other users in the system. In comparison with other SIC architectures limited to systems with only stationary users, this architecture modifies the SIC process to track the effects of resampling on the MAI that would have been distributed across the Doppler dimension when the users experienced different mean Doppler shifts. With multiple iterations, the receiver was shown to be capable of separating a two-user packet collected during the Kauai acoustic communications experiment conducted off the western side of Kauai, HI, USA in summer 2011 (KAM11) [60] with one user moving and the other stationary. A block-wise implementation allowed the receiver to track the time-varying CIRs for each user, maintaining successful decoding throughout the 10.5-s packet and converging the SIC portion of the receiver within two iterations [59].

VI. Long-Range Communication in Deep Water

While research activities in underwater acoustic communication predominantly have been in shallow water, long-range acoustic communications in deep water has attracted attention in recent years in conjunction with the development of autonomous underwater vehicles (AUVs) or gliders [61]. In principle, the aforementioned TR approaches are equally applicable to communications in deep water. However, channel characteristics are quite different from shallow water, including a deep sound channel, convergence and shadow zones, depending on the geometry between the source and the receiver. In addition, low-frequency signals (e.g., below 100 Hz), as compared to typical telemetry frequencies (10–20 kHz) in shallow water, are used to propagate over thousands of kilometers in the deep ocean with less attenuation, such as those employed for acoustic tomography/thermometry over the past three decades [62].

Acoustic tomography typically has used maximal length sequences (m-sequence) as they have a good temporal cross-correlation property that yields a sharp peak in the time series, accurately marking the arrival time of the acoustic energy. The tomography signals also can be treated as a binary phase shift-keying (BPSK) communication signal [63]. Using the archive data collected during tomography experiments, Song *et al.* [36] demonstrated that an information rate of 37.5 b/s can be achieved at a 75-Hz carrier frequency over basin scale ($\sim$3250 km) which exploits either 1) spatial diversity provided by a vertical array ($M = 20$ elements) for a single transmission; or 2) temporal diversity (17 transmissions) provided by the time-varying ocean itself with a single receive element. The feasibility of global-scale acoustic communications also was explored between a moving source (57 Hz) and a single bottom-moored receiver over a ~9000-km range using the SAC concept described in Section IV-E [64]. Most recently, transarctic acoustic telemetry was demonstrated using two transmissions from a moored source (20.5 Hz) to an 8-element vertical array over a distance of ~2720 km in the ice-covered Arctic where either spatial (2 or 3 elements) or temporal (two transmissions) diversity was utilized [65].

A. Multiuser Communication

Over the last decade, the Japan Agency for Marine-Earth Science and Technology (JAMSTEC, Yokosuka, Japan) carried out a series of long-range acoustic communication experiments in deep water, south of Japan. The TR approach has been applied successfully to the experimental data collected at various ranges (10–1000 km) using BPSK modulation and achieved a data rate of up to 100 b/s utilizing a 100-Hz bandwidth (450–550 Hz) [66]–[68]. A recent experiment involving a source and a 18-element VRA both deployed at around the sound channel axis suggested that even a higher data rate (400 b/s) can be achieved over a 600-km range using a higher order constellation such as 16-QAM and TR-DFE [61].

Multiuser communications also have been investigated where the multiple users are spatially distributed in depth or range while a 114-m-long, 20-element VRA is deployed to around the sound channel axis (~1000 m). Signals received independently from ranges of 150 and 180 km at various depths were combined asynchronously to generate multiuser communication sequences for subsequent processing (ATR-DFE), achieving an aggregate data rate of 300 b/s for up to three users. In addition, two users separated by 3 km in range were able to transmit informations simultaneously to the base station at ~500-km range [47].

B. Communication With a Glider

The primary use of underwater gliders has been to collect oceanographic data within the water column (e.g., temperature and conductivity) and transmit their data to shore while periodically downloading instructions at the surface via a satellite connection. However, the mission of gliders has expanded lately into acoustics. Send *et al.* [69] reported the successful deployment of Spray gliders equipped with a commercial acoustic modem for data retrieval from subsurface moorings and seafloor systems installed with a similar modem in deep water. Several tests of acoustic communications between gliders and between gliders and other platforms have also taken place at a few kilometer ranges in shallow water to demonstrate the capability of gliders for global observation programs such as the Ocean Observatories Initiative (OOI) [70]. Separately, gliders equipped with an acoustic recording system (ARS) have been used for passive monitoring, for example, to collect marine mammal data during an experiment in Monterey Bay, CA, USA, in 2006 [71].

Gliders are in constant motion diving to depths of ~1000 m and traveling horizontally at about 0.25 m/s (half a knot) for several hours during a single dive. Thus, they naturally generate a combination of spatial and temporal diversity that can be utilized for long-range acoustic communication. Previously, similar spatial/temporal diversity was investigated for SAC [64] exploiting the relative motion between a source and a receiver where the moving source was confined to only horizontal motion (i.e., constant depth). The potential of implementing SAC with a single receiving hydrophone on a glider has been explored recently by Song *et al.* [72], using the data transmitted from the broadband tomography source (75 Hz) moored on the bottom off Kauai, HI, USA, while a glider moved away from the source at ranges up to ~200 km in deep water.

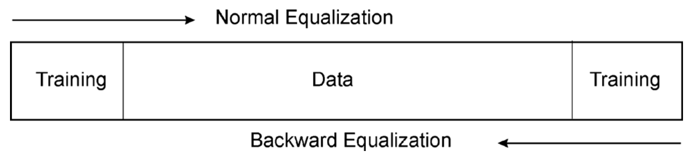

Fig. 5. Bidirectional equalization.

VII. Recent Advances and Future Work

A. Bidirectional Equalization

It is well known that the nonlinear DFE involving a feedback filter outperforms the LE in terms of reducing noise enhancement when the effect of decision errors on performance is neglected, especially for channels with spectral nulls [12]. However, the DFE suffers from error propagation caused by the feedback of incorrect decisions. In addition, the DFE based on the minimum mean squared error (MMSE) criterion is suboptimal from a probability of error viewpoint whereas an optimal approach is maximum-likelihood sequence detection (MLSE). The drawback of the MLSE is that it is prohibitively expensive to implement in most channels of practical interest.

In an effort to mitigate error propagation and improve the performance of a conventional DFE, a bidirectional DFE (BiDFE) was introduced in wireless channels for packet-based communication systems as illustrated in Fig. 5 [73], [74]. The BiDFE consists of two parallel DFE structures, one to equalize the received signal in a causal fashion and the other to equalize the time-reversed version of the received signal in a noncausal fashion. For example, a minimum-phase channel becomes a maximum-phase channel and *vice versa*. The approach thus exploits the difference in error propagation (a form of "diversity") between the forward and backward DFEs since the error burst proceeds in opposite directions with a low correlation for the errors [75]. Balakrishnan and Johnson [76] then proposed combining the two data streams rather than selecting one or the other and showed about 1-dB improvement based on simulations for symmetric and nonsymmetric channels.

The concept of a BiDFE combining the soft outputs of forward and backward DFEs has been extended to multichannel DFEs (i.e., Bi-M-DFEs and Bi-TR-DFEs) for underwater acoustic channels which are highly frequency selective (due to multipath) and time varying. It was demonstrated using experimental data (10–20 kHz) that the performance can be enhanced by 0.4–1.8 dB in terms of output SNR at the expense of doubling the training symbols. In particular, a larger improvement (e.g., 1.8 dB) was achieved for time-varying channels where the channel diversity in opposite directions was more profound [77].

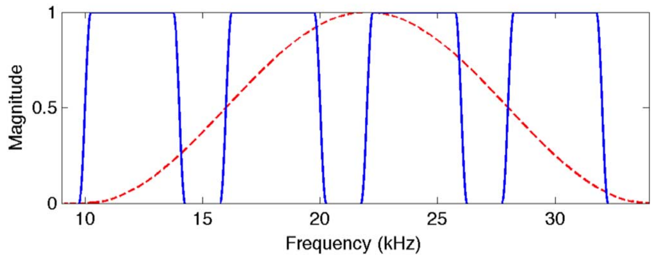

Fig. 6. Spectra of the raised cosine filters used as shaping pulses for communication signals (10–32 kHz) during KAM11: four multibands (solid) and a single wideband (dashed). Each subband width (solid) is 4.5 kHz. The excess bandwidth for each subband is 12.5% compared to 100% for the wideband signal. Reproduction of Fig. 2 in [78].

B. Bandwidth Efficiency

A simple way of achieving higher data rates is to increase the signal bandwidth (or transmission/signaling rate). For a given multipath channel, however, an increase in signaling rate will lead to larger ISI spans (measured in symbols), thus requiring longer channel equalizers. The extent of ocean multipath potentially can limit the system performance through large noise enhancement (i.e., self-generated noise) and increased sensitivity to numerical errors, along with increased receiver complexity [31]. Consequently, UWA communication systems frequently use modest bandwidths (e.g., few kilohertz), resulting in low transmission rates. In order to mitigate large ISI spans in a $\sim$100-m water region off the west coast of Kauai, HI, USA (KauaiEx) [79], Roy *et al.* [80] explored multiband transmissions where the available bandwidth (27–50 kHz) was divided into several subbands with guard bands in between. More recently, Song and Badiey [81] presented TR-based multiband communication results (TR-DFE) exploiting a large available bandwidth (10–32 kHz) at 3-km range during the KAM11 experiment (similar location as KauaiEx) as shown in Fig. 6 (solid line).

On the other hand, an interesting finding was reported in [82] based on M-DFEs that an increase in transmission rate yielded better system performance. A simple explanation was given to the time variability of the channel. That is, a smaller symbol duration with high signaling rate enables better channel/phase tracking since the channel remains coherent over a large number of symbols at the price of increased complexity. A similar finding also was reported by Song *et al.* [83], [52] using various data sets in terms of channels, constellations, and bandwidths that indeed the performance can improve with an increase in symbol rate. However, the channel was relatively time invariant over the duration of data packets (3 s).

Fortunately, during the same KAM11 experiment in [81], wideband single-carrier communication signals were also transmitted, providing a rare opportunity to compare two different approaches (multiple bands versus a single wideband) under similar environmental conditions using the same hardware (both source and receiver) [78]. Although each approach was not optimized for its best performance, it was found that a single wideband approach (Fig. 6, dashed line) can be beneficial in terms of spectral efficiency with modest computational complexity using a TR-DFE. A plausible explanation was that an increase in diversity with the use of wideband transmissions enables resolving more multipath components, i.e., broadband diversity [12]. From the TR perspective, the resulting increase in channel complexity leads to an improved q-function behavior, resulting in performance enhancement.

C. Effects of Ocean Environments on Performance

The performance of acoustic communications in the ocean is greatly affected by the dynamics of the propagation medium through the time-varying CIR and the intensity of received signals (i.e., SNR). The dynamic ocean processes include surface wave movement and fluctuations within the medium itself such as internal waves. For example, solitary waves are shoreward traveling nonlinear waves, commonly generated by tidal flows over sharp bathymetric variations [33]. They have length scales of tens of meters and periods as short as a few minutes, affecting the spatial and temporal coherence of the acoustic signal. Therefore, it is important to understand the correlation of communication performance and the surrounding ocean environment for reliable operation, performance prediction, and, if possible, optimal placement of sources and receivers.

There have been a few efforts to study the impact of ocean variability on phase coherent acoustic communications. Preisig and Deane [84] focused exclusively on the surface gravity waves in the surf zone and revealed that the acoustic focusing by the curvature of the wave crest presents challenges to tracking the intense, rapidly fluctuating CIR. On the other hand, Carbone and Hodgkiss [33] compared the performance in 100-m water depth off the coast of San Diego, CA, USA, between when solitary waves are absent and present. To isolate the environmental variations in the water column, a source (18 kHz) and receiving arrays separated by 6-km were moored near the bottom in a downward-refracting environment. Recently, Song *et al.* [85] examined communication performance in the midfrequency band (813 and 1627 Hz) during the passage of a packet of strong internal waves over a 12-h period crossing an approximately 20-km acoustic track.

To look at the long-term impact of ocean variability, Song *et al.* [35] analyzed KauaiEx data [79] for a near-seafloor source (8–16 kHz) collected over 27 h. The transmissions involved both significant interaction with surface waves and water column changes from a well-mixed to downward-refracting environment and were processed using a TR-DFE. In the same area (KAM08), the impact of source depth on the communication performance (10–20 kHz) also was investigated for various source depths and receiver subarrays during an extended period (35 h). In the aforementioned cases, the pseudorandom noise sequence (m-sequences) was treated as a BPSK communication signal to evaluate the performance.

A recent example of long-term communication performance using TR-DFEs (KAM11) is shown in Fig. 7 for various source depths in the same area as KauaiEx and KAM08. Of particular interest is the use of wideband signals (10–32 kHz) and QPSK modulations for two full days. The 10-s-long communication sequence was transmitted every 2 h from a stationary 8-element source array to a 16-element vertical receive array covering two-thirds of the water column at a 3-km range. Note that the communication performance can change as much as 10 dB in just a few hours for the same hardware and fixed geometry

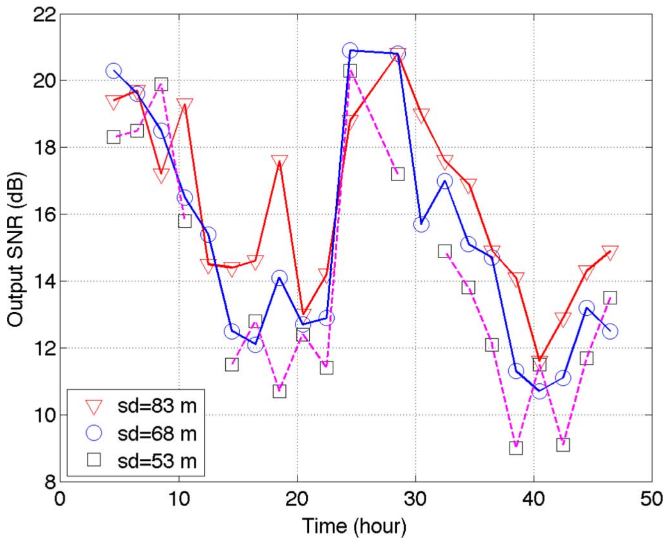

Fig. 7. Performance of QPSK communications at a 3-km range in 100-m deep water for various source depths over a two-day period during the KAM11 experiment [60].

due to the environment-induced fluctuations. There appears to be some periodicity (e.g., tidal period) involved, but the two-day observation alone has not been long enough to draw any conclusions. While there were concurrent oceanographic measurements including thermistor strings and a waverider buoy, it has been quite challenging to fully understand the performance behavior as related to the fluctuating ocean environment. Accordingly, better understanding of the coupling of oceanography, acoustics, and communications will be important for reliable operation and performance predication in dynamic ocean environments.

VIII. Conclusion

TR communications first proposed in the early 2000s have evolved substantially over the last decade as an alternative to conventional multichannel equalization (M-DFE). TR exploits spatial diversity to achieve spatial and temporal focusing, directly applicable to communications in complex ocean environments. The original concept, however, was limited to two-way, active, downlink MISO communications in time-invariant environments. In addition, temporal focusing was not perfect with residual ISI, resulting in performance degradation and saturation. Consequently, TR processing was followed by single-channel equalization to remove the residual ISI (TR-DFE), providing nearly optimal performance. TR is equally applicable to one-way, passive, uplink SIMO communications of practical interests with the same performance. The benefit of passive TR or PPC has been its capability of handling time-varying channels simply by using a block-based approach with channel updates. Spatial focusing enabled multiuser/MIMO communications and subsequently was enhanced by ATR. The block-based TR-DFE has been demonstrated successfully in both shallow and deep water for various frequencies (from 50 Hz to 30 kHz) with different bandwidths (from a few hertz to 20 kHz) and distances (a few to thousands of kilometers), exploiting spatial or temporal diversity or both and achieving a high spectral efficiency. While many advances have been made in algorithm developments, this review has suggested that there is ample research to be done with an emphasis on the coupling of oceanography, acoustics, and communications for design and performance characterization of underwater acoustic communication systems in dynamic ocean environments.

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