Time reversal communication with a mobile source (L)

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Broadband underwater acoustic communication signals undergo either a compression or dilation in the presence of relative motion between a source and a receiver. Consequently, underwater acoustic communications with a mobile source/receiver require Doppler compensation through resampling. However, resampling may not be necessary when a channel-estimate-based time reversal approach is applied with frequent channel updates. Using experimental data (20–30 kHz), it is demonstrated that the performance of time reversal communication without resampling is similar to the case with resampling, along with the benefit of a modest computational saving.

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I. INTRODUCTION

Over the last decade time reversal communication has emerged as an alternative to conventional adaptive multichannel decision-feedback equalizers (DFEs) in time-varying, multipath underwater environments. Time reversal (TR) exploits spatial diversity to mitigate intersymbol interference (ISI) and provides near-optimal performance in conjunction with channel equalization that removes the residual ISI. Consequently, multichannel TR combining followed by a single channel DFE (TR-DFE) is routinely employed and further extended to time-varying channels using a block-based approach with frequent channel updates. Since knowledge of the channel is incorporated, TR-DFE can be viewed as a channel-estimate-based DFE. Separately, Stojanovic et al. proposed optimal beamforming and equalization to reduce the complexity of multichannel equalization from a $K$-dimensional element space to a smaller $P$-dimensional beam space. The TR approach then corresponds to a special case where $P = 1$ without sacrificing performance.

Much of the published work on underwater acoustic communications involves a stationary source and receiver in fluctuating ocean environments. In the presence of relative motion between a source and a receiver, broadband underwater acoustic communication signals experience a Doppler compression or dilation, which depends on the propagation path. The differences in the compression or dilation will lead to a Doppler spread in a multipath ocean environment. The motion and geometry typically are constrained in shallow underwater channels: (1) the motion is uniform with a constant speed and horizontal 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 is compensated by resampling the received signal. On the other hand, the residual Doppler for different paths collectively leads to a time-varying channel impulse response (CIR), even in the absence of environmental fluctuations in the medium such as surface waves, internal waves, etc., at various time scales. The resampling approach has been applied successfully to mobile underwater communications in the literature.

The motivation of this Letter is to explore the possibility of removing the resampling process in channel-estimate-based TR communications by constraining the block size for channel updates. Section II reviews the basics associated with Doppler due to a source moving in free space and then discusses a source moving in an oceanic waveguide based on the normal mode theory. Using the experimental data (20–30 kHz) described in Sec. III, the performance of TR communications with resampling is compared to the case without resampling in Sec. IV. The concluding remark is provided in Sec. V.

II. DOPPLER COMPENSATION: RESAMPLING

The acoustic field generated by a moving source in a waveguide is not trivial even for a monochromatic continuous wave (CW) signal due to the multipath phenomena. We will use the normal-mode representation of the field due to a CW point source developed by Hawker. A thorough derivation based on the spectral representation can be found in Ref. 12.

Using the analytic signal representation, a transmitted communication signal $x(t)$ in the passband with a carrier frequency $f_c$ is related to the complex signal $\tilde{x}(t)$ at baseband

$$ x(t) = \text{Re}\{\tilde{x}(t)e^{j2\pi f_c t}\}. \quad (1) $$

A. Free space

Figure 1 depicts a source and a receiver in a free space. The source denoted by $S$ moves with a constant speed $v$ along a linear path and an angle $\theta$ defined at time $t = 0$. Let us denote $R_0$ as the source and receiver separation at time $t = 0$, the center of the observation interval $T$. Then the instantaneous source/receiver separation $R(t)$ is parameterized as

$$ R(t) = (R_0 + v^2t^2 + 2vtR_0 \sin \theta)^{1/2}, \quad t \in [-T/2, T/2]. \quad (2) $$

A convenient parametric representation of $R(t)$ is the Taylor series
B. Oceanic waveguide

In a waveguide, the acoustic field generated by a moving source can be derived from normal mode theory,\textsuperscript{11} assuming that the medium is horizontally stratified with the sound speed profile being only a function of water depth, \( c(z) \). Similar to Sec. II A, a point source is assumed to move along a linear path with a constant speed \( v \) at a fixed depth \((z_0)\), while a stationary receiver is located at a different depth \((z_r)\). In this case, Fig. 1 can be interpreted as a two-dimensional horizontal plane \((x,y)\) onto which both the source and receiver in a waveguide are projected (i.e., top view).

\( R(t) \) then denotes the horizontal distance from the source to the receiver along the projected source path at time \( t \). The benefit of the normal mode representation is enabling each mode (or path) to travel the same horizontal distance \( R(t) \), but with a mode-dependent horizontal propagation speed \( c_p \), which is determined by the discrete horizontal wavenumber (i.e., modal eigenvalue), \( k_p = 2\pi f_p/c_p \). In turn, each mode generates a distinct Doppler effect corresponding to \( \beta_p = v/c_p \), referred to as a modal Doppler shift.\textsuperscript{15} The extent of a Doppler spread will depend on the spreading of the eigenvalues \( k_p \) or \( c_p \). Using the ray-mode analogy, each mode has a different propagation angle \( \phi_p \) with respect to the horizontal through \( k_p = k(z) \cos \phi_p \). For a geometry where the range is much greater than the water depth (i.e., far-field), all significant rays (modes) arriving at a receiver will have a small angular spread and can be represented by a single (mean) Doppler shift \( \beta \).

While resampling by a single Doppler parameter \( \beta \) mostly achieves Doppler compensation, the residual Doppler \( \Delta \beta_p = (\beta - \beta_p) \) in the phase for each path will lead collectively to a time-varying CIR in a multipath ocean environment. Thus, the use of TR in mobile communications requires periodic channel updates after Doppler compensation to mitigate the mismatch in CIR, even in a stable ocean environment.\textsuperscript{10} Regardless of the source motion, however, channel updates usually are necessary to accommodate channel fluctuations induced by time-varying ocean environments. This is especially true of the high-frequency waveforms commonly used for acoustic telemetry (e.g., 10–20 kHz).

On the other hand, it is well known that in the frequency domain TR (or phase conjugation) corresponds to an unnormalized version of matched field processing (MFP) for source localization in an oceanic waveguide.\textsuperscript{16,17} In MFP, source motion leads to a mismatch between the assumed (fixed) and actual (moving) acoustic fields, degrading the localization performance. Based on the generalized ambiguity function analysis, however, Song\textsuperscript{18} suggested that the impact of source motion can be minimal provided the observation time window \((T_{obs})\) in conjunction with the radial speed \((v_r)\) does not exceed about half the horizontal wavelength \((\lambda_p/2)\), i.e., \( v_r T_{obs} \leq \lambda_p/2 \). In other words, a moving source can be treated as quasi-stationary over half the horizontal wavelength in radial travel distance. It is reasonable to assume that the horizontal wavelength can be approximated on the order of a wavelength \((\lambda)\) when taking into account the mode propagation angle \( \phi_p \) which depends on the source/receiver geometry and ocean environment.\textsuperscript{12} The implication for TR

\[ R(t) = R_0 + (v \sin \theta) t + \frac{v^2 \cos^2 \theta t^2}{2!} + \cdots. \] \hspace{1cm} (3)

Assuming a small observation interval \( T \), the above expression can be truncated to the linear term, neglecting the second-order radial acceleration. Then the receiver observes the radial component of the source velocity, \( v_r = v \sin \theta \). Positive values of \( v_r \) denote motion away from the receiver, while negative values denote motion toward the receiver.

The time variation of the propagation time \( \tau(t) \) due to source motion becomes

\[ \tau(t) \equiv \frac{R(t)}{c} = \tau_0 + \frac{v_r}{c} t, \] \hspace{1cm} (4)

where \( \tau_0 = R_0/c \) is the propagation delay at time \( t = 0 \) and \( c \) is the medium sound speed. Define \( \beta = v/c \ll 1 \) as a Doppler parameter or factor and ignore the time-independent term \( \tau_0 \) for convenience. Then the time-delayed received signal \( r(t) \), excluding the noise contribution and attenuation, can be expressed from Eq. (1),

\[ r(t) = x(t - \tau(t)) = \text{Re}\{\tilde{x}((1 - \beta)t)e^{-j2\pi\beta t} e^{j2\pi\beta t}\}. \] \hspace{1cm} (5)

Note that the source motion affects the received signal in two ways through \( \beta \). First, the carrier frequency \( f_c \) is changed by the amount of \( f_{d} = \beta f_c \), known as a Doppler frequency shift. Second, more importantly, the Doppler effect compresses or dilates the original complex baseband signal such that \( \tilde{x}((1 - \beta)t) \) is observed by the receiver. The process of undoing compression or dilation to recover \( \tilde{x}(t) \) is referred to as resampling, usually implemented by an efficient finite impulse response (FIR) polyphase filter and linear interpolation.\textsuperscript{8} In summary, Doppler compensation requires both modified carrier frequency shifting and resampling.

By limiting the observation interval \((T)\), Doppler is a linear modulation as evident in Eq. (3). A typical packet size for acoustic telemetry (e.g., a few s) satisfies this condition in shallow water environments. In the literature\textsuperscript{2,13,14} a simple extension to a multipath environment is that several scaled and delayed copies of \( \tilde{x}(t) \) will contribute to the received field when each copy is compressed or dilated by a different Doppler parameter \( \beta \).

FIG. 1. (Color online) Source and receiver geometry in free space or viewed from the top in an oceanic waveguide. The source is moving at a constant speed \( v \) along the linear path with respect to a stationary receiver.
communications is that resampling may not be necessary when frequent channel updates are employed at intervals comparable to a wavelength period in travel distance.

III. KAM11 EXPERIMENT

The KAM11 experiment was conducted off the western side of Kauai, HI, in a roughly 100-m deep water, from June 23 to July 12, 2011. Both fixed and towed source transmissions were carried out to multiple receiving arrays over ranges of 1–8 km along with additional towed source transmissions. In this Letter, we analyze the data captured during a towed-source run on July 8 (JD189 164700).13

The experimental geometry is illustrated in Fig. 2(a). A vertical receiving array (VRA) was moored in 106-m deep water. The VRA consisted of 16 elements spanning a 56.25 m aperture with 3.75 m element spacing, covering the middle and lower portion of the water column in a downward-refracting environment. A source at 35-m depth was towed at a radial speed of approximately 1.1 m/s away (southward) from the VRA at a distance of 1.3 km at the time of transmission (16:47:00).

The transmitted signal was a BPSK (binary-phase shift-keying) communication sequence using the 20–30 kHz frequency band with a carrier frequency of 25 kHz. The symbols were shaped with a root-raised cosine filter with a roll-off factor of 1. The packet size was \( T = 10.5 \) s with a symbol rate of \( R = 5 \) ksym/s. The dominant Doppler shift was estimated \( f_\text{d} = -19 \) Hz with \( \beta = 7.6 \times 10^{-4} \) while the Doppler varies slightly across the VRA.13 The same Doppler shift will be applied to all receive elements for Doppler compensation.

The block diagram for Doppler compensation is illustrated in Fig. 3 involving two steps separated by a vertical line. First, the real passband signal at each element of the VRA \( r(t) \) is demodulated to a complex baseband signal \( \tilde{r}(t) \) using a modified carrier frequency \( 1 - \beta \tilde{r}_c \), followed by a low-pass filter. The complex baseband signal \( \tilde{r}(t) \) then is resampled appropriately to compensate for Doppler dilation or compression, generating a resampled baseband signal \( \tilde{r}_{rs}(t) \).

An efficient FIR polyphase filter\(^8\) followed by a linear interpolation is employed for resampling. The key question in this Letter is whether one can use the complex baseband signal \( \tilde{r}(t) \) instead of \( \tilde{r}_{rs}(t) \) for communication with a moving source.

IV. PERFORMANCE COMPARISON

As a baseline, a block-based TR-DFE is applied first to the resampled data \( \tilde{r}_{rs}(t) \) along the VRA. The LMS (least mean square) algorithm is employed for adaptive channel estimation while the RLS (recursive least squares) algorithm is used for the adaptive DFE. The block size was chosen \( N_B = 150 \) symbols corresponding to a 30 ms time window, minimizing a potential mismatch in the CIR. The number of training symbols was \( N_T = 500 \) used for both channel estimation and equalization. The initial channel estimate using the training symbols across the VRA is shown in Fig. 2(b), indicating about a 20 ms delay spread spanning \( L = 100 \) symbols. Then the temporal evolution of the CIR estimate in the TR-DFE is displayed in Fig. 4 (bottom row) using previously detected symbols (decision-directed mode) at three representative array depths (bottom, middle, and top): Ch. 1 (95.4 m), Ch. 9 (66.4 m), and Ch. 16 (40.2 m). Clearly, the source motion is compensated with straightened arrivals due to resampling. The communication performance was 8.7 dB in terms of output SNR and almost error-free. In addition, a bidirectional TR-DFE was applied to further improve the performance,\(^9\) yielding an error-free output SNR of 10.4 dB (almost 2 dB increase). A fractionally spaced DFE (two samples per symbol) was adopted for the DFE feedforward filter, and the number of feedforward and feedback filter taps were 60 and 30, respectively. The LMS step size was \( \Delta = 0.004 \) and the RLS forgetting factor was \( \zeta = 1 - \Delta = 0.996 \).

Now, the block-based TR-DFE is applied to the complex baseband data \( \tilde{r}(t) \) prior to resampling (see Fig. 3). For comparison purposes, all parameters are kept identical to the above baseline case. Over the \( T_{\text{obs}} = 30 \) ms time window \( (N_B = 150) \), the source travels about 3 cm at the radial speed of 1.1 m/s, which is half the wavelength (6 cm) at the 25 kHz carrier frequency. The temporal evolution of the CIR...
estimates without resampling is shown in Fig. 4 (top row), displaying slanted arrivals as opposed to the counterparts after resampling (bottom row). The performance for conventional and bi-directional TR-DFE was 8.8 dB and 10.4 dB, respectively. Not surprisingly, the performance achieved without resampling is similar to the one with resampling, confirming the quasi-stationarity of a moving source over a limited time window. It should be mentioned that an increase in block size almost linearly degraded the performance from 10.4 dB (N_B = 150) down to 9.5 dB (N_B = 350) until it broke down abruptly when N_B = 400 corresponding to about 9 cm in travel distance. The computational time for the block-based TR-DFE without resampling was about 60 s on a desktop, shorter than about 90 s in the baseline case involving additional resampling. Note that explicit phase tracking was not required due to a small block size in TR-DFE regardless of resampling.

V. CONCLUSIONS

In this Letter, the notion that broadband acoustic communication with a mobile source requires Doppler compensation through resampling has been challenged using the block-based TR-DFE with frequent channel updates. Analogous to TR in the frequency domain, in MFP it was suggested that the impact of source motion can be minimal provided the observation time window (block size) is constrained to about a wavelength in terms of a travel distance. Using experimental data (20–30 kHz) collected during a recent shallow water experiment (KAM11), it was demonstrated that indeed the performance of TR communication without resampling was similar to the one achieved with conventional resampling. Further, the similar performance was achieved with the benefit of a modest computational saving.

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