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A Perspective for Time-Varying Channel Compensation with Model-Based Adaptive Passive Time-Reversal

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Abstract: Underwater communications mainly rely on acoustic propagation which is strongly affected by frequency-dependent attenuation, shallow water multipath propagation and significant Doppler spread/shift induced by source-receiver-surface motion. Time-reversal based techniques offer a low complexity solution to decrease interferences caused by multipath, but a complete equalization cannot be reached (it saturates when maximize signal to noise ratio) and these techniques in conventional form are quite sensible to channel variations along the transmission. Acoustic propagation modeling in high frequency regime can yield physical-based information that is potentially useful to channel compensation methods as the passive time-reversal (pTR), which is often employed in Digital Acoustic Underwater Communications (DAUC) systems because of its low computational cost. Aiming to overcome the difficulties of pTR to solve time-variations in underwater channels, it is intended to insert physical knowledge from acoustic propagation modeling in the pTR filtering. Investigation is being done by the authors about the influence of channel physical parameters on propagation of coherent acoustic signals transmitted through shallow water waveguides and received in a vertical line array of sensors. Time-variant approach is used, as required to model high frequency acoustic propagation on realistic scenarios, and applied to a DAUC simulator containing an adaptive passive time-reversal receiver (ApTR). The understanding about the effects of changes in physical features of the channel over the propagation can lead to design ApTR filters which could help to improve the communications system performance. This work presents a short extension and review of the paper [12], which tested Doppler distortion induced by source-surface motion and ApTR compensation for a DAUC system on a simulated time-variant channel, in the scope of model-based equalization. Environmental focusing approach in high frequency underwater acoustics is intended to be explored in future, based on the idea that a set of oceanic acoustic physical parameters – which are generally estimated in well-known low frequency matched field processing problems like geoacoustic assessment, ocean tomography and source localization – could be conveniently used on adaptive filters for channel compensation in DAUC systems. *Copyright © 2015 IFSA Publishing, S. L.*

Keywords: Digital acoustic underwater communications, Passive time reversal, Coherent signaling, Environmental-Based channel compensation, Matched field processing, Time-Variant channel modelling.

1. Introduction

High data rate throughput in Digital Acoustic Underwater Communications is a challenging subject studied by the scientific community, finding applications in point-to-point (P2P) communications and underwater acoustic networks (UAN) used for, *e.g.*, offshore activities and oceanic research observatories [1]. The complicated acoustic propagation in ocean waveguide channels makes the high rate communications between two node sensors become a hard task, mainly because of multipath propagation and Doppler distortion, often requiring equalization. Equalization often uses channel impulse response estimates from an initial set of probe symbols (training sequence) to reach by use of least squares based algorithm the filter coefficients that compensates time-frequency distortion imposed by the waveguide during a transmission, then mitigating ISI and improving communications performance. However classical algorithms for equalization, *e.g.*, linear equalizers and decision feedback equalizers demand high complexity and sometimes are vulnerable to mathematical errors related to convergence, especially in case of long frame signals transmitted along shallow water waveguides. In this sense, passive time-reversal (pTR) processing described in next lines is an attractive choice given its low complexity and reasonable robustness, even though it yields lower accuracy (it is designed to find least signal-to-noise ratio, not reaching complete equalization), which is a drawback when compared to the earlier case.

Any channel estimation technique must deal with frequency selective attenuation, time dispersion from multipath propagation and frequency dispersion due to Doppler Effect. Coherent signaling is required for high data throughput because of its improved bandwidth efficiency, a desired characteristic in the DAUC system design considering the bandwidth limitation imposed by the ocean channel. Additionally, one can note that conventional equalizers do not use acoustic physical parameters of the ocean channel, at least in a direct form. This is not specifically a design requirement, but at first look one could expect to explore more incisively those parameters, because they rule the physics of propagation in the channel. This work is part of an investigation to deploy adaptively the passive time reversal technique aiming to reach channel compensation in P2P wireless underwater communication using single-input-multiple-output (SIMO) structure.

Passive time-reversal (pTR) – and its frequency domain version named passive phase conjugation (PPC) [2] – is inspired in active time-reversal (TR) which compensates the channel exploring the retro-focusing propagation property of the wave equation in wave-guide. The TR concept was first demonstrated in underwater acoustics by [3]. In its passive version the processing can be performed synthetically, exploring spatial and temporal diversity of the acoustic channel by software setup. Nonetheless, the pTR equalizer requires to estimate the channel impulse response (CIR) which is usually obtained by an initial channel probe, being unable to track for channel variability along the data transmission. The ApTR is designed to overcome this problem through the employment of time-varying approach for filtering and channel modeling. Further, the environmental focusing method is discussed based on the possibility of use information from time-variant channel estimates and acoustic propagation modeling to perform cross-correlation between the observed channel and a carefully chosen data bank of replicas. The last could be built from model-based filtering that deploys distorted versions of the initial probe based on rules incorporated from the effects observed of distortion tests for acoustic propagation physical parameters.

Section II describes the theoretical background for time-variant DAUC channel modelling. Section III presents the ApTR, including a preliminary test in simulated ocean waveguide channel, and discuss the environmental focusing method. Discussion and conclusion are presented in Section IV.

2. Theoretical Background

2.1. CIR Modelling for DAUC

Acoustic propagation modeling in shallow water must considers occurrence of multiple paths, causing channel impulse responses characterized by delayed arrivals and generating ISI that imposes errors in demodulation. Additionally, realistic scenarios require to do assumptions about ocean dynamics and relative motion between source and receiver, since they cause time-frequency dispersion and demand for compensation of Doppler distortion. In time-invariant channel approach for DAUC, beam trace acoustic propagation model is the most adequate among other choices based in full field models because beam trace allows to range-dependency, have fast computation in high frequency regime and yields results with acceptable accuracy given the use of infinite-frequency approximation from ray theory for the Helmholtz equation solution. Nonetheless, these models rely on the assumption of time-invariance, which goes against the fact that there are naturally temporal variations in ocean channel whose time scale can be quite significant in high frequency underwater acoustics and severely affect DAUC systems performance. Becomes important to employ a time-variant approach aiming to reach more realistic CIR modeling for channel compensation.

The work of [4] presented the Time-Variable Acoustic Propagation Model (TVAPM) that can simulate underwater acoustic propagation in time-variant channel using sequential runs of the Bellhop model [5]. The set of sequential runs of the ray model aims to build a matrix that provides information about the time-variant CIR and implements Doppler distortion to simulate the received signal. The fundamental assumptions to dynamic propagation simulations with TVAPM are: source and array can be placed anywhere in a three-dimensional box limited by specified bathymetry and by a pressure release surface which can include a wind-driven surface model (Jonswap spectrum model was used in this work); a transmitted signal is specified as input; linear velocities can be attributed to both source and array and the corresponding positions are updated progressively along transmissions. This model was used in the present work to perform time-variant CIR simulations and link them to a coherent DAUC model containing the ApTR equalizer. Fig. 1 shows the Eigen rays structure of acoustic propagation in a simulated ocean waveguide computed by the Bellhop model for one run of a set of time-invariant runs computed along the TVAPM processing, therefore representing a snapshot observation of the time-variant simulated channel.

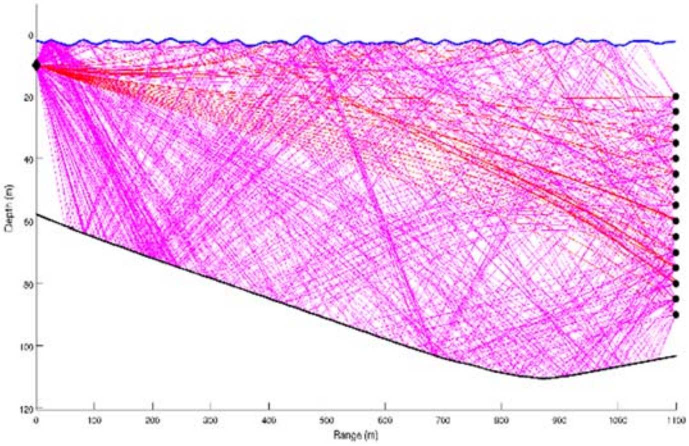

Fig. 1. Trace of Eigen rays in a static waveguide computed with time-invariant acoustic propagation model (Bellhop).

Each sensor of the vertical array captures the transmitted signal convolved by channel impulse responses, yielding distorted signals whose distortion are ruled mainly by the boundaries conditions and sound speed profile in water column of the underwater waveguide. The channel impulse responses have beyond a main arrival, other delayed arrivals with specific amplitudes. Fig. 2 shows the delayed arrivals caused by multipath propagation as function of depth.

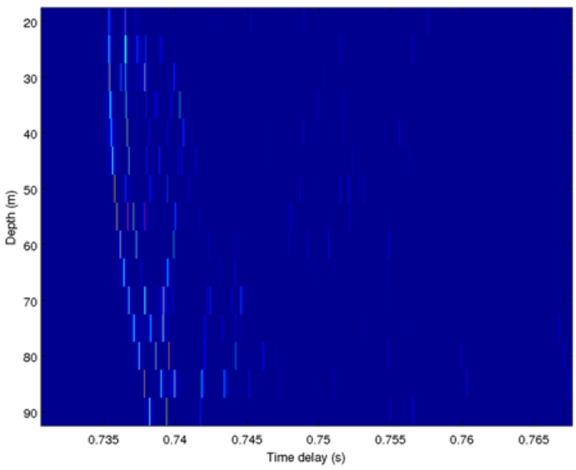

Fig. 2. Color plot for impulse responses along depth characterized by delayed arrivals caused by multipath propagation.

2.2. Doppler Distortion in Wave-guide

The Doppler distorted signal can be computed for each path of the multipath propagation through the insertion of a compression factor β_p . This factor performs compression/expansion in time and frequency shift in the signal (with compression/expansion in the corresponding spectrum). Fig. 3 shows the bi-static scattering geometry scenario [6], where the signal transmitted by a moving source (T) is reflected in a moving point of scatterer (S) and received in a moving receptor (T). The velocity vectors for the transmitter, receiver and point scatterer are $\vec{v}_t$, $\vec{v}_r$ and $\vec{v}_s$ respectively; $\hat{n}_{t,p}$ and $\hat{n}_{r,p}$ are the unity vectors in direction of propagation of the transmitted and received waves, respectively, for each path.

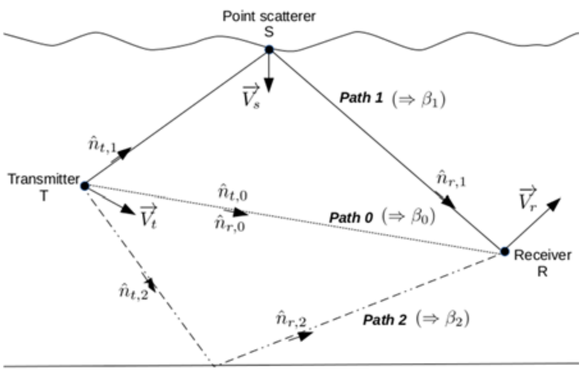

Fig. 3. Bi-static scattering geometry (full line and dash-dot line) and mono-static geometry (fine dot line).

The time compression factor β_p for a single path in the bi-static scattering geometry (paths 1 and 2) is given by [6]:

$$\beta_p = \frac{(1-\vec{v}_s \cdot \hat{n}_{t,p}/c)(1-\vec{v}_r \cdot \hat{n}_{r,p}/c)}{(1-\vec{v}_s \cdot \hat{n}_{r,p}/c)(1-\vec{v}_t \cdot \hat{n}_{t,p}/c)} - 1, \tag{1}$$

where c is the sound speed in water, symbol denotes dot product and $\vec{v}_x \cdot \hat{n}_{x,p}$ represents the projection of a given velocity vector ($\vec{v}_t$, $\vec{v}_r$ or $\vec{v}_s$) in the ray path direction. The same expression is valid to computations in mono-static geometry (path 0), just considering that $\vec{v}_s$ is null and $\hat{n}_{r,0}$ is equal to $\hat{n}_{t,0}$. The compression factors (β_0 for the direct path, β_1 for the free surface reflected path and β_2 for the bottom reflected path) represent the compression (or dilation) suffered by the signal when travelling to the receiver through each path.

Consider a transmitted band pass signal $\tilde{s}(t)$ with carrier angular frequency ω_c and low-pass equivalent signal $s(t)$ containing the information bit sequence $a(n)$ shaped by the pulse shape $p(t)$ sampled at the symbol interval T_s, as follows

$$\tilde{s}(t) = \mathrm{Re}\{s(t)e^{j\omega_c t}\} \Leftrightarrow s(t) = a(n)p(t-nT_s) \tag{2}$$

The Doppler distorted band pass signal $\tilde{s}_D(t)$ is the sum of the distorted path signals, which consider the Doppler compressional factor β_p in each path p to perform time compression/expansion and frequency shift

$$\tilde{s}_D(t) = \sum_p \mathrm{Re}\{s((1+\beta_p)t)e^{j\omega_c(1+\beta_p)t}\} \tag{3}$$

Using baseband equivalent notation, the received noiseless signal for a single path can be represented as the convolution of the path distorted signal $s_D(t)$ with the single path impulse response $g_p(\nu)$. Performing algebra manipulation, the time-variant channel impulse response in the $p-th$ path is given by [2]

$$h_p(t,\mu) = \frac{1}{1+\beta_p} g_p\left(\frac{\mu+\beta_p t}{1+\beta_p}\right) e^{j\omega_c \frac{\beta_p}{1+\beta_p}(t-\mu)} \tag{4}$$

with

$$y_p(t) = \int s(t-\mu)h_p(t-\mu)d\mu \tag{5}$$

And in the $i-th$ hydrophone, the time-variant CIR is

$$h_i(t,\mu) = \sum_p h_{p,i}(t,\mu)\delta(t-\mu_{p,i}) \tag{6}$$

The g-function (composed by path component functions g_p) was numerically computed with the acoustic propagation model using Eigen rays search over the beam trace approach solution of the Helmholtz equation. However, in terms of illustrative compact representation only, it is straightforward to show bellow the compact solution reached by normal modes model for a simplified Pekeris waveguide [14], since it represents at some degree an analogous approach to reach an approximate solution for g-function (but with completely different methodology). The point is the advantage in represent approximately the physical process in a very compact form. The frequency domain solution is

$$G(\vec{r},\omega) = \frac{-i\rho\omega^2}{2D}\sum_{p=1}^{P} a_p(k_{r,p})\sin(k_{z,p}z)\sin(k_{z,p}z_s)H_0^{(1)}(k_{r,p}r), \tag{7}$$

where ω denotes angular frequency, r is horizontal range, ρ is water density, D is depth, a_p is a modal amplitude, $H_0^{(1)}$ is the Hankel function of first type, z is the receiver depth, z_s is the source depth, k_z is the modal vertical wavenumber and k_r is the modal horizontal wavenumber. Applying to Equation (7) the inverse Fourier transform yields an approximation of the time domain g-function for the Pekeris waveguide case, where its mode decomposition gives the g_p sub-function corresponding to each path, as showed in Equation (4).

3. APTR Processing

The passive time-reversal technique performs for each channel the correlation between the reverse-conjugated estimated CIR and the observed CIR. The sum over the channels will yield a function that allows for a straightforward performance analysis of pTR based equalizers.

Considering a time-variant channel impulse response $h_i(t,\mu)$ and its estimate $\hat{h}_i(t,\mu)$, where i is the hydrophone index of a vertical line array, t denotes time and μ denotes delay, the $Q(t,\mu)$-function is the summation along the array of the cross-correlation function between the CIR and the corresponding estimate, as follows

$$Q(t,\mu) = \sum_i \int \hat{h}_i^*(t,-\nu)\hat{h}_i(t,\mu-\nu)d\nu \tag{8}$$

For the time-invariant case, since the CIR does not vary along time, the variable t can be suppressed. Considering the usual assumptions of pTR, that the sensor array spans the most energetic area of the water column for the normal mode orthogonality
to hold,

$$Q(t=0,\mu) \cong \delta(t=0,\mu) \tag{9}$$

the pTR output signal is given by

$$z(t)=\sum_{i=1}^{L} z_i(t)=\int Q(\mu)I(t-\mu)d\mu, \qquad (10)$$

where $I(t)$ contains the information data signal $a(t)$ and the auto-correlation of the pulse shape $p(t)$ used in the transmitted signal and the receiver, as follows

$$I(t)=\int a(\mu)R(t-\mu)d\mu \qquad (11)$$

and

$$R(t)=\int p^*(-\mu)p(t-\mu)d\mu$$

Since the passive time-reversal processing is completely done in the Q-function, Equation (10) represents (as well for the pTR case) the ApTR output signal.

3.0.1. Static CIR and Conventional pTR

Usually the conventional pTR obtains the estimated CIR by an initial probe. In static channel case, considering the usual assumptions of pTR, the Equation (8) have impulse-like shape and it is valid the Relation (9).

3.0.2. Time-variant CIR and Conventional pTR

For the time-variant case with pTR using initial channel estimate, the function $Q(t,\mu)$ becomes

$$Q(t,\mu)=\sum_i \int \hat{h}_i^*(0,-\nu)\hat{h}_i(\mu-\nu)d\nu \qquad (12)$$

and due to the $\hat{h}_i(t,\mu)$ variability there will be mismatch between the CIR and the corresponding estimate. It results that, as time goes by, the impulse-like shape is lost and the ISI increases.

3.0.3. Time-variant CIR and ApTR

The ApTR, instead of estimating CIR only at the time of an initial probe, estimates the time-variant CIR along the data frame, making possible to maintain the impulse-like shape of the $Q(t,\mu)$ -function. In such conditions, the validity of Relation (9) is kept and ISI is mitigated.

Under the ideal case, it is assumed that the estimated (simulated) time-variant CIR matches perfectly with the observed CIR. However, more realistic approach requires to estimate the time-variant channel using a suitable method. This question is discussed in subsection 3.1.

Fig. 4 shows the ApTR block diagram. The signal $s(t)$ containing the information sequence is transmitted through the time-variant channels, the received signals are ApTR filtered and the sum of the processed channels generates the output signal $z(t)$, over which the coherent demodulation yields the estimated message.

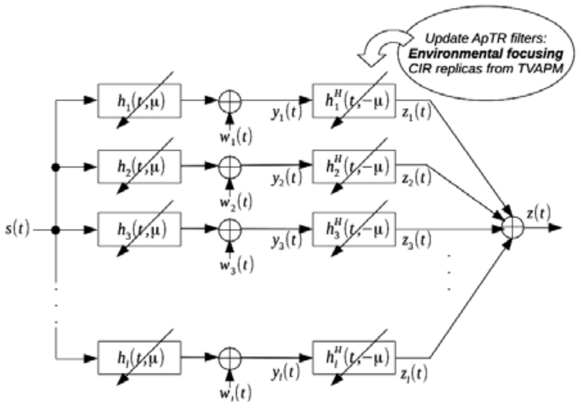

Fig. 4 Time-variant ApTR block diagram.

3.1. Environmental Focusing

A well-known technique that employs acoustic propagation modeling is the matched field processing (MFP). It was firstly proposed by [7] and consists in cross-correlate replicas of acoustic pressure field from a propagation model with the observed field in a receiver array, aiming to estimate a specific set of physical parameters. Generally the searched parameters aim to solve three classes of problems: passive source localization, matched field inversion for geoacoustic parameters and ocean acoustic tomography (estimation of water column sound speed profile or the closely related ocean temperature field). Important benchmarking in MFP is found in [8-11].

The environmental focusing method for ApTR equalization is inspired in the focalization technique [12] often used in MFP for source localization. Focalization, which simultaneously focuses and localizes, eliminates the stringent requirement of accurate knowledge about the ocean-acoustic environment by including the environment in the parameter search space. The main idea of environmental focusing in DAUC is to create a data bank of carefully chosen CIR (search space) that are then used in correlation processing to reach the synthetic CIR with best acoustic fitness to the observed CIR and use it to update ApTR filters. Environmental-based modeling yields useful information about the channel that can be used to design filters that incorporate suitable range of distortion over the initial CIR probe. Therefore, the physics involved in acoustic propagation would contribute to build the search space, where a less rigid requirement for probe channel matching is achieved, causing improvement in SNR output.

Channel tracking using a data bank strongly depends on how well channel variability effects are inserted in the search space. For instance, it is known that even small changes in source or receiver positions cause corresponding changes in acoustic field that severely affect the demodulation of coherent DAUC signals, then the search space must contain replicas that contain such geometrical variability in a fine scale. Previous coarse knowledge about the local environmental/geometric parameters where the system is employed contributes to choose a suitable filter design, since acoustic propagation paths are quite dependent of the channel type, *e.g.*, sound speed profile, water depth, boundaries conditions and source-receiver-surface positions. The influence of acoustic propagation physical parameters in high frequency regime must be investigated because its understanding strongly contributes to build a suitable search space. This method is being investigated and as part of this research the present work shows short results about the influence of the source range parameter for a waveguide with oscillating free surface. In future, with inclusion of information about effects of others significant parameters, reasonable results in realistic scenarios are expect to be found. However, it is still required detailed analysis of tests with sea trial data.

3.2. Simulated CIR Test

The following test shows the results from DAUC simulation using TVAPM to model time-variant CIR. Assumption of known simulated channels for feeding the corresponding ApTR filters is done, so that the ideal case is achieved (i.e., best possible match between estimated CIR and observed CIR is used). Even though no update through focusing method is done yet, the simulation results will show the Doppler compensation and equalization being performed by the ApTR.

Fig. 5 shows the scenario used in this test.

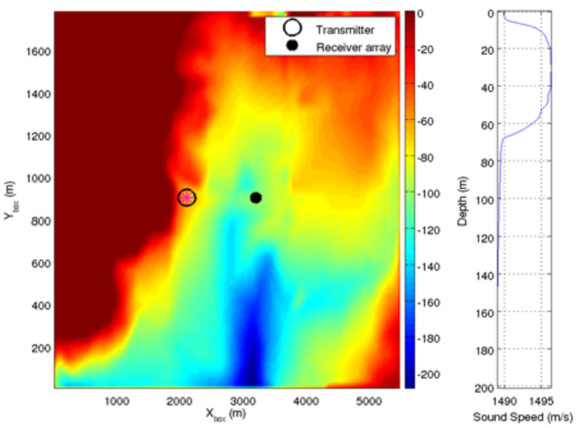

Fig. 5. Scenario used in the preliminary test with TVAPM integrated to the DAUC simulator.

The moving source (a big circle-star marks the initial position) and static array (small circle) are positioned inside a tridimensional environmental-based box with range-dependent bottom bathymetry and time-variant upper free surface. The upper surface is computed in TVAPM with a wind-driven model using Jonswap spectrum [4], where wind was set to direction 090 degrees and intensity $10\,m/s$. The runs were done with the sound source active sensor developing velocities $0\,m/s$, $0.4\,m/s$ and $1\,m/s$. BPSK signals were transmitted with carrier frequency $10\,kHz$ at bit rate $1000\,bit/s$. The data frame contained 1024 symbols.

4. Discussion and Conclusion

Results shows ApTR filters achieving compensation for Doppler Effect and equalization (this last limited by least SNR design inherent to pTR processors). Fig. 6 shows the received constellations after be processed by conventional pTR (first column), received constellations after be processed by ApTR (second column) and again the ApTR result but with rotation lock (third column). Each plotting line from top to bottom corresponds to the runs with source velocities $0\,m/s$, $0.4\,m/s$ and $1\,m/s$, respectively.

In the case of pTR received signal constellation, symbol clouds expand themselves rotating in ring-like form. This strong distortion is caused by Doppler Effect generated due to source motion along the transmission. In the case of ApTR processed received constellation the dense clouds clearly indicate the Doppler compensation and equalization. One can note in column 2 constellation rotation still occurring after ApTR processing, but for the symbol clouds as a whole. It is an effect from the cross-correlation between Doppler distorted CIR. The explanation comes from substituting time-variant CIR given in Equation (4) and Equation (6) into the Q-function of the Equation (8) and performing the integral. One can note that the result contain a reminiscent phase factor (not showed), which causes the constellation clouds rotation. This was solved performing tracking of the constellation rotation angle using the short m-sequence probe that was previously inserted in the frames for time synchronization. The estimated constellation angle is then used to correct the rotation, yielding the results showed in column 3.

Concluding, the ApTR processor was proposed to perform equalization jointly to Doppler compensation, aiming to mitigate ISI and improve performance in coherent underwater acoustic communications links whose sensors are in SIMO configuration. Model based adaptive procedure is an attractive approach that requires a method to update the equalizer filters with time-variant CIR estimates. In this sense, the environmental focusing method was suggested for investigation, inspired in the low-frequency matched field technique and exploring the understanding about the influence of acoustic propagation physical parameters through channel modeling.

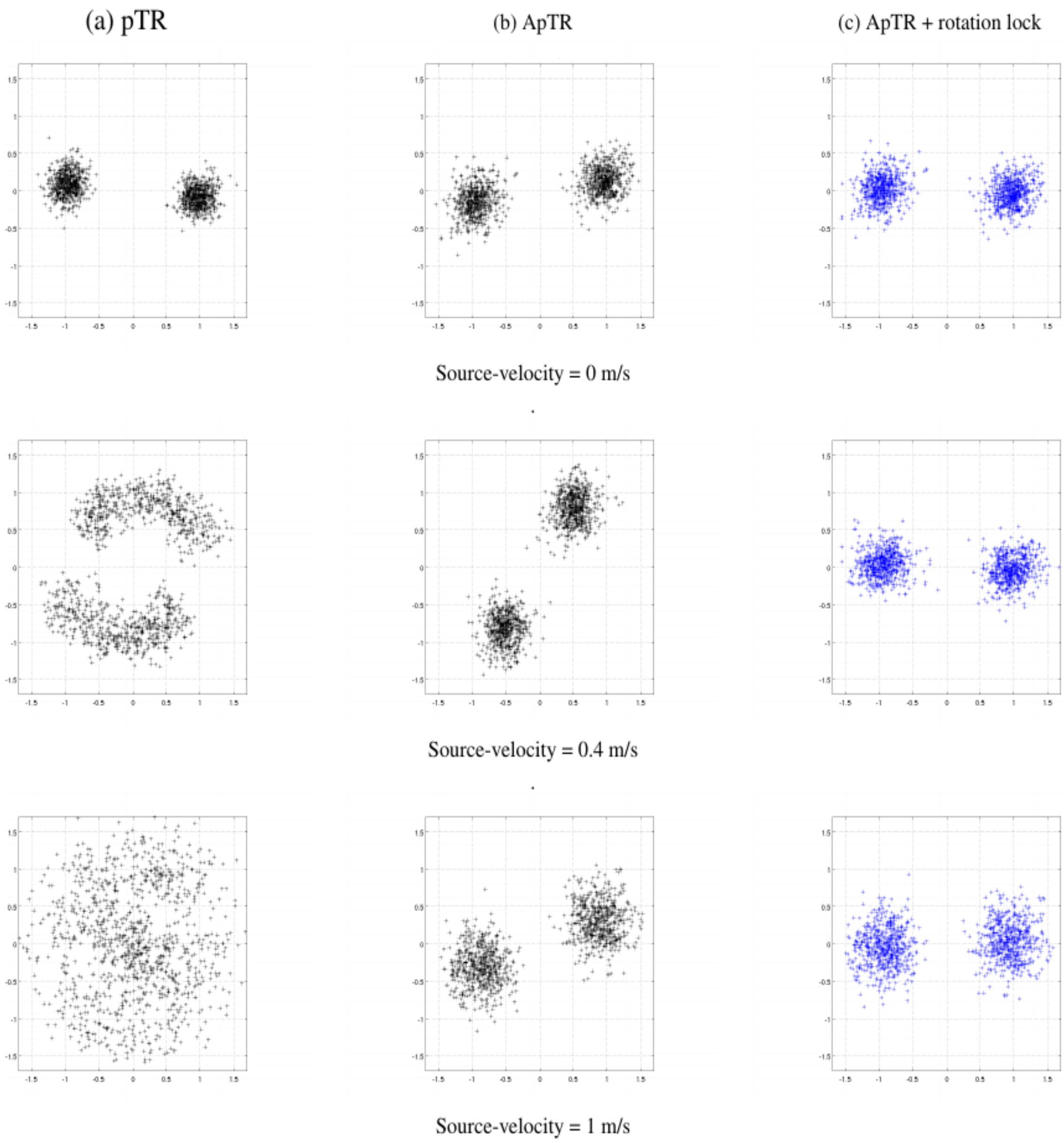

Fig. 6. Constellations for moving source and wind-driven modelled surface. Column (a): pTR; Column (b): ApTR without constellation angle correction. Column (c): ApTR with constellation angle correction.

Using suitable replicas set computed from information of realistic environment modeling, the estimates of time-variant CIR feed the corresponding ApTR filters. Results are expected to achieve in the future environmental-based equalization of underwater communications signaling. The corresponding DAUC performance improvement could benefit P2P communications or the link between two UAN nodes.

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