# Time-variant Adaptive Passive Time Reversal Equaliser and a Perspective for Environmental Focusing Method

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Abstract: High digital data throughput in Underwater Acoustic Communications (UAComm) is a challenging subject, specially in shallow water where the channel is a wave-guide causing multipath propagation and where Doppler effect usually occurs due to relative source-receiver motion jointly to ocean dynamics. The source and receiver sensors can be used for telemetry in point-to-point underwater communications or as nodes of an underwater acoustic network within the scope of oceanic research observatory or offshore activities. However, channel tracking is required for reliable digital underwater communications between the sensors, which is a hard task due to the complicated propagation of acoustic waves in the ocean. Equalisation is often required to perform a compensation method aiming to overcome the inter-symbol interference (ISI) caused by multipath propagation. The motivation of this work is to propose a compensation method deploying the adaptive passive time-reversal (ApTR) equaliser, aiming to perform ISI mitigation jointly to Doppler compensation in time-variant channels. The benefit given by ApTR processing would be the performance improvement in underwater communications between an active sensor and a vertical line array of receiver sensors, relying in well-succeed time-variant channel impulse response estimation. Furthermore, this position paper discusses the perspective of use an environmental focusing method for channel estimation within the ApTR equaliser, based on the idea that a set of oceanic acoustic physical parameters – which are generally estimated in low-frequency matched field processing problems like geoacoustic assessment, ocean tomography and source localization – could be conveniently used for channel compensation in high frequency underwater communications using a carefully chosen search space of replicas. The results are two fold: in one hand the equalisation shall improve the UAComm system, and in the other hand, the best match of channel parameters consequently yields a refined local environmental assessment.

## 1 INTRODUCTION

The use of high data rate signalling for UAComm 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 reasearch observatories (Vilaipornsawai et al., 2014). The complicated acoustic propagation in ocean waveguide channels makes hard the task of establish high rate data troughput between two node sensors, maily because of multipath propagation and Doppler distortion, often requiring equalisation. The equaliser can use channel estimates in the signal processing to reach compensation for time-frequency distortion imposed by the variable channel, aiming to mitigate ISI and improve communications performance.

Any channel estimation technique must deal with frequency selective attenuation, time dispersion from multipath propagation and frequency dispersion due to Doppler effect. Further, the use of coherent signalling is desirable for high data throughput because of its improved bandwidth efficiency, a desired characteristic in the UAComm system design considering the bandwidth limitation imposed by the ocean channel. Additionally, one can note that conventional equalisers 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 proposes to deploy the adaptive passive time reversal (here after named ApTR) for achieve channel equalisation in P2P wireless underwater communication, using single-input-multiple-output (SIMO) structure as required by passive time reversal technique.

Passive time-reversal (pTR) – and its frequency domain version named passive phase conjugation (PPC) (Gomes et al., 2008) – 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 (Parvulescu, 1961). In its passive version the processing can be performed synthetically, exploring spatial and temporal diversity of the acoustic channel by software setup. Nonetheless, the conventional pTR equaliser 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 equaliser is designed to overcome this problem. Further, it is proposed for investigation the environmental focusing method aiming to estimate the time-variant channel during the data transmission, performing correlation between the observed channel and carefully chosen replicas in a data bank which is pre-computed for a restrict area using time-variant high-frequency acoustic propagation model.

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

# 2 THEORETICAL BACKGROUND

## 2.1 CIR Modelling for UAComm

Acoustic propagation in shallow water performs multiple paths, causing ISI that contributes to errors in demodulation. Additionally, ocean dynamics and relative motion between source and receiver cause time-frequency dispersion, requiring compensation for Doppler distortion. Under the time-invariant channel assumption, the ray tracing acoustic propagation model is suitable for UAComm, because it allows to range-dependency, have fast computation and use the infinite-frequency approximation from the ray theory in the solution of the Helmholtz equation. However, there are naturally temporal variations in the ocean channel that make the time-variant approach necessary for more realistic CIR modelling.

The work of (Rodriguez and Silva, 2012) 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 (Porter, 2011). 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 the 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 UAComm model containing the ApTR equaliser.

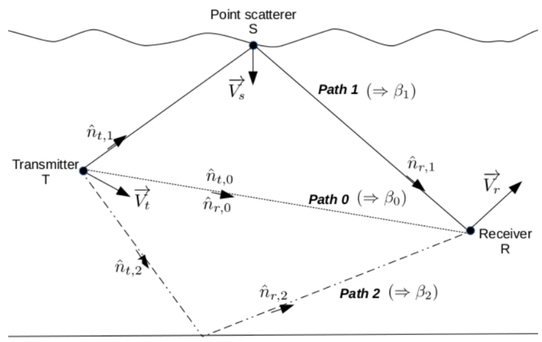


Figure 1: Bi-static scattering geometry (full line and dash-dot line) and mono-static geometry (fine dot line).

## 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 ($\beta_p$). This factor performs compression/expansion in time and frequency shift (with compression/expansion in the corresponding spectrum) in the signal.

Figure 1 shows the bi-static scattering geometry scenario (Ziomek, 1995), where the signal transmitted by a moving source (T) is reflected in a moving point scatterer (S) and received in a moving receptor (R). 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.

The time compression factor $\beta_p$ for a single path in the bi-static scattering geometry (paths 1 and 2) is

(Ziomek, 1995; Gomes et al., 2008)

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

where $c$ is the sound speed in the medium, the symbol $\bullet$ denotes dot product and $\vec{V}_x \bullet \hat{n}_{x,p}$ 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 compute the 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 ($\beta_0$ for the direct path, $\beta_1$ for the free surface reflected path and $\beta_2$ 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 bandpass signal $\tilde{s}(t)$ with carrier angular frequency $\omega_c$ 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:

$$\tilde{s}(t) = Re\{s(t)e^{j\omega_c t}\}; \quad s(t) = a(n)p(t-nT_s) \qquad (2)$$

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

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

Using baseband equivalent notation, the received noiseless signal for a single path can be represented as the convolution of the path distorted signal $s_{D_p}(t)$ 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 (Gomes et al., 2008):

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

with

$$y_p(t) = \int s(t-\mu)h_p(t,\mu)d\mu \qquad (5)$$

In the $i$-$th$ hydrophone, the time-variant CIR is:

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

# 3 APTR EQUALISER

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 analysis of the performance of passive time-reversal based equalisers.

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 $\mu$ 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

$$Q(t,\mu) = \sum_i \int \hat{h}_i^*(t,-\nu)h_i(t,\mu-\nu)d\nu \qquad (7)$$

For the time-invariant case, since the CIR do 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,

$$Q(t=0,\mu) \simeq \delta(t=0,\mu) \qquad (8)$$

The pTR output signal is given by

$$z(t) = \sum_{i=1}^{L} z_i(t) = \int Q(\mu)I(t-\mu)d\mu \qquad (9)$$

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

$$I(t) = \int a(\mu)R(t-\mu)d\mu \qquad (10)$$

and

$$R(t) = \int p^*(-\mu)p(t-\mu)d\mu \qquad (11)$$

Since the ApTR adaptive processing is completely done in the Q-function, equation (9) represents (as well for the pTR case) the ApTR output signal.

### 3.0.1 Static CIR and Conventional pTR

Usually the conventional pTR equaliser obtains the estimated CIR by an initial probe. In static channel case, considering the usual assumptions of pTR, the equation (7) have impulse-like shape and it is valid the relation (8).

### 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

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

and due to the $h_i(t,\mu)$ 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.

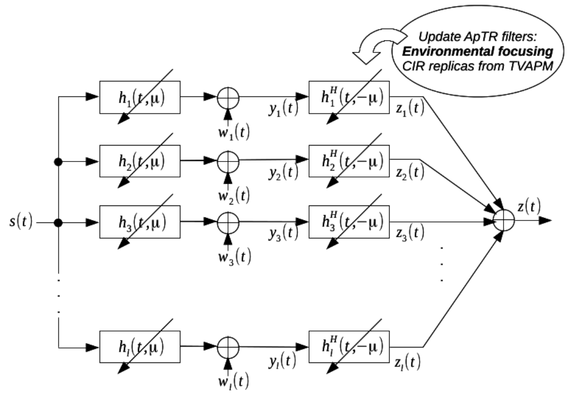


Figure 2: Time-variant ApTR equalizer block diagram.

### 3.0.3 Time-variant CIR and ApTR

The ApTR, instead of estimating the CIR only at the time of a 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 (8) 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.

Figure 2 shows the ApTR equaliser block diagram. The signal $s(t)$ containing the information sequence is transmitted through the time-variant channel and the received signals are processed in the ApTR equalizer, such that the sum of the processed channels generates the ApTR output signal $z(t)$, over which the coherent demodulation yields the estimated message.

## 3.1 Environmental Focusing

A well-known technique that employs acoustic propagation modelling is matched field processing (MFP). It performs correlations between replicas of acoustic pressure field from a propagation model with the observed field in a receiver array, aiming to estimate an specific set of physical parameters. It was firstly proposed by (Hinich, 1973). Generally the searched parameters aims to solve three classes of problems: passive source localization, matched field inversion for geoacoustic parameters and ocean acoustic tomography to perform estimation of water column sound speed profile (or the closely related ocean temperature field). Important benchmarking in MFP are found in (Bucker, 1976; Tolstoy et al., 1991; Jesus, 1993; Baggeroer et al., 1993).

The environmental focusing method for ApTR equalisation proposed in this work is inspired in the focalization technique (Collins and Kuperman, 1991), used in MFP for source localization. Focalization, which simultaneously focuses and localizes, eliminates the stringent requirement of accurate knowledge of the ocean-acoustic environment by including the environment in the parameter search space. The idea of environmental focusing in UAComm is to create a data bank of carefully chosen channel replicas (search space) that reaches the CIR with best acoustic fitness, which should update the ApTR filters. Generally, it is quite difficult to accurately model the ocean channel. Therefore, to make it a less rigid requirement, it is used this focusing method aiming to reach the replica that best matches the observed channel, instead of use an outdated initial probe. In fact, this method does not use the initial probe to be performed during the signal transmission. The method is based in careful pre-computation (and at large number) of time-variant channel replicas for a specific restrict area, acting as a probe-independent process (just as it happens in MFP). The success in channel tracking using the replicas data bank strongly depends of how well the channel variability was inserted in the search space. For example, it is known that even small changes in the source/receiver positions cause corresponding changes in acoustic field that severely affect the demodulation of coherent UAComm signals, then the pre-computed search space must contain replicas that track such geometrical variability in a fine scale. It can be expected considerable high computational cost in this procedure and for that reason it is need at least a coarse knowledge of the environmental/geometric physical parameters of the local where the system is employed. The influence of acoustic propagation physical parameters must be investigated, because its understanding strongly contributes to build a suitable search space of replicas. It should not be deliberately performed a brute force method for choosing the *a priori* sets of parameters aiming to avoid the corresponding huge computational cost. The area where the UAComm system will be employed must be analysed, being another basis for reasonable selection of *a priori* candidates.

This environmental focusing method proposed for ApTR equalisation is intended to be investigated and extensively tested by the author in future work. It is expected that the well-succeed experience of low-frequency MFP technique could be advantageously used in model-based adaptive passive time-reversal equalisation for high-frequency (*i.e.*, 3–50 kHz band) underwater communications. An evidence that the cases are similar also resides in the fact that in both

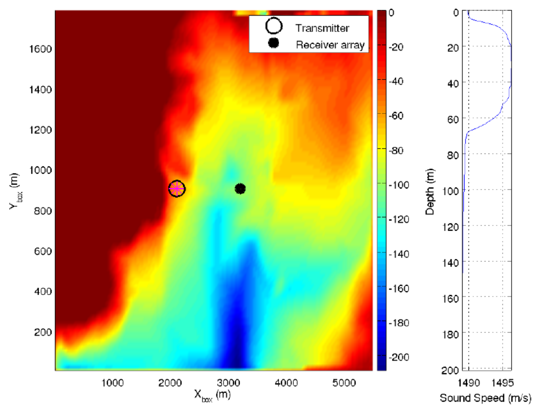


Figure 3: Scenario used in the preliminary test with TVAPM integrated to the UAComm model.

problems the same wave equation with boundary conditions must be solved. The results would yield the set of acoustic physical parameters that comes from the best fit between replicas and the observed CIR, being the process performed for each channel corresponding to each receiver sensor of the vertical line array. Such replicas with best fitness should be used to feed the respective ApTR filters and the sum over the channels yields the equalised output signal.

## 3.2 Simulated CIR Test

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

Figure 3 shows the scenario considered in this test. The moving source (the star with big circle marks the initial position) and the static array (small circle) are positioned inside a 3-D environmental-based box with range-dependent bottom bathymetry. The upper surface is computed inside the TVAPM with wind-driven model using spectrum JONSWAP (Rodriguez and Silva, 2012), direction 090 degrees, intensity 10 m/s . The runs were done with the sound source active sensor developing the velocities of 0 m/s, 0.4 m/s and 1 m/s. BPSK signals were transmitted with a carrier frequency 10 kHz. The data frame contained 1024 symbols with bit rate 1000 bit/s. The results show that the ApTR filters achieved the compensation for Doppler effect and the equalisation. Figure 4 shows the received constellations after be processed by conventional pTR (plottings in the first column), after be processed by ApTR (plottings in the second column) and after be processed by ApTR with rotation lock (plottings in the third column). Each line of plottings, from top to bottom, corresponds to the runs with source velocities 0, 0.4 and 1 m/s, respectively.

In the case of received signal constellation for the pTR equaliser, the clouds expand themselves rotating, in ring-like form. This strong distortion is caused by the Doppler effect from the source motion along the transmission. In the case of the received constellation after the ApTR equaliser the dense clouds clearly indicate the Doppler compensation and equalisation. Column 2 still shows that constellation rotation still occurs (for the clouds as a whole) after the ApTR processing. It is an effect from the cross-correlation between Doppler distorted CIR. Substituting the time-variant CIR of equation (4) and (6) into the Q-function of the equation (7) and performing the integral, one can note that the result contain a remain phase factor (not showed), which causes the constellation clouds rotation. This was solved performing the tracking of the constellation rotation angle using the short m-sequence probe which 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.

# 4 CONCLUSION

The ApTR processor was proposed to perform equalisation jointly to Doppler compensation, aiming to mitigate ISI and improve performance in coherent underwater acoustic communications links whose sensors are in SIMO configuration. The adaptive procedure requires a method to update the equaliser filters with the time-variant CIR estimates. In this sense, the environmental focusing method was suggested for investigation, inspired in the low-frequency matched field technique, exploring the understanding about the influence of acoustic propagation physical parameters for channel modelling, since they rule the physics of the acoustic propagation in the ocean channel.

Using suitable replicas of realistic environment, reached from well-tuned modelling that considers some *a priori* coarse information relative to the specific area where the system is used, the estimates of time-variant CIR feed the ApTR filters. As results are expected to achieve environmental-based equalisation of communications signalling and refined local environmental assessment. The corresponding UAComm performance improvement could benefit P2P communications or the link between two UAN nodes.

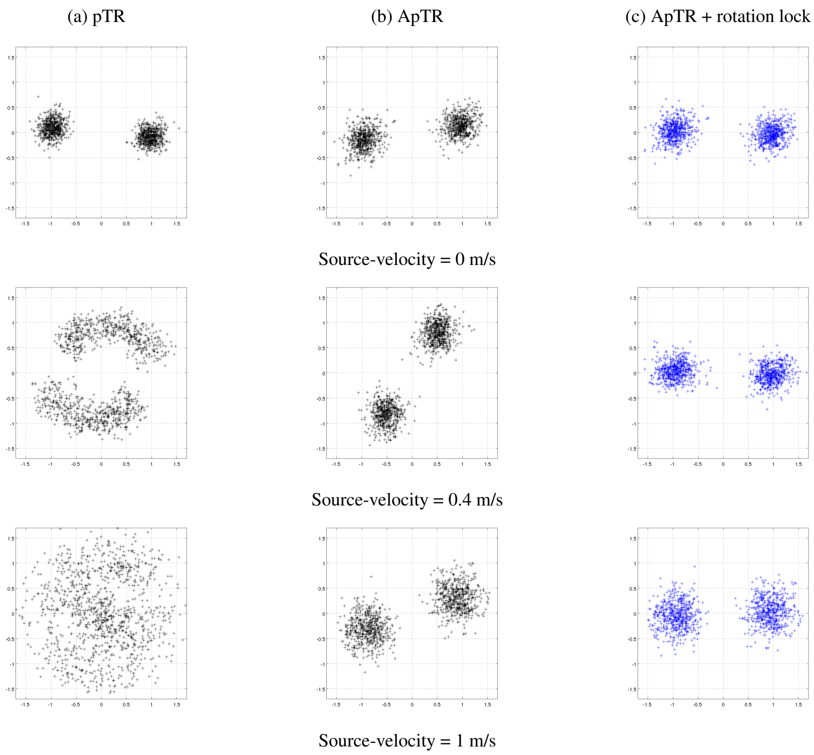


Figure 4: Constellations for moving source and wind-driven modelled surface. Column (a): pTR; Column (b): ApTR without constellation angle correction. Column (c): ApTR with constellation angle correction.

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