Proceedings of the 10th Mediterranean Conference
on Control and Automation - MED2002
Lisbon, Portugal, July 9-12, 2002.

BLIND CHANNEL ESTIMATION WITH DATA FROM THE INTIMATE '96 SEA TRIAL

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Keywords: channel estimation, multipath, time-frequency, LFM.

Abstract

Blind multipath channel estimation is studied by time-frequency (TF) analysis. For a linear frequency modulated source, the technique is based on its instantaneous frequency estimation, followed by an approximate formulation of matched-filtering. Tests concern at-sea recorded data during the INTIMATE '96 experiment.

1 Introduction

Channel estimation is nowadays a well-known problem, examples of which can be found, *e.g.*, in sonar[1], radar, communications[2, 3] or geophysics[4, 5, 6, 7, 8, 9]. Depending on the structure of the channel to be estimated, and on the knowledge of the emitted signal, a variety of methods has emerged, ranging from minimum entropy[5, 10] to least squares-based deconvolution [1, 3]. If the emitted signal is known, a correlator receiver can be used for channel estimation, what constitutes one application of the classical matched-filter. For the purpose of channel estimation, it is related to the principle that, if the source signal bandwidth is rather large, then its auto-correlation is well approximated by a Dirac impulse function. If the channel behaves as a multiple time delay-attenuation channel, here designated as multipath channel (MC), then the peaks in the correlator output give the estimates of the amplitudes and time-delays. It can be shown that, if the true arrival times differ by more than the duration of the signal autocorrelation function, and in the presence of white Gaussian noise, the correlator is equivalent to the maximum likelihood estimator[4]. Usually, the core of MC estimation remains at estimating the time-delays, since their estimates do not depend on the amplitudes, whereas they depend on the source signal. These estimates are then used for amplitude estimation. Frequency-domain approaches have also been proposed by Kirsteins[9] and Vaccaro *et al.*[1]. Since a delay in the time domain is equivalent to multiplication by an exponential in the frequency domain, the corresponding frequency domain problem consists of fitting weighted complex exponentials to the spectrum of the received signal. In the former approach, an iterative method is used, while the later is based on least-squares estimation.

In underwater acoustics, the environment is often modelled as an MC, and the estimate of the time--delays is required, for example, for tomographic applications, namely travel-time tomography[11]. The present paper does not intend to give estimates of the true amplitudes and time-delays of the channel. It presents a channel estimation method, valid for the case of an LFM source signal with unknown instantaneous frequency. The method is based on a modified time-frequency formulation of the matched-filter. It starts by the estimation of the LFM source signal instantaneous frequency, giving at the end the estimate of the channel impulse response. This is a sub-optimal method, relatively to the matched-filter, and has the advantage of not requiring the complete emitted signal knowledge.

This paper is structured as follows. Next section describes the data model. In Sec. 3, the TF distributions used in the work are briefly summarized. Section 4 explains the adopted TF formulation of the matched-filter, followed by the description of instantaneous frequency estimation, in Sec. 5. Application to real data from the INTIMATE '96 sea trial is described in Sec. 6, followed by the conclusions.

2 Data model and problem formulation

This section justifies the assumed data model, physically justifying the structure of the underlying MC that is to be estimated.

Prediction of the acoustic field due to a source in the ocean, is not a trivial problem, taking into account the wave equation that governs propagation[12]. However, a multitude of models and respective numerical implementations have emerged, to describe sound propagation in the sea. From these, ray tracing has been adopted in this work. It is based on the assumption that a set of rays depart from the source location, each of them carrying information about the emitted signal. The information on each ray consists of a delayed and weighted version of the emitted signal. Each ray travels along a path, determined by the sound speed profile and the medium boundaries, and, in shallow water, usually reflects back and forth between the sea surface and bottom. Given a set of wavelengths of interest, this model is as accurate as higher the ratio between each constant-polarity path length and each wavelength. A constant-polarity path length refers to the distance between the source location and the first reflection point (on the surface or bottom), to each distance between two reflection points, and to the distance between the last reflection point and the receiver. Ray tracing allows to regard the ocean as a filter, with a given impulse response (IR) $h_l(t)$, representing propagation from the source to a given l-index receiver($l = 1, \ ..., \ L$). This filter is considered to behave as an MC, whose IR $h_l(t)$ can be represented by:

$$h_l(t) \ = \ \sum_{m=1}^{M_l} a_{lm}\delta\left(t-\tau_{lm}\right), \qquad (1)$$

where $\{a_{lm}, \ \tau_{lm}; \ l = 1, \ ..., \ L; \ m = 1, \ ..., \ M\}$ are respectively the attenuations and time-delays, along the M_l acoustic paths[13]. The amplitudes and time-delays of $h_l(t)$ can be grouped into 2 vectors, respectively:

$$\mathbf{a}_l = [a_{l1}, a_{l2}, ..., a_{lM}]^T; \boldsymbol{\tau}_l = [\tau_{l1}, \tau_{l2}, ..., \tau_{lM}]^T. \quad (2)$$

The L channels are considered linear and time--invariant (LTI) during the period comprising emission and propagation. Following this model, the emitted signal $s(t)$ is convolved with L LTI single-channel systems $h_l(t)$, giving rise to L received signals $r_l(t)$, as follows:

$$r_l(t) = \sum_{m=1}^{M_l} a_{lm}s\left(t-\tau_{lm}\right) + \xi_l(t), \ l = 1, \ ..., \ L, \ (3)$$

where the noise $\xi_l(t)$ is modelled as a white zero mean random process.

In a complete formulation of the MC estimation problem, the 3 deterministic parameters M_l, $\mathbf{a}_l$ and $\boldsymbol{\tau}_l$ are to be estimated, since they are necessary and sufficient to describe the channel IRs $h_l(t)$. The discrimination ability of the channel estimator depends on the combined effect of the proximity between the IR's impulses and the inverse of the emitted signal band. In general, this discrimation is greater at later impulses, when talking about underwater acoustic channels. Also, as the receiving system has no knowledge about the transmission instant, the channel estimate is a function of relative time.

3 Time-frequency analysis

In the present work, the acoustic source emits a deterministic non-stationary signal. In this framework, advantage has been taken from the use of time-frequency distributions (TFDs), to analyze and process the received signal, in order to estimate the multipath channel's IR.

For the purpose of channel estimation, the classical Wigner-Ville (WV) distribution and the signal-dependent radially Gaussian kernel (RGK) distribution are relevant. The former satisfies the Moyal's formula, what entails naturally a time-frequency formulation of the matched-filter. The later, by the reduction of interference terms, in the TF plane, allows to estimate the instantaneous frequency of the emitted signal. For this distribution, the attenuation of the interference terms was achieved according to[14]

$$RGK_x(t,f) = \mathrm{IFT2}\left[\Phi_{RGK,x}(\nu,\tau) AF_x(\nu,\tau)\right], \quad (4)$$

where $\Phi_{RGK,x}(t,f)$, the kernel of the distribution, is adapted to the signal $x(t)$ to be analyzed, $AF_x(\nu,\tau)$ stands for the ambiguity function of $x(t)$, and IFT2$[\cdot]$ designates the bi-dimensional inverse Fourier transform operator. The kernel design is formulated as a constrained optimization problem.

4 Channel estimation

This section starts by referring to the estimation of an arbitrary channel, where the knowledge of the emitted signal is assumed. Then, a particular case is presented, namely for the context of the MC excited by an LFM signal. Both the classical MF and the presented TF channel estimates are compared, this last allowing for the inclusion of estimated information of the LFM, namely its instantaneous frequency.

4.1 Correlation-based channel estimation

The correlator receiver makes use of the principle behind the MF, a well-known filter that can be used in the identification of MCs, when the emitted and received signals are available[13]. Here, the terms *correlator receiver* and *matched-filter (MF)* will be used indifferently, with the same meaning, for simplicity.

The MF output is the correlation function

$$\Gamma_{r,s}(\tau) = \int r(t)s^*(t-\tau)\mathrm{d}t. \quad (5)$$

This leads to an MF-based channel estimator, given by the modulus of the right hand side of the above expression:

$$\hat{h}_{MF}(t) = \left|\int r(t)s^*(t-\tau)\mathrm{d}t\right|. \quad (6)$$

A natural question that arose in this work, considering the energetic interpretation of signal representation achieved by quadratic TFDs like the WV distribution, concerned the possibility of performing the MF in the TF domain. The answer to this question can be found, by considering Moyal's formula[15], which, for the case of the WV distribution, stands as:

$$\iint_{-\infty}^{\infty} WV_{x_1,x_2}(t,f) WV^*_{x_3,x_4}(t,f)\ \mathrm{d}t\ \mathrm{d}f = \left[\int_{-\infty}^{\infty} x_1(t)x_3^*(t)\mathrm{d}t\right]\left[\int_{-\infty}^{\infty} x_2(t)x_4^*(t)\mathrm{d}t\right]^*. \quad (7)$$

For the particular cases $x_1(t) = r(t)$, $x_2(t) = r(t)$, $x_3(t) = s(t-\tau)$ and $x_4(t) = s(t-\tau)$, (7) transforms into

$$\iint_{-\infty}^{\infty} WV_{r,r}(t,f) WV^*_{s(t-\tau),s(t-\tau)}(t,f)\ \mathrm{d}t\ \ \mathrm{df} = \left|\int_{-\infty}^{\infty} r(t)s^*(t-\tau)\mathrm{d}t\right|^2. \quad (8)$$

This expresses the equivalence between the MF and a 'correlation' in the TF domain, with respect to the variable t. In other words, if the receiver had knowledge about the source signal, then an MF could be done equivalently in the time domain, by (6), or in the TF domain, by the TF-based channel estimator given by the square root of the left hand side of (8).

4.2 Multipath LFM-driven channel estimation

The MF and TF channel estimators presented in the previous section assume particular structures, in the context of an MC and an emitted LFM signal, as seen below.

Considering the MF, for the channel structure (1), and for each particular true time-delay value $\tau = \tau_p$, $1 \le p \le M$, the correlation takes the value

$$\Gamma_{r,s}(\tau_p) = \quad a_p E_s + a_m \sum_{\substack{m=1\\m\neq p}}^{M} \Gamma_{s,s}(\tau_p - \tau_m), \qquad (9)$$

where E_s is the energy of $s(t)$. For values τ_p such that $\Gamma_{r,s}(\tau_p - \tau_m) = 0$, $\forall m \neq p$, the matched-filter output is proportional to the amplitude a_p of the channel, at time-lag τ_p. The condition $\Gamma_{r,s}(\tau_p - \tau_m) = 0$ is verified if the separation between the time-delays τ_p and τ_m is greater than the duration of the auto-correlation function $\Gamma_{s,s}(\tau)$. If this is the case for all the time-delay pairs, the main peaks of $\Gamma_{r,s}(\tau)$ are located in τ_m, $m = 1, \ldots, M$, and the matched-filter is an optimum estimator of the channel amplitudes a_m and time-delays τ_m[4]. If there are some pairs of time-delays separated by less than the duration of $\Gamma_{s,s}(\tau)$, it is difficult to resolve all the individual signals from the MF output, and the overlap often introduces errors into the amplitude and arrival time estimates. Let us now particularize the MF for the case of an LFM signal at emission. Let $s_{LFM}(t)$ be a complex LFM signal, zero outside the interval $[0, T]$, with modulation rate α and instantaneous frequency f_0, at $t = 0$:

$$s_{LFM}(t) = \mathrm{e}^{\mathrm{j}2\pi\left(\frac{\alpha}{2}t^2 + f_0 t\right)}. \qquad (10)$$

The envelope of the cross-correlation function, hence the expression (6), for this particular case, is given by

$$\left|\sum_{m=1}^{M} a_m \left(T - |\Delta_m|\right) \mathrm{e}^{\mathrm{j}\pi\Delta_m(2f_0+\alpha T)} \right.$$
$$\left. \times \mathrm{sinc}\pi\alpha\Delta_m \left(T - |\Delta_m|\right) \mathrm{rect}\left[\Delta_m/(2T)\right]\right|, (11)$$

with $\Delta_m = \tau - \tau_m$.

Each of the sinc functions in (11), for $\tau \geq \tau_m$, has the first zero given by

$$z_{MF} \quad = \quad \tau_m + \frac{T - \sqrt{T^2 - 4/\alpha}}{2}. \qquad (12)$$

The quantity $2(z_{MF} - \tau_m)$ gives an idea of the resolution of the MF, for MC identification, when driven by an LFM input.

Considering now the TF-based estimator, a sub-optimal estimator that does not require the complete knowledge of the source signal, may be obtained, in the particular case of the LFM signal, making, in (8), $s(t) = s_{LFM,\infty}(t)$. The WV distribution of $s_{LFM,\infty}(t)$ is given by

$$WV_{s_{LFM,\infty}} = \delta[f - f_i(t)], \qquad (13)$$

what implies a particularization of (8) to

$$\int_{-\infty}^{\infty} WV_{r,r}[t, f_i(t-\tau)]\ \mathrm{d}t$$
$$= \left|\int_{-\infty}^{\infty} r(t) s^*_{LFM,\infty}(t-\tau)\mathrm{d}t\right|^2. \quad (14)$$

For an emitted LFM signal, the right hand side of (14) can be interpreted as a modified version of the MF, where the input signal has been replaced by its infinite-duration version. This modified MF is equivalent to the integration of $WV_r(t, f)$, along a delayed version of the instantaneous frequency. In this case, the corresponding TF-based channel estimator is given by

$$\hat{h}_{TFI}(t) = \int_{-\infty}^{\infty} WV_{r,r}[t, f_i(t-\tau)]\ \mathrm{d}t. \qquad (15)$$

The explicit expression for the channel estimate is given by

$$T\left|\sum_{m=1}^{M} a_m \mathrm{e}^{\mathrm{j}\pi\Delta_m[2f_0+\alpha(T-\Delta_m)]}\mathrm{sinc}\pi\alpha T\Delta_m\right|, \quad (16)$$

which does not differ significantly from the homologous MF estimate (11). Each of the sinc functions in (16), for $\tau \geq \tau_m$, has the first zero given by

$$z_{TF} \quad = \quad \tau_m + \frac{1}{\alpha T}. \qquad (17)$$

Similarly to the MF context, in (12), the quantity $2(z_{TF} - \tau_m)$ gives an idea of the resolution of the TF channel estimate. As can be easily observed, both resolutions of the MF and TF estimates are approximately the same, for large values of $T >> 2/\sqrt{\alpha}$.

As the true instantaneous frequency $f_i(t)$ is not available at the receiver, it is natural to use an estimate $\hat{f}_i(t)$ in (15), leading finally to the TF blind channel estimator

$$\hat{h}_{TFB}(t) = \int_{-\infty}^{\infty} WV_{r,r}[t, \hat{f}_i(t-\tau)]\ \mathrm{d}t. \qquad (18)$$

5 Instantaneous frequency estimation

To estimate $f_i(t)$, one measure of the received signal's instantaneous frequency, using the conventional analytic signal definition[16], would not be a good procedure to adopt, since the multicomponent structure would imply an erroneous measure which, at each time t, would take into account all the present components.

In this work, the estimation of $f_i(t)$ has taken advantage of signal-dependent distributions, as stated below. Let us designate by $I(t,f)$ an ideal linear signal-dependent TFD, infinitely concentrated around the instantaneous frequency line, for finite or infinite duration signals with only a frequency modulation component. The ideal distribution of an emitted LFM sweep would be

$$I_s(t,f) = \delta\left[f - (f_0 + \alpha t)\right] \mathrm{rect}\left(\frac{t-T/2}{T}\right), \quad (19)$$

and, for the received signal,

$$I_r(t,f) = \sum_{m=1}^{M} a_m^2 \delta\left\{f - \left[f_0 + \alpha\left(t - \tau_m\right)\right]\right\} \times \mathrm{rect}\left(\frac{t - \tau_m - T/2}{T}\right). \quad (20)$$

It would be trivial to identify weighted versions of the source distribution $I_s(t,f)$ in $I_r(t,f)$, replicated so many times as the number of arrivals M. As the early arrivals would be represented in $I_r(t,f)$ by large amplitudes along the instantaneous frequency of the source signal, maximization of $I_r(t,f)$ with respect to t, within a given band of interest B, would “pick” the strongest arrival, giving an unbiased estimate of $f_i(t)$. Of course, within the available possible non-linear TFDs, analysis is constrained by the particular characteristics of the kernel, and by finite data lengths. Nonetheless, it seems reasonable to apply the maximization with respect to t, to a signal-dependent distribution of the received signal –in this work, the RGK distribution[1], $RGK_r(t,f)$–, what will give an accurate estimate of $f_i(t)$, if $RGK_r(t,f)$ is a reasonable approximation of $I_r(t,f)$, *i.e.*, if $RGK_r(t,f)$ attains good interference rejection, without causing a significant broadening of the signal components.

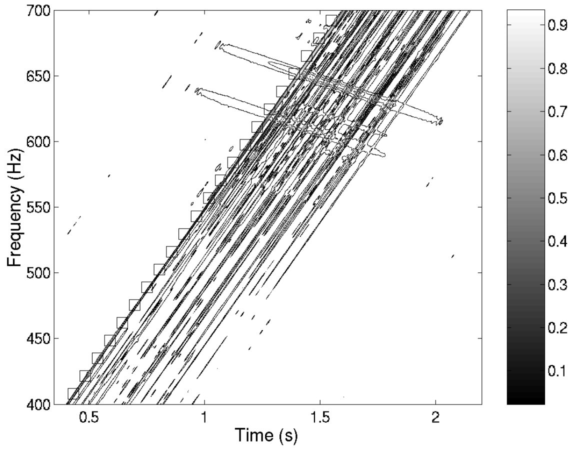


Figure 1: Instantaneous frequency estimate, obtained by maximization, with respect to time, of $RGK_r(t,f)$. The instantaneous frequency estimate $\hat{f}_i(t)$ is represented by squares.

Fig. 1 illustrates instantaneous frequency estimation, by maximization of $RGK_r(t,f)$, with respect to time. The distribution shown on the figure represents a realization of a simulated received signal on the hydrophone at 35 m, corresponding to a scenario very similar to the real data acquisition scenario of the INTIMATE '96 sea trial[17].

An analysis of the instantaneous frequency estimator characteristics is difficult to carry on. This is mainly due to the variability of the kernel of a signal-dependent distribution, with the signal.

[1]The instantaneous frequency estimate thus obtained is not rigorously a function of t, due to its definition as the ‘inverse’ of a non-injective function.

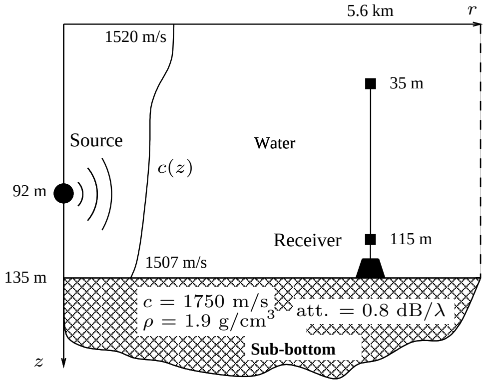


Figure 2: INTIMATE '96 real data environment scenario considered in this chapter.

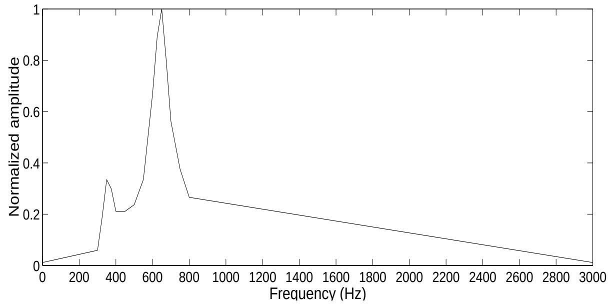


Figure 3: Electro-acoustic transducer amplitude spectrum, in the INTIMATE '96 sea trial.

6 Experimental results

6.1 Sea trial description

The INTIMATE '96 sea trial was primarily designed as an acoustic tomography experiment to observe internal tides. Details of the experimental setup have appeared elsewhere[17]. The sea trial was conducted in the continental platform near the town of Nazaré, off the west coast of Portugal, during June 1996, and consisted of several phases during which the acoustic source was either stationary or being towed along predetermined paths. The results in this section concern data acquired in phase 1, on the hydrophones at 35 and 115 m, during approximately 1 h, from which 13 groups of received signal realizations (snapshots) were taken. Each group consists of $N = 10$ snapshots, each with a duration of 8 s. During this phase, the scenario is as shown in Fig. 2, modelled as comprising a water layer superimposed to a half space with constant density.

The source signal used in the INTIMATE '96 sea trial was a 300–800 Hz LFM sweep with 2-s duration, repeated every 8 s, and emitted in practice by an electro-acoustic transducer of type Janus--Helmholtz. The transducer presented a main resonance at 650 Hz and a secondary resonance at 350 Hz, as measured on the device, and seen in Fig. 3. A model $s(t)$ of the emitted signal is represented in Fig. 4. This signal is well approximated by the product of the 300–800 Hz LFM sweep by an instantaneous amplitude approximately equal to the transducer amplitude spectrum in Fig. 3, in the LFM's instantaneous frequency range. According to this approximation,

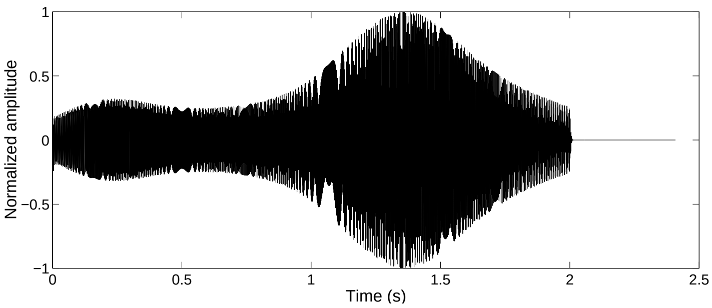


Figure 4: Model of the true source signal –real part of $s(t)$– in the INTIMATE '96 sea trial.

the instantaneous frequency $f_i(t)$ of $s(t)$ is essentially the same as the instantaneous frequency of the LFM sweep, as seen in the representation of the instantaneous frequency of $s(t)$, determined using the analytical signal definition[16] –Fig. 5. Note that the difference between $s(t)$ and a 'pure' LFM signal with constant instantaneous amplitude imply slight modifications in derivations (10–17). This represents an additional optimality reduction in the TF estimator, relatively to that mentioned in Sec. 4.2, due to the fact that the instantaneous amplitude of $s(t)$ is not taken into account, in the TF-based channel estimation. This sub-optimality can imply larger sidelobes in the channel estimate, as compared to the MF estimate.

The SNR has been estimated to be approximately 10 dB within the frequency band of interest[13].

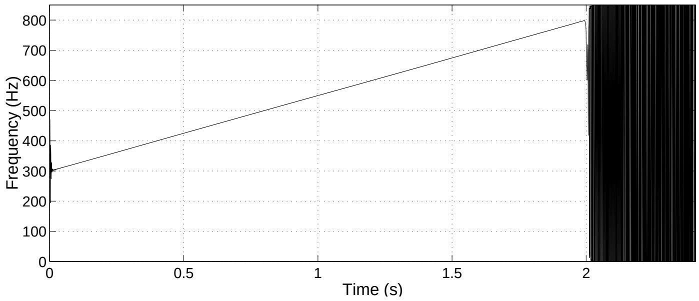


Figure 5: Instantaneous frequency of $s(t)$, in the INTIMATE '96 sea trial.

6.2 Channel estimation results

Proceeding as suggested in [13], for amplitude and time-delay estimation, in the presence of a set of snapshots at reception, the effectively used channel estimators were given by

$$\hat{h}_{CE}(t) = \frac{1}{N} \sum_{n=1}^{N} \hat{h}_{CE,n}(t), \tag{21}$$

where CE is to be replaced by MF, TFI and TFB, in order to obtain averaged versions of the estimators (6), (15) and (18), respectively. Similarly, the effectively considered instantaneous frequency estimate is given by

$$\hat{f}_i(t) = \frac{1}{N} \sum_{n=1}^{N} \hat{f}_{i,n}(t), \tag{22}$$

where $\hat{f}_{i,n}(t)$ was obtained by maximization of the RGK distribution of each snapshot, as described in Sec. 5.

Let us start the analysis of the results, considering the hydrophone at 115 m. A reliable model of the true impulse response, obtained by the normal mode propagation model C-SNAP[18], is shown in Fig. 6 (a). As can be seen, it consists essentially of a set of leading peaks with large amplitude, followed by a peaky pattern with small amplitude. To obtain the estimates, the first group of 10 snapshots was considered, for averaging. The channel estimate obtained by the average of the MF (6), using the signal model in Fig. 4, is depicted in Fig. 6 (b). As expected, the resolution of the MF constrains the discrimination of the early peaks of the channel IR. To have an idea of

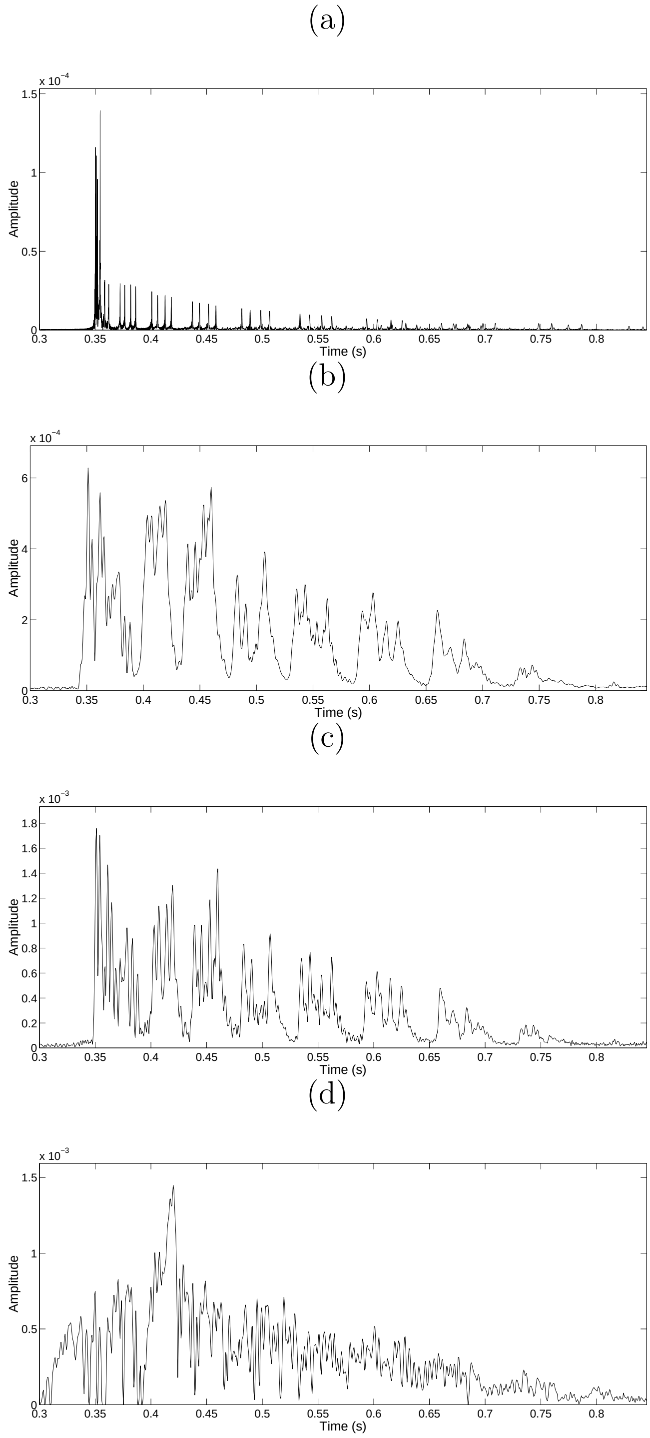
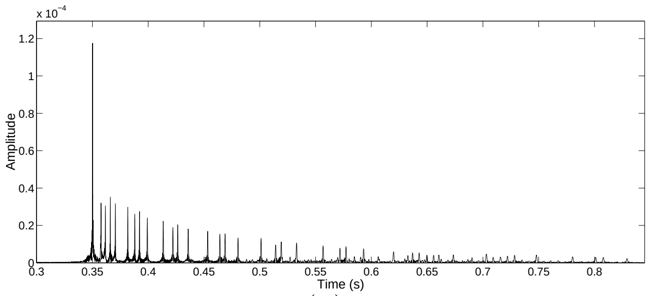


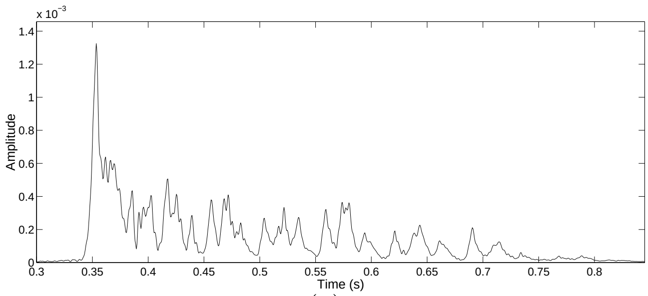
Figure 6: Channel model and estimates, corresponding to the hydrophone at 115m, for the first 10 snapshots. (a): model; (b): MF; (c): TF, with the knowledge of $f_i(t)$; (d): TF, without the knowledge of $f_i(t)$.

the maximum attainable quality of the TF channel estimator, an estimate was obtained by the average of the coherent integration (15), for the

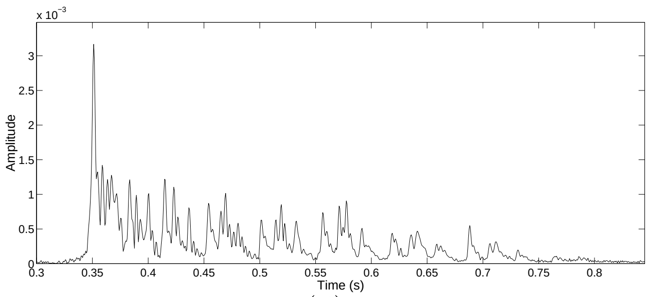
(a)



(b)



(c)



(d)

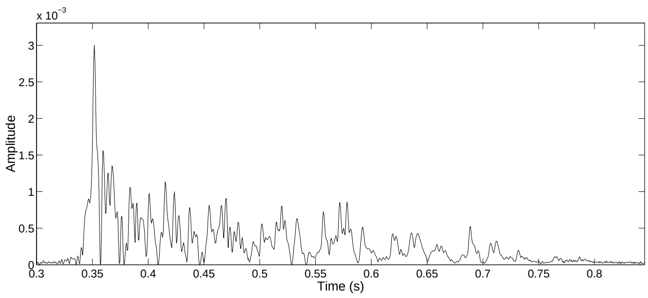


Figure 7: Channel model and estimates, corresponding to the hydrophone at 35m, for the first 10 snapshots. (a): model; (b): MF; (c): TF, with the knowledge of $f_i(t)$; (d): TF, without the knowledge of $f_i(t)$.

10 snapshots, and is shown in Fig. 6 (c). This estimate exhibits a slightly higher resolution than the MF. That can be explained by a larger duration of the auto-correlation function of the emitted signal, as compared to the coherent integration of the WV distribution of the emitted signal. This reflects the sub-optimality mentioned in the previous section, with the compromise of possibly estimating false impulses for the IR.

Assuming then that the knowledge of $f_i(t)$ is not available, the instantaneous frequency estimate was obtained as described in Sec. 5, and then inserted into the channel estimator (18), leading to the blind channel estimate depicted in Fig. 6 (d). It is clear that this is a poor-quality estimate, and one can conclude that this result is due to a poor estimate of $f_i(t)$.

Considering the hydrophone at 35 m, one can see its corresponding IR on Fig. 7 (a). The channel estimates obtained by the MF, and by the TF channel estimator, using $f_i(t)$, are shown in Fig. 7 (b) and (c), respectively. Finally, the blind channel estimate is shown in Fig. 7 (d), and, contrarily to the homologous 115-m hydrophone case of 6 (d), this channel estimate presents a good quality, as can be seen by comparation with the case where $f_i(t)$ is assumed to be known, in Fig. 7 (c).

The quality of the channel estimate is thus naturally dependent on the quality of the instantaneous frequency estimate. More importantly, by means of a careful observation of Figs. 6 (a) and 7 (a), it can be seen that there is a difference between the structure of the early impulses of the IRs corresponding to the hydrophones at 115 and 35 m, respectively. In the 115 m case, this set of close unresolved peaks corresponds to a spread in the TF plane, what, added to the low-pass filtering of the WV distribution, in the calculus of the RGK (4), will give a biased estimate $\hat{f}_i(t)$. This difficulty is not observed for the 35 m-IR, due to the clear separation between the earlier peak and the remaining peaks.

A sketch of the evolution with time, of the channel estimates coresponding to the hydrophone at 35 m, for the 1h-data, is drawn in Figs. 8 and 9, respectively, for the MF and the TF channel estimator, where no knowledge about the emitted

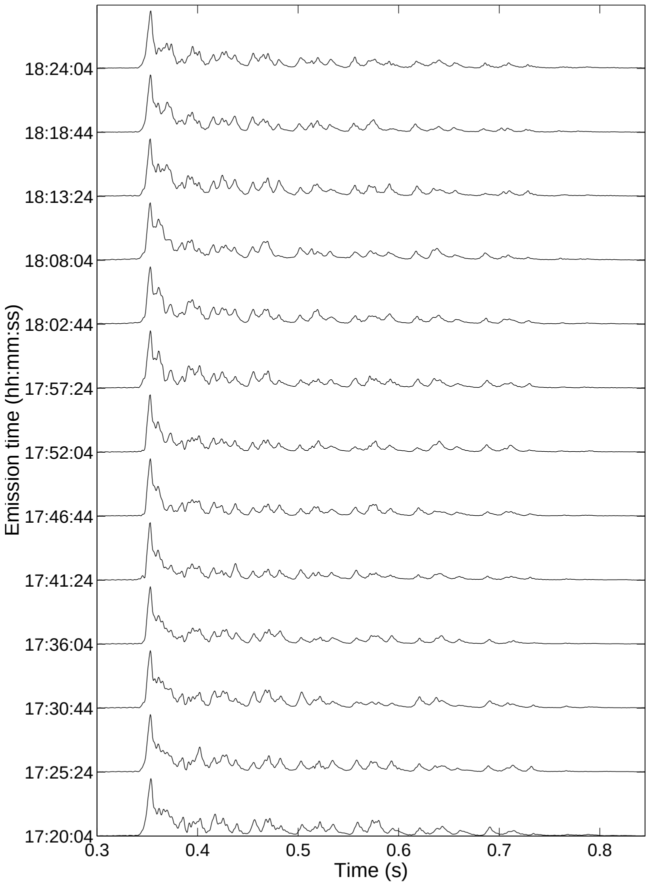


Figure 8: Channel estimates as a function of time, obtained with the MF, for the hydrophone at 35m. Emission times are middle point of the intervals considered for the 13 estimates.

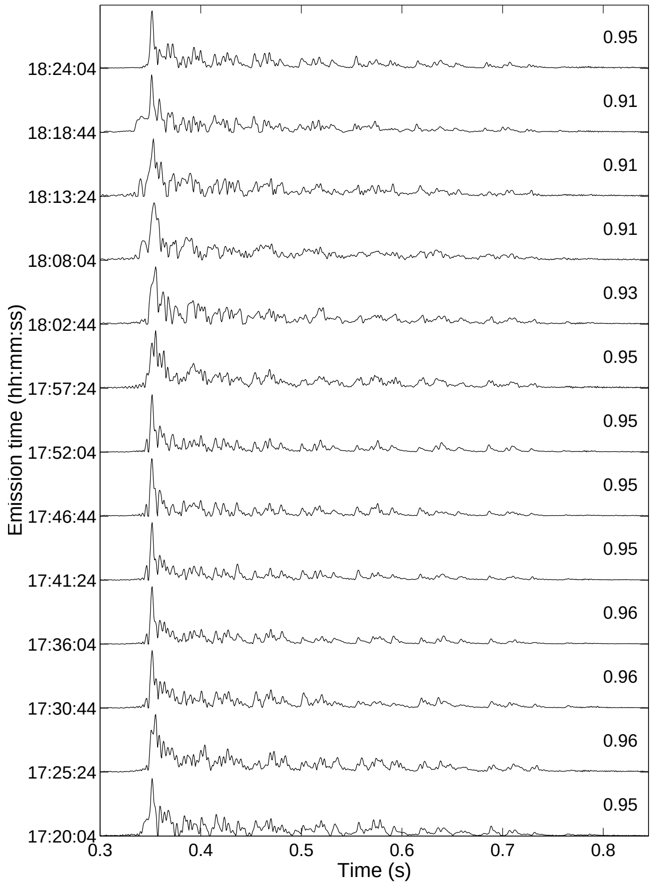


Figure 9: Channel estimates as a function of time, obtained by coherent integration of the WV distribution, for the hydrophone at 35m. Correlation coefficients relative to the MF are shown on the right hand side.

signal was used. The TF blind channel estimates are very similar to their MF counterparts, as can be confirmed by the correlation coefficients shown on the right hand side of Fig. 9.

Conclusions

A blind sub-optimal TF channel estimator has been proposed, for the case of an MC driven by a deterministic LFM signal. The method was tested on 1 h-duration real data from the INTIMATE '96 sea trial. Two main topics emerge as a conclusion. The first is that, if the emitted signal is known, then both the MF and the TF channel estimators are similar in structure and resolution. The TF estimator attains a slightly better resolution, at the cost of sub-optimality. As a second topic, if the emitted signal is unknown, in what case the MF is inapplicable, then a yet reliable TF estimator is obtained. Departing from the estimate of the unknown instantaneous frequency of the emitted signal, the quality of this estimator is consequently highly sensible to the quality of the instantaneous frequency estimate.

As seen above, the instantaneous frequency could not be accurately estimated in every hydrophone. This fact has to be taken into account, on a real application where the final goal is to estimate physical parameters of the propagation medium, with a preceeding channel estimation step. This implies a careful choice of the depth at which the hydrophone is to be placed. In the future, the TF channel estimator should be statistically characterized, and extended to a more general class of emitted signals.

Acknowledgements

Thanks are due to the INTIMATE '96 team for the real data acquisition, to FCT, for funding, under fellowship PRAXIS XXI (BM/19298/99), and to A. Quinquis and C. Gervaise for the valuable reception at ENSIETA.

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