MODEL-AIDED DATA ADAPTIVE SUPPRESSION OF REVERBERATION

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Abstract We propose a processing methodology which is based on the piece-wise modeling, adaptive estimation, and removal of reverberation components. An important feature of this approach is that backscatter models and information from pre-processing are used to guide the modeling of the reverberation components and the design of the algorithms to estimate them. The proposed processing methodology is developed for the case when strong, highly temporally localized reverberation components plus a weak target echo are present in single channel time series data and the only prior information available about the reverberator is the approximate location and extent. For this case we derive a reverberation suppression algorithm which is based on the reduced-rank modeling of the reverberator transfer function followed by application of the Principal Component Inverse (PCI) method of reduced-rank adaptive interference cancellation. The algorithm is tested using simulated and real reverberation data.

1. Introduction

An important problem in active sonar is the detection of targets in the presence of bottom and surface reverberation. It is well known that active sonars using standard delay-sum beamforming and matched filtering perform poorly in the presence of strong reverberation. This performance loss generally arises from the suboptimum properties of standard sonar signal processing when strong 'signal-like' reverberation components are present (delay-sum beamforming and matched filtering are optimum only for detecting and localizing a single point scatterer in white gaussian noise). These 'signal-like' components arise from abrupt changes in ocean boundaries (e.g., cliffs, trenches, pinnacles, seamounts, facet-like surface waves etc.) [1]. The poor performance leads to masking of targets and high false alarm rates.

The design of an optimum receiver (e.g., likelihood-ratio test) requires the spatial-temporal statistical properties of the reverberation, ambient noise, and target acoustic pressure fields. Conceptually, the statistical properties of the acoustic back-scattered field are predictable if all the pertinent scattering physics, propagation physics, and ocean-acoustic parameters (ocean surface, bathymetry, water column properties etc.) were known. The difficulty in practice is that excessive complexity is required to exactly model the acoustic backscattering and propagation. An equally important problem is that the available bathymetric/environmental data is often incomplete and known with limited accuracy. For example, the sound velocity profile and geo-acoustic parameters are usually measured at a small number of locations. In addition, bathymetric surveys may miss small-scale bottom features which can be important reverberators. Hence the optimum receiver can not be implemented. However, it is conjectured that sufficient information is available about the scattering mechanisms, bathymetry, and ocean environment to provide useful predictions to aid in modeling and suppressing the reverberation components. For example, at least the 'coarse' reverberation structure (e.g., location, energy distribution and extent) should be predictable using acoustic backscattering models such as the two-way PE method [2] or *BISSM2* [3] in conjuction with bathymetric and environmental data bases.

The methodology that we propose for processing the reverberation data is to identify, categorize, model, and remove the reverberation components in a piece-wise fashion based on their ease of separability from the ambient noise and target signals. The modeling and estimation of the reverberation components are guided by backscatter model predictions and preliminary analysis of the data, such as standard beamforming and matched filtering. In essence, the estimation step is used to compensate for the uncertainty in the backscatter predictions.

This piece-wise approach to modeling and processing complicated data is similar to that proposed by Kirsteins and Tufts [4] for processing Arctic sea noise data and also by Middleton [5] for strong non-gaussian interference. Middleton proposed that one way of dealing with non-gaussian interference is to estimate and then subtract out the strong non-gaussian interference prior to signal processing and reduce the problem to one of processing in ambient noise [5]. The above methodology is also closely related to the residual signal analysis scheme of Costas [6] In [6] Costas considers the problem of recovering the individual signals that the received waveform is composed of, that is, the received waveform consists of a sum of several signals plus ambient noise and the problem is to estimate each of the signal components. He proposes a cooperative arrangement between estimation processors in which each processor acts as an adaptive interference canceller for all the other processors while estimating its own signal [6].

In this paper we develop the methodology for the case when strong, 'signal-like' or temporally localized reverberators plus a weak target echo are present in single channel time series data and the only prior information available about the reverberator is the approximate location and duration (see Figure 1). For this case we derive a novel algorithm for estimating and removing the strong, temporally localized reverberators. The algorithm is based on the reduced-rank modeling of the measured reverberator transfer function followed by application of the PCI method [8–12]. of reduced-rank interference cancellation.

The rest of the paper is organized as follows: In the next section we derive the reduced-rank reverberation suppression algorithm. We then present some simulated and real reverberation data examples in Section 3, and in Section 4 concluding remarks.

2. Reduced-Rank Estimation and Removal of Temporally Localized Reverberators

In this section we motivate the use of a reduced-rank model to represent the measured reverberator and ocean-acoustic channel transfer function, and then derive the reduced-rank reverberation suppression algorithm for the single channel case e.g., beam outputs or individual hydrophones. For a review and discussion on reduced-rank modeling the reader is referred to [15,16].

The observed reverberation plus target time series, denoted as $d(t)$, can be written as

$$d(t) = h(t) * s(t) + T(t) * s(t) + n(t), \qquad (1)$$

where the operator '$*$' denotes convolution, $h(t)$ is the combined impulse response of the reverberator(s) and ocean-acoustic channel, $T(t)$ is the combined impulse response of the target and ocean-acoustic channel, $s(t)$ is the transmitted signal, and $n(t)$ is some noise component (e.g., ambient noise). In the frequency domain the first convolution in (1) is

$$D(w) = H(w)\ S(w), \qquad (2)$$

where $D(w)$, $H(w)$, and $S(w)$ are the Fourier transforms of $d(t)$, $h(t)$, and $s(t)$ respectively. Typically $S(w)$ is restricted to some band, say between w_1 and w_2. Therefore we consider $H(w)$ only in the interval $[w_1, w_2]$. Next sample $H(w)$ to form the discrete sequence

$$H_n = H(w_1 + \Delta w n), \qquad n = 0, 1, \ldots N-1, \qquad (3)$$

where $w_2 = w_1 + \Delta w(N-1)$.

Let us now assume that only one strong, temporally localized reverberation component is present in $d(t)$ (1). More precisely, by temporally localized reverberation component we mean that most of the energy of the combined reverberator and ocean-acoustic channel impulse response is concentrated in a small time interval (e.g., features A and B in Figure 1). We now argue that the sequence $\{H_n\}$ can be approximated using a reduced-rank model when $h(t)$ is temporally localized. Our argument starts with the observation that since $h(t)$ can be expressed as

$$h(t) = \frac{1}{\pi} \int H(w) e^{iwt}\, dw, \qquad (4)$$

it can be regarded as a 'Fourier transform' of the 'waveform' $H(w)$ ($h(t)$ is the complex conjugate of the Fourier transform of $(1/2\pi)\overline{H}(w)$). If $h(t)$ is concentrated in some time interval $[\tau_1, \tau_2]$, we can then say that the 'waveform' $H(w)$ is approximately bandlimited. It is well known that a segment of N samples from a bandpass stationary random process with a rectangular power spectrum of bandwidth β and sampled at rate f_s can be accurately approximated using a linear expansion of only $(2\beta/f_s)N$ discrete prolate spheroidal sequences (DPSS) [16]. Hence from the above discussion, we infer that the data vector

$$\mathbf{h} = [H_0 H_1 \cdots H_{N-1}]^T \qquad (5)$$

should roughly be representable using a linear expansion of about r terms where,

$$r = \frac{(\tau_2 - \tau_1)}{f_s} N \tag{6}$$

and $f_s = 1/\Delta w$. Formula (6) can be simplified by noting that as $\Delta w \to 0$, it tends to

$$r = (\tau_2 - \tau_1)(w_2 - w_1) \tag{7}$$

recalling that $[w_1, w_2]$ is the signal band. It is emphasized that formula (7) is exact only for a bandpass stationary random process with a rectangular spectrum. In general it should only be used as a heuristic rule or bound to gain rough insight into the dimensionality of $H(w)$. The actual dimensionality will vary depending on the shape of $h(t)$.

We say that **h** is low rank when the dimensionality of **h** is much less than the dimensionality of the background noise (e.g., rank of noise covariance matrix) [15,16]. Therefore, **h** is expected to be low rank when $(\tau_2 - \tau_1)(w_2 - w_1) << N$ and the background noise is full rank (e.g., white noise). Thus the PCI method [8–12] can be used to remove temporally localized reverberators. A detailed review of the PCI method is provided in Appendix A.

First some preliminaries before presenting the reverberation suppression algorithm. The signal spectrum $S(w)$ within the band of interest $[w_1, w_2]$ is assumed to be non-zero everywhere and with no deep notches. The need for this will be seen later in the algorithm steps. Secondly, the target echo is assumed to be much weaker than the reverberator and the regions where strong, temporally localized reverberation components are present have been identified. The target echo is required to be weak to minimize the influence of the signal when estimating the reverberation component. We now give the steps of the reverberation suppression algorithm:

1. **Fourier transform the data**

$$\tilde{D}_k(w) = \int_{T_1^k}^{T_2^k} d(t) e^{-iwt}\, \mathrm{d}t, \tag{8}$$

where the interval $[T_1^k, T_2^k]$ encompasses the kth reverberation component in formula (1). The interval $[T_1^k, T_2^k]$ is determined from backscatter models and/or preliminary analysis of the data, e.g., matched filtering.

2. **Measurement of reverberator-channel and target transfer function**

Calculate

$$\tilde{G}_n^k = \frac{\tilde{D}_k(w_1 + \Delta w n)}{S(w_1 + \Delta w n)}, \qquad n = 0, 1, \ldots N-1, \tag{9}$$

It is important that $S(w)$ has no deep notches within $[w_1, w_2]$. The effect of deep notches is to enhance any noise components which may be present in $\tilde{D}_k(w)$.

3. **Apply PCI method to remove reverberation component**

The steps of the PCI method presented in Appendix A are applied to the sequence $\{\tilde{G}_n^k\}$ (9) to estimate the reverberator response H_n and then subtract it from $\{\tilde{G}_n^k\}$. The residual contains an estimate of the target response and background noise.

Formula (7) can be used to gain insight into the rank of the interference. The rank of the interference is more precisely determined by computing the singular value decomposition of the data matrix and then finding the number of dominant singular values (see Appendix A).

Discussion It is pointed out that the only prior information necessary to implement the algorithm is the approximate location and duration of the reverberator-channel response. This can be determined from preliminary analysis of the data or backscatter models. No information regarding the shape of the reverberator-channel response is needed.

Tufts et al. [7] also proposed using the PCI method to suppress reverberation. In [7] the PCI method is applied directly to the reverberation time series data. The new approach presented here applies the PCI method to the measured reverberator-channel transfer function and only requires that the reverberator-channel impulse response is temporally localized. It does not require that the reverberation time series possess any low rank properties.

The steps of the proposed processing methodology are now summarized. They are as follows:

1. Identify and locate the regions where strong, temporally localized reverberators are present using preliminary analysis, e.g., standard beamforming and matched filtering, and backscatter model reverberation level predictions. The validity of the reduced-rank model for the reverberator-channel transfer function is determined using formula (7) and from the number of dominant singular values the data matrix has (see Appendix A).
2. Estimate and remove the strong, temporally localized reverberators using the PCI method.
3. Perform target detection/localization on the residual data.

3. Experimental Results

We now present some simulated and real data examples. We start by giving an example to show how the low rank structure of the PCI method data matrix (12) can arise.

Low Rank Data Matrix Suppose our received waveform is

$$d(t) = \alpha_1 s(t - \tau_1) + \alpha_2 s(t - \tau_2), \tag{10}$$

for example, arising from two point scatterers over direct path propagation. It can then be shown that

$$H(w) = \alpha_1 e^{-i\tau_1 w} + \alpha_2 e^{-i\tau_2 w}. \tag{11}$$

If we sample (11) over some interval and arrange the samples into the matrix given by formula (12), it is easy to show the data matrix is rank 2 (e.g., every row of the data matrix is some linear combination of the same two discrete exponential sequences).

Computer Generated Data Next we present a computer generated example where a weak target echo is recovered in the presence of a strong, highly temporally localized reverberator plus white Gaussian noise. The bandwidth of the reverberator is chosen to be 0.0488 Hz and the sampling rate is 1 Hz. The envelopes of the entire observed impulse response (reverberator, target and noise) and the target echo are plotted in Figure 2a. Note that the signal has been deconvolved from the data. The PCI method using a rank of 5 is applied to remove the reverberator and the residual is plotted in Figure 2b. Note that the target echo is distinctly observable in the residual.

Real Reverberation We now apply the algorithm to some real shallow water reverberation data. The data was collected in area where the average depth was about 110 m. The reverberation data used in this example is the output basebanded time series from a single beam. The transmitted waveform is a 375–395 Hz HFM pulse of 2.45 s duration. In Figure 3 the envelope of the matched filter output for a single beam is plotted. It can be seen that there are several distinct peaks present in the matched filter output, particularly the feature labeled ‘A’. We now use the PCI method to estimate and remove the reverberation component corresponding to peak ‘A’.

To apply the algorithm, a 5.3 s segment of data about the reverberator was taken and its Fourier transform evaluated at 100 equi-spaced points in the interval 375–395 Hz. The measured impulse response after the signal has been deconvolved from the data is plotted in Figure 4 (see step 2 of algorithm). Insight into the rank of the reverberator can now be determined using the measured reverberator response (Figure 4) to estimate the reverberator duration ($\sim$ 0.4 s) and then substituting it into formula (7), obtaining $r = 8$ ($= 0.4 \times 20$). The data matrix (see step 3 of algorithm) had actually 4 dominant singular values so a rank of 4 was used to implement the PCI method. The estimated reverberator and residual impulse response are plotted in Figures 5 and 6. It can be seen from Figure 6 that the reverberator has been accurately removed.

4. Conclusion

A new algorithm has been presented for suppressing strong, temporally localized reverberation components. The only prior information the algorithm requires is the approximate location and duration of the reverberation component. This can be obtained from preliminary analysis, e.g., beamforming and matched filtering or backscatter model predictions. Future work includes extending the algorithm to the strong target echo case and performance analysis.

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A. Review of PCI Method

The PCI method [8–12] of adaptive interference cancellation and detection is based on reduced-rank nulling of interference. The PCI method exploits the low rank structure of the data covariance matrix or equivalently, of the data matrix. An important advantage of the PCI method over conventional methods such as adaptive control loops and the Sample Matrix Inverse method is that it achieves a much more rapid rate of adaptation [8–12].

In the PCI method, the interference is first regarded as a 'signal' to be enhanced. Then reduced-rank signal enhancement [10,13,14] is applied to obtain an estimate of the interference. This interference is subtracted from the observed data and the residual is then processed to extract the desired signal information, e.g., presence of signal. The key idea in the signal enhancement algorithm [10,13,14] and the PCI method is to arrange the data samples in some matrix form which exploits the low rank structure of the data. More specifically, the steps for the time series version of the PCI method are as follows:

1. **Construction of Data Matrix**

   The data sequence $\{H_n\}_{n=0}^{N-1}$ is arranged into the forward-backward matrix

$$\mathcal{H} = \begin{bmatrix} H_{L-1} & H_{L-2} & \dots & H_0 \\ H_L & H_{L-1} & \dots & H_1 \\ \vdots & \vdots & & \vdots \\ H_{N-1} & H_{N-2} & \dots & H_{N-L} \\ \overline{H}_0 & \overline{H}_1 & \dots & \overline{H}_{L-1} \\ \overline{H}_1 & \overline{H}_2 & \dots & \overline{H}_L \\ \vdots & \vdots & & \vdots \\ \overline{H}_{N-L} & \overline{H}_{N-L+1} & \dots & \overline{H}_{N-1} \end{bmatrix}, \tag{12}$$

where L is the number of columns and '$\overline{\ }$' denotes complex conjugate.

2. **Estimating the Interference Component**

   The interference waveform is estimated by arithmetically averaging all multiple occurences of each data sample in the low rank approximation to $\mathcal{H}$, denoted as $\mathcal{H}_r$, which is found as the solution to

$$\min_{\substack{\mathcal{H}_r \\ \mathcal{H}_r \text{ subject to rank}[\mathcal{H}_r] = r}} \| \mathcal{H} - \mathcal{H}_r \|_F^2 . \tag{13}$$

The solution to (13) can be easily calculated using the singular value decomposition (SVD) of the data matrix $\mathcal{H}$. Also, the rank 'r' can be estimated by determining the number of dominant singular values present.

3. **Removing the Interference Component**

   The estimated interference waveform, denoted as $\tilde{H}_n$, is subtracted from the data:

$$R_n = H_n - \tilde{H}_n. \tag{14}$$

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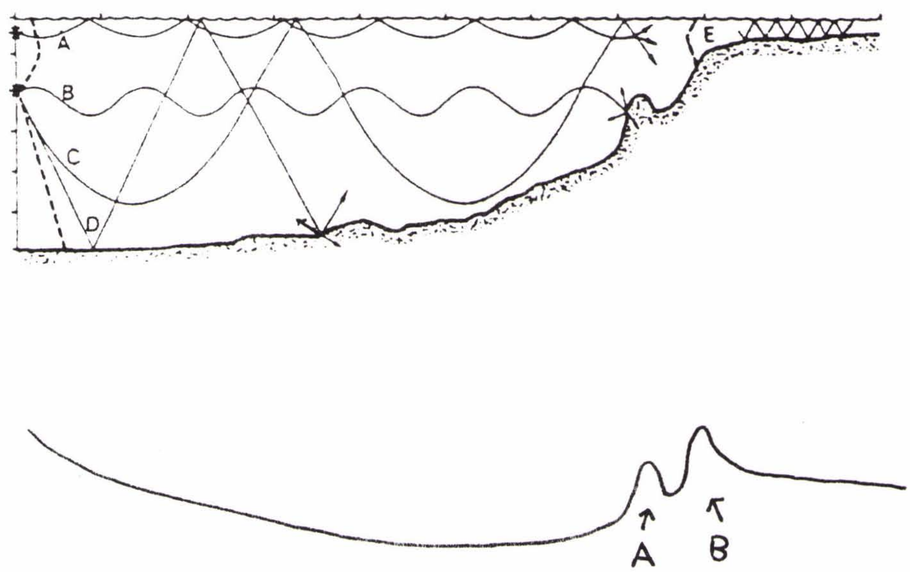

reverberation level

Figure 1. Example of prior information available.

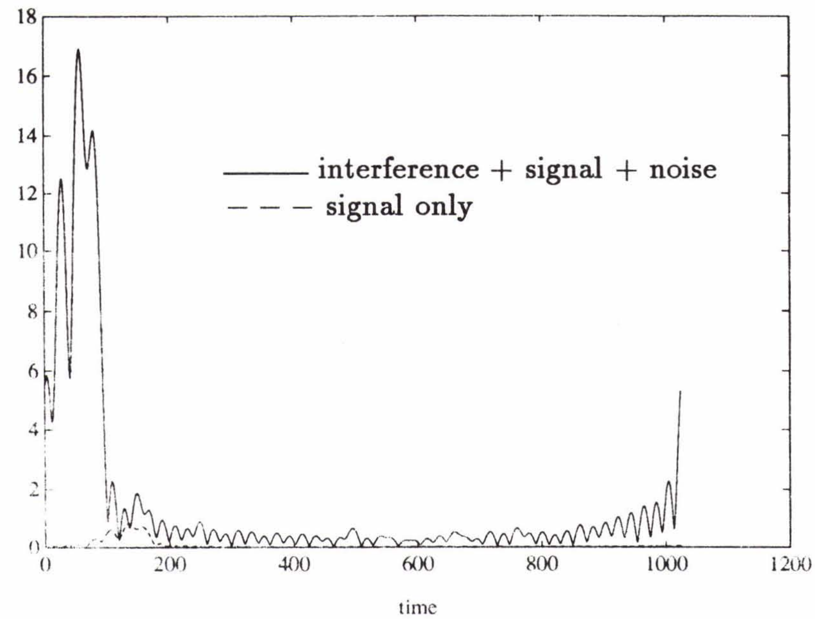

Fig. 2a) Interference, signal and noise impulse response.

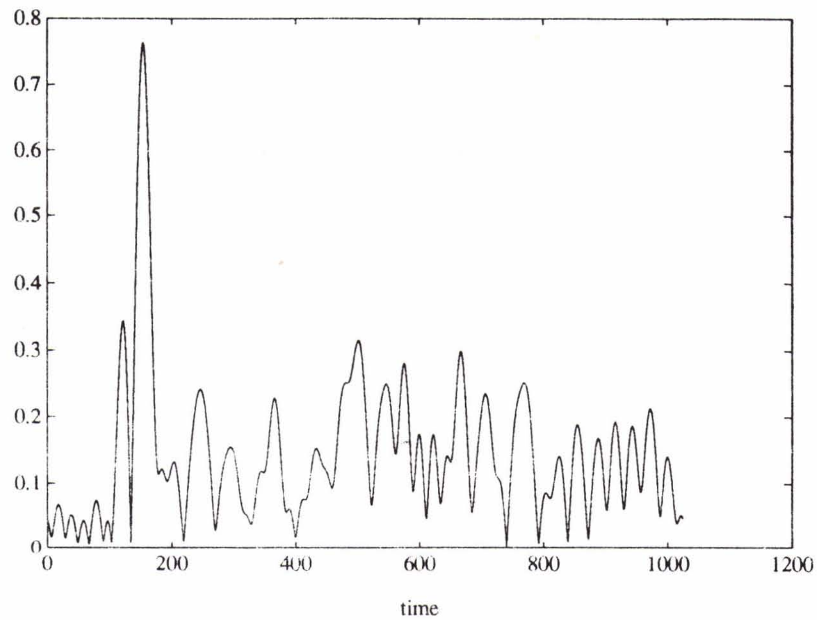

Fig. 2b) Residual impulse response after estimated interference has been subtracted.

Figure 2. Computer generated data example.

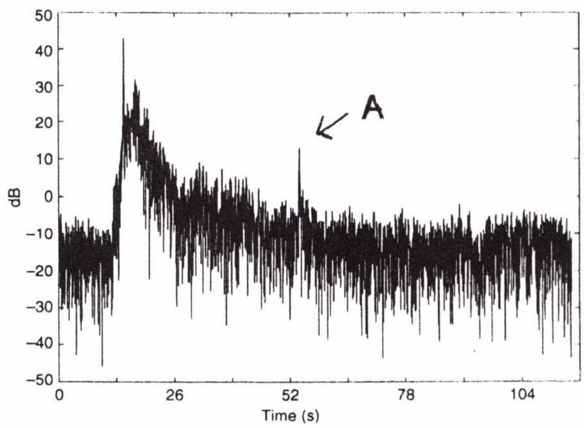

Figure 3. Single beam matched filter output.

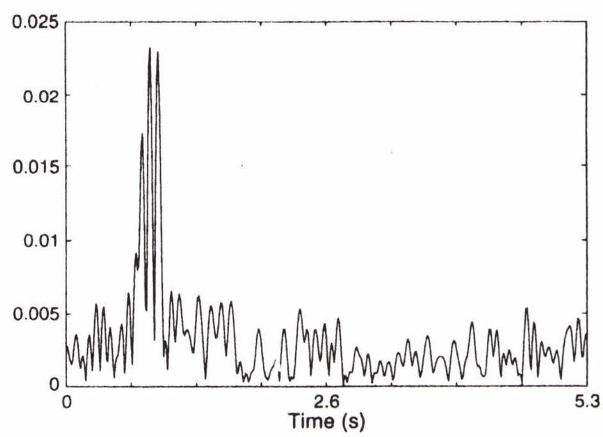

Figure 4. Measured impulse response of reverberator.

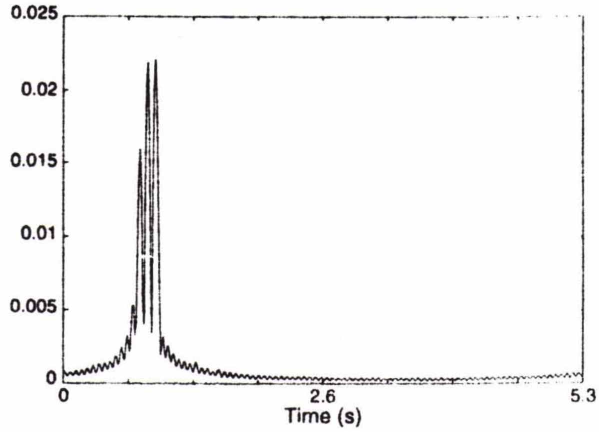

Figure 5. Estimated reverberator impulse response using reduced-rank signal enhancement.

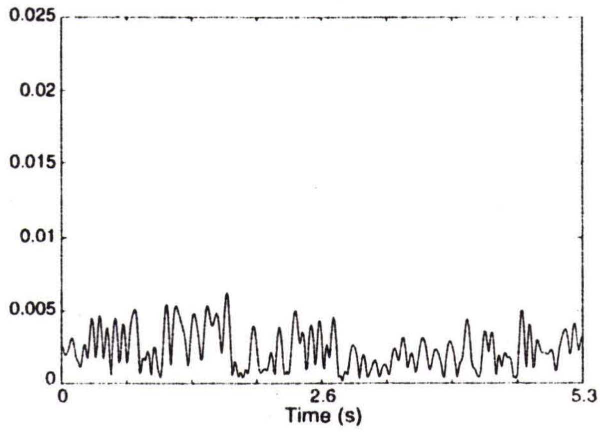

Figure 6. Residual impulse response after estimated reverberator component has been subtracted.