International Journal of Engineering and Techniques - Volume 3 Issue 2, March-April 2017

RESEARCH ARTICLE OPEN ACCESS

DWT and EMD for Deghosting of Seismic Signals Analysing Vibration and Geomagnetic Anomalies

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Abstract:

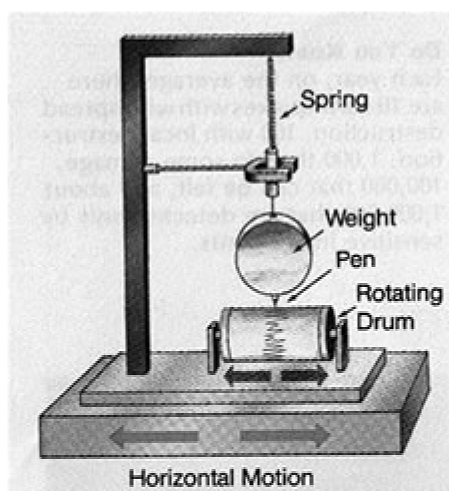
This project is about the seismic wave signal Parameter enhancement with vibration analysis and geomagnetic signal anomalies. In this project, we are going to detect the seismic signal using seismograph. The ghosting effects were occurring and it will be suppressed using the filters. We propose to show the benefit of 1D convolutional filter, to remove all the non-energetic wave-field in order to provide a better imaging of the reflecting wave-field. In this paper, wave signals are decomposed into intrinsic (characteristic) modes via Discrete Wavelet Transform [4] (DWT), Empirical Mode Decomposition [1] (EMD) and the relationship between seismic activities are investigated.

Keywords — ***DWT, EMD, GHOSTING EFFECTS***

I. INTRODUCTION

An earthquake is a natural disaster caused by an unexpected release of seismic energy due to extreme stress within the earth’s crust. Such energy is released because of aggressive movements of the tectonic plates in active fault zones. The accumulated energy, containing immense pressure, is transferred from the earth’s crust to its surface in the form of seismic waves. These waves can be either rolling or travelling parallel to the surface, which lead to the destruction of anything that falls within its path. The complexity of seismic data processing has contributed to the development of several efficient signal processing tools such as wavelet transforms or spike deconvolution. When the seismic trace (1-D signal) density is high, the high similarity enables the design of a great variety of filters depends on the signal slope, to increase both the signal-to-noise and the signal-to-interference ratios. Common applications on seismic signals processing are noise filtering and migration operations. The application under consideration is a seismic equivalent to occlusiosn removal. Coherent noises are caused by peculiar wave propagation. They arise as structured signals in seismic records, hampering subsequent geophysical data processing and interpretation with high amplitude directional band-pass stripes. This work investigates their removal with two breeds of selective multiscalar directional decompositions. They further allow to enlighten some similarities between traditional and geophysical data processing that deserve further investigation, for the benefit of signal processing

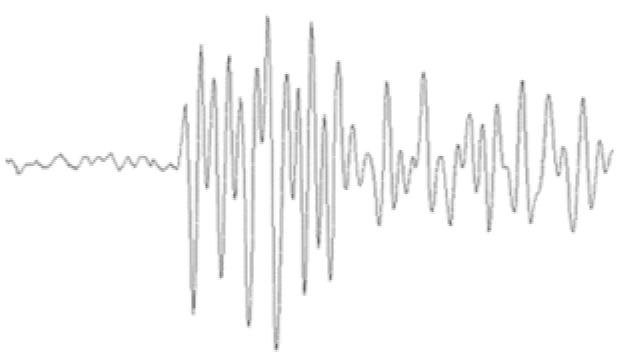
A. How are earthquakes recorded?



Earthquakes are recorded by instruments called seismographs. The graph output of seismograph is known as seismogram. The seismograph has a base that sets firmly in the ground, and a heavy weight that hangs free. When an earthquake causes the ground to shake, the base of the seismograph shakes too. Instead the spring or string that it is hanging from absorbs all the movement. The difference in positions between the shaking part of the seismograph and the motionless part will be recorded.

B. How do scientists measure the size of earthquakes?

The size of an earthquake depends on the size of the fault and the amount of slip on the fault, but scientists can not be simply measuring with a measuring tape since faults are many kilometers deep beneath the earth's surface. So, how do they measure an earthquake? The use of seismogram recordings made on the seismographs at the surface of the earth to determine how large the earthquake was. A short wiggly line means a small earthquake, and if the wiggly line is long means a large earthquake. The length of the wiggle depends on the size of the fault, and the wiggly size depends on the amount of slip.



The size of the earthquake is known as magnitude. There is one magnitude for each earthquake. Scientists also talk about the intensity of shaking from an earthquake, and this varies will be depending on where you are during the earthquake.

C. Seismic signals

The surface of the Earth is in constant slight movement and the motion at any point arises from both local effects, e.g., disturbances made by mankind or wind-induced rocking of trees. If we look at the records from an observatory seismometer in its carefully constructed vault, or from a geophone whose spike is simply driven into the ground, we find that these largely consist of such seismic 'noise'. This feature arises from the excitation of seismic waves, away from the receiver, by some natural or artificial source.

D. Noisy Seismic signals

The Radiation of signal from a seismic source, be it an explosion or a shear rupture, is usually a more or less complicated displacement step function or velocity impulse of finite duration from milliseconds up to a few minutes at the most. According to the Fourier theorem any arbitrary transient function f(t) in the time domain can be represented by an equivalent function F(w) in the frequency domain, the Fourier transform of f(t). The following relations are:

$$f(t) = (2\pi)^{-1} \int_{-\infty}^{\infty} F(\omega)\, \exp(i\omega t)\, d\omega$$

$$F(\omega) = \int_{-\infty}^{\infty} f(t) \exp(-i\omega t) dt = |F(\omega)| \exp(i\phi(\omega))$$

E. Seismogram analysis

The primary sources of information in seismology are the records of seismic events obtained at the Earth's surface. To model the nature of the wave trains recorded by a seismometer, we have to take

an account of the entire process thereby the seismic energy reaches the recording site. This may be divided into three major elements. First is the generation of the waves by the source, secondly, the passage of the waves through the Earth to the vicinity of the receiver and finally to detect and record the characteristics of the receiver itself.

In this paper, the analysis of the seismic wave for the numerical analysis is based on linear time-frequency presentation and the signal toolbox in Matlab, the results of obtained are discussed and the measures are presented.

II. EXISTING SYSTEM

LMS Filter (Least Mean Square Filter)

The Least mean square (LMS), Recursive least square (RLS), Fast Transversal Recursive least square (FTRLS) are the existing filters. Implementation aspects of these algorithms, where their computational complexity and Signal to Noise ratio are examined. These algorithms use small input and output delay.

III. DISADVANTGES OF EXISTING SYSTEM

- Requires large amounts of memory to store seismic files due to the added noise;
- Makes errors, can be frustrating without adequate support;
- Even though the target seismic signal is present in the environment, this method permits the prediction of noise matrix for non-stationary noise sources. The noise matrix is calculated by estimating the power spectral density of the noise present in the filter.
- The power spectral density is poor in the LMS filter.

IV. PROPOSED SYSTEM

Proposed Algorithm 1

- 1D convolutional filter

Proposed Algorithm 2

- The vibration analysis using DWT EMD

A. Proposed Algorithm 1

1D convolutional filter:

Convolution is a formal mathematical operation, like multiplication, addition, and integration. Addition takes two numbers and produces a third number as output, while convolution takes two signals and produces a third signal. In linear systems, convolution is used to explain the relationship between three different signals: the input signal, the impulse response, and the output signal.

Figure below shows the notation when convolution is used with linear systems. An input signal, x[n], enters a linear system with an impulse response, h[n], which will result in an output signal, y[n]. In equation form: x[n] * h[n] = y[n]. Expression in words, the input signal convolved into the impulse response which is equal to the output signal. Addition is represented by the plus, +, and multiplication by the cross, × and then convolution is represented by the star, *.A star in a computer program means multiplication, while in an equation a star means convolution.

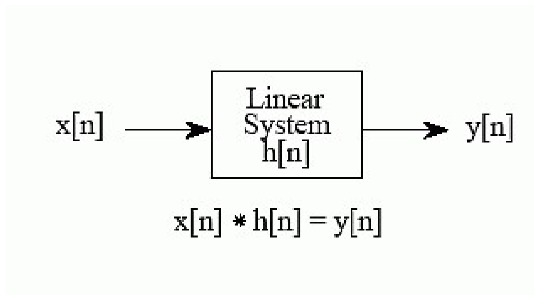


Figure illustrates, the two additional examples of how convolution is used to process signals. The inverting attenuator, flips the signal top-for-bottom, and reduces its amplitude. The discrete derivative (also called the first difference), results in an output signal related to the slope of the input signal were the feature adopted[2]

B. Proposed Algorithm 2

The vibration analysis using DWT and EMD:

The EMD procedure separates the original non-stationary signal into a finite sum of IMFs, which is based on local natural properties of the signal. Each IMF should satisfy the following two conditions:

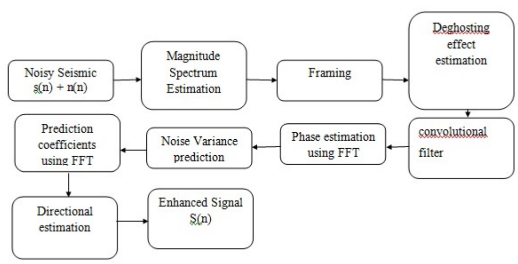
1) The number of the extreme and the number of the zero crossings of the function must be equal or at least differ by one.

2) At any point of the function, the mean value of the envelopes defined by the local

extreme should be zero. The first condition ensures that all the maxima have positive values and all the minima have negative values. The second condition ensures the physical meaning of instantaneous frequency through the local symmetry of the function with respect to zero.

The process of the EMD decomposition of the original signal X(t) can be divided into several finite computational steps, which are briefly described below. First, identify all the local extreme and connect all the local maxima by a cubic line , defined as the upper envelope ymax(t). Respectively, connect all the local minima to produce the lower envelope ymin(t). All the data should be enclosed between the aforementioned envelopes.

V. BLOCK DIAGRAM



Magnitude Spectrum Estimation

The magnitude of a variable, on the other hand, is the measure of how far, regardless of direction, its quantity differs from zero. So magnitudes are always positive values.

Framing

Convert the signal to number of frames which is decided by the length of the signal.

Non Negative Factorization window coefficients

Non-negative matrix factorization (NMF), also known as non-negative matrix approximation is a group of algorithms in multivariate analysis and linear algebra.

Window coefficients

The DFT/FFT contains an implicit periodic extension and the periodic flag enables a signal window with a periodic window to have perfect periodic extension. When 'periodic' is specified, hamming computes a length L+1 window and returns the first L points

Phase estimation using FFT

We've mentioned that most FFTs return their information in the form of a complex pair: a real part and an imaginary part. We have been discussing about the data that an FFT returns more in terms of the amplitude (or magnitude) and phase of a given frequency bin. The importance of the discussion is that we understand more or less how to get from the complex number pair to the magnitude/phase pair, which is generally, for computer music, more useful.

Noise Variance prediction

Seismic wave noise is estimated here.

Prediction coefficients using FFT

After converting into frequency domain, the harmonics of the signal is estimated.

Directional estimation

The direction of the velocity of the seismic wave is estimated.

Enhanced Signal

The noisy seismic wave signal is enhanced.

VI. PARAMETER ESTIMATION

The estimation of parameters of interest relying on noisy observations is a central task in statistical signal processing. An estimator is a procedure that relies on the noisy measurements to provide an estimate of the parameter of interest on wavelet decomposition[5]. Mathematically, an estimator is a function g(·) of the measurements ˜y providing an estimate ˆθ of the value of the unknown parameter θ, i.e., ˆθ =g(˜y). At least two properties of an estimator are desirable: unbiased and small mean-squared estimation error (MSEE). Bias is a measures of the average deviation of the estimate from the true value and is defined as

$$\text{Bias}(\hat{\theta}) = \mathsf{E}\{\hat{\theta} - \theta\}, \quad (1)$$

where E{·} denotes the expected value operator. An estimator is unbiased whenever on the average it provides the right value, i.e., Bias(ˆθ) = 0. Otherwise the estimator is said to be biased. The MSEE measures the average of the square of errors and is defined as

$$\mathrm{MSEE}(\hat{\theta}) = \mathsf{E}\{(\hat{\theta} - \mathsf{E}\{\hat{\theta}\})^2\}\,. \quad (2)$$

It is desirable for an estimator to have small MSEE as this implies smaller fluctuations around the expected value E{ˆθ}. Such limit depends on the statistical model and is independent of the estimation technique used and algorithmic implementation. MSEE achieves the smallest possible value indicated by the CRB. We denote with pY (y, θ) the PDF of the measurements where Y are random variables and θ is a deterministic parameter. The second step is to compute the likelihood of the observations. The LF of the observations is a function of the parameter θ

$$\ell(\theta) = p_{\mathbf{Y}}(\tilde{y}, \theta)\,,$$

and is defined as

where ˜y denotes the observed measurements. The final step, is to maximize the LF over the parameter space.

$$\hat{\theta}_{\mathrm{ML}} = \underset{\theta}{\mathrm{argmax}}\,\ell(\theta)\,,$$

where ˆθML is the ML estimate of the parameter. The maximization in (1.4) may be tackled by different means. At best, an analytic solution may be found.

VII. CONCLUSION

Time-scale directional filters are a powerful tool that can significantly improve seismic data processing, thanks to the enhancement in the seismic wave detection, separation and tracking capabilities. The two techniques used in this context yield two complementary approaches, with a different balance in slowness resolution and redundancy. Both are able to attenuate the main part of the chevron-like coherent noise occluding meaningful geologic information. The time-scale LSST will be focusing on a reduced distortion level of the signal of interest. The similarity of the time-scale LSST with that of other directional transforms used in image processing, the quest for a vaster family of adaptive transforms linking proposed methods deserves further investigations.

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