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Application of Pattern Filtering Method in Cross-hole Tomography Signal Processing

Miaorong Lv and Xu Liu
College of Petroleum Engineering, Changzhou University, Changzhou, China

***Abstract*—Noise reduction and signal separation are very important basal works in signal processing. Pattern Filtering Method（PFM） , which is introduced as a new method for signal processing in this article, is applied to cross-hole tomography signal processing. Acoustic Computed Tomography (CT) signal with noise disturbance, which is acquired from a mine, has been processed by means of PFM, different signals and noises have been separated reasonably. A powerful Pattern Filtering software has been developed by Delphi tool, and batch signals can be more efficiently processed. It is showed that PFM is a very effective tool in cross-hole tomography signal processing, and is prospective for utilization in other realms of engineering.**

Keywords—signal processing; cross-hole tomography; Pattern Filtering Method; time-frequency sub-wave; signal separation

I. INTRODUCTION

Filtering is necessary and effective for reducing noise in signal processing. The essence of filtering is signal separation in order to eliminate interference and obtain useful signal. Common filtering methods, which has been widely used, can be divided into two categories: Winner filtering and Kalman filtering [1]. However, most of engineering signal and environmental interference have wide continuous spectrum distribution. When using these methods to process signals, there are two obvious shortcomings[2]: (1) It is difficult to take the physical properties and the dynamic features of signal source into consideration; (2) there are distortion, aliasing and the lack of components after filtering. There are noise and interference everywhere in engineering measuring and testing. The first problem encountered in signal processing is how to separate them reasonably. It is unlikely to eliminate interference under the premise of ensuring the integrity of the filtering signal depending on these filtering methods. Research shows that the applicability of various filtering methods can be limited by the characteristics of signal itself. And the selection of both filtering model and filtering method is a very important procedure that should be paid attention to in signal processing. Otherwise, unreasonable signal model is responsible for additional processing noise, and improper signal processing method will also lead to mixture of signals with different physical properties and reduce the quality of separated signals.

A reasonable signal model which can well describe the signal source features has been established. Combined with this signal model, two or more aliasing signals can be separated to some extent of satisfaction by the optimized method called PFM, which is proposed in order to enhance the maneuverability of signal separation. This method has been successfully applied in the signal processing of speech and mechanical vibration[2]. Various speech components as well as different Mandarin monosyllabic consonants and vowels are reasonably separated.

More importantly, all signals processed by PFM are single-channel signal. Single-channel signal separation is one of the rapid developing branches; it is also a fundamental and challenging subject in signal processing in recent years. Applying PFM to the cross-hole tomography signal processing is the key point of this article.

The separation of various single-channel signals and the extraction of signal parameters are introduced as follows.

II. PRINCIPLE OF PFM

A. Basic Conceptions

Pattern filtering is to establish a reasonable signal model based on the physical characteristics of signal, and regard the model as a time-frequency sub-wave (TFSW) or an operator. A signal is decomposed into a number of TFSWs, in other words, a series of TFSW parameters are extracted from the processed signal by some optimization method. Through the reasonable selection, classification and reconstruction of these TFSWs, the signal extraction, separation and noise reduction are realized.

In the acoustic signal processing, we call this TFSW as phonon. The results show that signal can be regarded as a certain way of phonon superposition arranged in on the axis of time. More importantly, we can use these phonons to classify, select, and reconstruct different separated signals according to the natural source. The following form of TFSW is proposed:

$$y(t) = f(A,\alpha,t) \times g(\sum_{i=1}^{n} \beta_i \times t^{i-1}) \tag{1a}$$

Where, f(.) is an AM function; g(.) is a FM function; A: Amplitude; α: attenuation factor, s^{-2}, β_i: coefficient; β_1: initial phase, rad; β_2: angular velocity (rad.s^{-1}). n: generally not greater 3; t: time, s.

In this paper, we use the following simplified TFSW formula:

$$y(t) = e^{-\alpha(t-t_0)^2}\cos(\beta_2 t + \beta_1) \tag{1b}$$

The basic idea of pattern filtering is to decompose signal into a number of TFSW (or phonon), and to use formula (2) to approximate the measured signal:

$$F(t) \approx \sum_{i=1}^{m} f(A_i, \alpha_i, t) \times g(\sum_{j=1}^{n} \beta_{ij} t^{j-1}) \tag{2}$$

Where, the A_i, α_i, β_{ij} coefficients in the formula are determined by an optimization method. F(t) is the signal function.

B. PFM Processing for Engineering Signal

1) Signal Acquiring

Computerized Tomography (CT) is based on the theory of wave propagation, which studies the variation of wave propagation in underground medium. The variation of physical parameters at the boundary between boreholes is obtained from measured acoustic signals. The area between the two parallel boreholes is discretized into several regular grid elements according to the geological conditions, test conditions and detection accuracy requirements. These grid nodes at the two holes are used as the acoustic transmitting point and the receiving point of the CT test, respectively. When the acoustic wave is emitted at any node, all the receiving points should check or pick up the acoustic signal generated by the transmitting point.

The instrument used in this detection is DST- 4 acoustic detection system. The system includes underground EDM sound source instrument, cable winch, underground receiving pipe instrument, and ground microcomputer controlled data acquisition instrument. The system has the characteristics of high reception sensitivity, strong anti-interference ability, high sampling rate and high conversion accuracy. This acoustic CT test was accomplished in a mine in China. The measured signals in Fig. 2 to Fig 4 were all acquired from this test.

2) Calculation of TFSW Parameters

A series of TFSW parameters in formula (2) can be extracted from a measured signal under a certain precision by using the optimization theory. The optimal objective function can be expressed as:

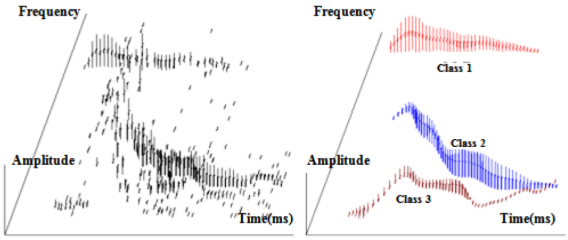
$$\min\text{:}\quad \sum_{k=1}^{N}\left\{F_k - \sum_{i=1}^{m} f(A_{ik}, \alpha_{ik}, t) \times g(\sum_{j=1}^{n} \beta_{jik} t^{j-1})\right\}^2 \tag{3a}$$

$$s.t.\begin{cases} 0 \le A_{ik} \le A_{\max} \\ 0 \le \alpha_{ik} \le \alpha_{\max} \\ \beta_{j\min} \le \beta_{jik} \le \beta_{j\max} \quad (1 \le j \le 3) \end{cases} \tag{3b}$$

Where, A_{jik}, α_{jik}, β_{jik} are the TFSW parameters. F_k is the k value of the residual time series which is the i-1 time optimal extraction of the original signal.

The optimal TFSW parameters can be calculated by any constrained optimization method, such as Newton's method, Quasi-Newton Methods and Davidon-Fletcher-Powell algorithm. This is a multi-step iterative calculation. When the energy of the remaining signal is less than a certain value, this calculation terminates. Here, we use a low amplitude as the threshold value.

The process of solving the optimization problem of formula (3) is equivalent to selecting a number of (α_{ik}, β_{jik}) sieves with different mesh size for the signal, screening and filtering according to the combinations of (α_{ik}, β_{jik}) parameters. Therefore, this method is vividly called pattern filtering. The results show that these TFSWs obtained by pattern filtering can simulate the original signal very realistically, and PFM can filter the random background noise quickly with a given precision.



(a) Pattern filter calculation (b)Smoothing and tuning processing

FIGURE I. DISPLAY DIAGRAM OF CHINESE SYLLABLE "YÀO"

3) Display and processing of TFSW

Fig. 1(a) is the TFSW distribution of Chinese syllable "yào" signal decomposed by formula (3) in a time-frequency-amplitude three-dimensional coordinates. The author omits the dimension and numerical value for convenience since they do not affect the display diagram.

A multiple effective operations can be carried out on the TFSWs. For example, parameter smoothing and modification can be operated, which will be introduced later. the processed signal can simulate the original signal as realistically as possible in Fig. 1(b). Classes 1, 2 and 3 in the figure represent three kinds of TFSW that can be handled in a feature set respectively. Further detailed research on these three kinds of TFSW can be carried out in order to obtain more information of the subject investigated [4][5].

III. APPLICATION OF PFM ON ENGINEERING SIGNAL PROCESSING

How to eliminate different interference during signal processing is a very important problem. The author applies PFM to cross-hole tomography signal denoising, weak signal extraction and signal separation.

A. Extraction of Signal Parameters by PFM

Fig. 2(a) is the 20 cross-hole CT signals acquired with very small interference, where the amplitude range of the signal is [-2000, 2095]. PFM calculation of the 20 signals was carried out when the low limit value is set 50. After decomposing a signal into TFSWs, the preliminary reconstruction signals are obtained in Fig. 2(b) and the corresponding noise is shown in Fig. 2(c). It can be seen that PFM can eliminate the background interference very effectively.

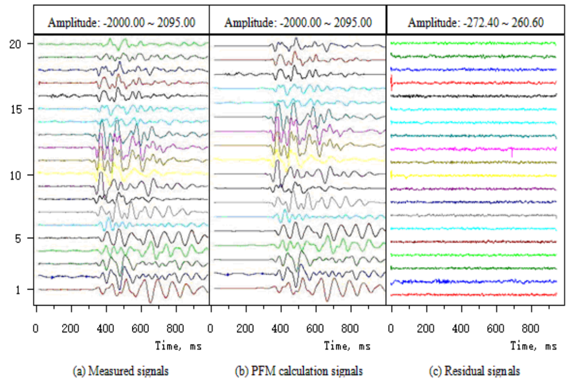


FIGURE II. CT SIGNALS WITH LITTLE NOISE, PROCESSED SIGNALS AND BACKGROUND NOISE AFTER PFM

Generally, there is no waveform in time domain before the take-off point because the measured acoustic signal does not reach the receiving sensor yet. Therefore the TFSWs which appear before the take-off point can be removed as interference directly. Through this processing, the final pattern filtering results are shown in Fig. 3. It can be considered that most of the TFSW obtained by PFM are effective because the interference is very small in Fig. 3. The TFSW of each signal obtained by PFM are summarized, and all kinds of small probability TFSW are excluded so as to realize the second screening by use of the principle of statistics.

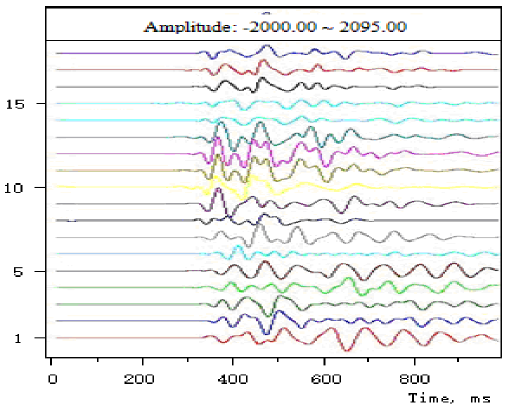


FIGURE III. DELETE INVALID TFSW BEFORE TOOK-OFF POINTS

TABLE I. STATISTICAL RESULTS OF EFFECTIVE TFSW PARAMETERS

	α/s^{-2}	β_1/rad	$\beta_2/rad.s^{-1}$	$A/m.s^{-2}$
Minimum	0. 000	−1. 353	0. 166	1205. 36
Maximum	1. 029	1. 230	2. 169	85591. 19
Average	0. 210	−0. 0165	1. 235	18408. 57

It also shows that although noise signal has a large number of TFSW, but the amplitude of the signal composed of them is very small, and is very limited on the influence of the filtering results. Table 1 can be obtained by statistical analysis of the effective TFSW parameters. The PFM processing of noisy signals can be carried out expediently by using this statistical result, which provides a scientific basis for eliminating the noise existing in these signals.

A number of TFSWs can be obtained under a certain filtering threshold by PFM. The reconstruction signal is the superposition of these TFSWs in the working time. The residual signal, from which the original signal subtracts the reconstructed one, can be considered as background noise. We call this reconstruction signal as primary filtering signal, and call the corresponding background noise as primary filtering noise. The intensity of background noise is determined by the filtering threshold. The smaller the threshold is, the more accurate the primary filtering signal will be.

Obviously, there will be more or less noise in the primary filtering signal, such as the fluctuation and divergence distribution of the TFSWs shown in Fig.1(a). However, the combination, optimization, superposition and smoothing of TFSW can be operated in order to eliminate the residual noise. The author calls such residual noise as the secondary filtering noise. The regular distribution of all series of TFSWs can be obtained after secondary or multi-step PFM operation. Each TFSW series has obvious physical meaning and can simulate the actual signals realistically.

B. PFM Processing of Signals with Interference

It is inevitable to be affected by various interference during signal acquiring and processing. Fig. 4(a) is the signal measured in the same wellbore as Fig. 2, but there is obvious noise in signals. After the TFSW of a signal are decomposed by PFM, the TFSW are filtered and checked by using the statistical parameters in Table 1. The results in Fig. 4(b) and the noise in Fig. 4(c) can be obtained.

The practice shows that it is a very effective way to separate the useful signal and noise by using the measured signal with small interference and the phonon statistical parameters in Table 1.

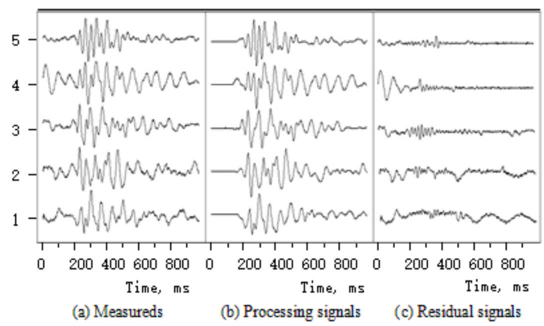


FIGURE IV. PFM PROCESSING WITH INTERFERENCE

C. Signal Extraction from Strong Background Interference

During some time of cross-hole tomography measuring in the mine, strong noise encountered, which brought great trouble to signal recognition and CT interpretation. Fig. 5(a) is some of these examples, from which you can see the rough position of the acoustic signal, but how to extract the useful signal from such a strong interference?

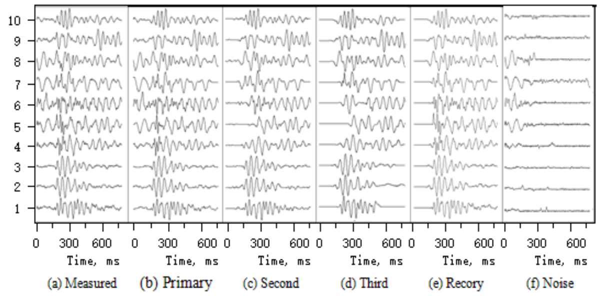


FIGURE V. SIGNAL PROCESSING WITH STRONG BACKGROUND INTERFERENCE

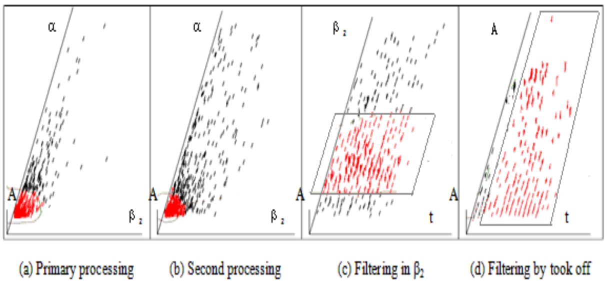


FIGURE VI. PFM PROCESSING STEPS OF SIGNAL UNDER STRONG NOISE

In fact, it is a relatively simple process to separate noise from useful signals by PFM. In order to extract cross-hole tomography signal from the strong noise. we take the 10 measured signals in Fig. 6(a) as an example, in which the smaller interference signals are marked No.1, 2, 8, 9, 10, and the rest is the stronger interference signal. The steps for PFM processing of these signals are as follows:

(1) These signals are read into computer memory and each signal is analyzed by PFM calculation.

(2) The calculated TFSWs are summarized and displayed in a three-dimensional space of "α-β_2-A" in Fig.6(a) and the primary filtering signals are shown in Fig. 5(b).

(3) It can be seen that the phonons of these 10 signals are concentrated in the lower left corner of the "α-β_2" plane. In order to exclude invalid phonons, these red-marked phonons are considered to be valid in the trap line as shown in Fig. 6(a). Other phonons are thought to be noise. Fig. 5(c) shows the corresponding reconstruction signals after the Second PFM processing.

(4) According to the information provided in Table 1, the TFSWs which are delineated in the range of $\alpha\in$ [0.001.20] and $\beta_2\in$ [0.10, 2.50] can be checked as valid phonons. The valid TFSWs are obtained in Fig. 6(c) and other phonons can be deleted.

(5) The remain TFSWs can be further filtered by the rough judgment of the take-off point because there are no any valid phonon arrived before this time in Fig. 6(d). These invalid TFSWs are deleted, and the corresponding reconstruction signals are shown in Fig. 5(d).

(6) From the comparison between processing signals and original signals, it can be found that some effective phonons have also been deleted due to PFM miss operation. The useful phonon can be recovered by using a similar step such as (1) ~ (5), and finally the recovery signal in Fig. 5(e) can be obtained. It is very difficult to distinguish whether a phonon is valid or not. The principle of TFSW separation and combination will be introduced in subsequent articles, and literature [2][6] have some introduction. Finally, the interference signals are shown in Fig. 5(f).

It can be seen that valid signals and noise can be separated effectively after limited steps of PFM processing as long as the parameter distribution range of valid TFSW is defined. The operation is very simple and fast. PFM processing is also a self-learning process. The more sophisticated the operator is, the higher processing efficiency can be. PFM can be used to extract the useful signal from the measured signal and eliminate various noise.

The above-mentioned process is implemented on PFM software developed by Delphi, which is divided into three large modules, namely, the signal decomposing and display module, the PFM processing module of single-channel signal and the batch PFM processing module with reference. With the useful message like Table 1, batch signals can be more efficiently processed.

IV. CONCLUSIONS

The theory of PFM is a kind of new attempt and exploration in the signal processing. In this paper, this method is applied to cross-hole tomography signal, which realizes the reasonable separation of signal and noise, and the extraction of useful signal from strong noise environment, and satisfactory results are obtained. The following conclusions are drawn:

(1) The theory and method of PFM can not only be applied to speech signal, but also process engineering signal to realize complete and reasonable separation of a single-channel signal. By studying the variation of the TFSW parameters after separation, all kinds of information and regularity of the signal can also be obtained.

(2) PFM is used for processing signal, and the efficiency of signal processing can be greatly improved by using the characteristics of parameter distribution set of useful signal.

(3) The experiment shows that any superposition can be separated as long as there are differences between the signal and noise. we should fully understand and make use of the signal's inherent characteristics in order to separate the signal quickly and reasonably.

As a kind of new exploration in signal processing, t PFM itself still has many shortcomings, which needs to be improved step by step in the future research.

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