

<Original Paper>

Fault Diagnosis in Gear Using Adaptive Signal Processing and Time-Frequency Analysis

능동 신호 처리 및 시간 주파수 해석을 이용한 기어의 이상 진단

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ABSTRACT

Impulsive sound and vibration signals in gear are often associated with their faults. Thus these impulsive sound and vibration signals can be used as indicators in the diagnosis of gear fault. The early detection of impulsive signal due to gear fault prevents from complete failure in gear. However it is often difficult to make objective measurement of impulsive signals because of background noise signals. In order to ease the detection of impulsive signals embedded in background noise, we enhance the impulsive signals using adaptive signal processing and then analyze them in time and frequency domain analysis.

요 약

기어에서 충격성 진동 및 소음은 치차의 이상과 연관이 있다. 따라서 충격 진동 및 소리는 기어의 이상 진단에 사용 되어 질 수 있다. 또한 이들 충격파를 조기에 정확하게 탐지하여 기어의 이상을 진단하면 완전 파손을 방지 할 수가 있다. 그러나 주변 소음 및 노이즈 신호 때문에 객관적인 충격파의 탐지가 어렵기 때문에, 본 논문은 이러한 숨겨진 충격 신호를 능동 신호 처리 기법을 이용하여 조기에 찾아내고 이것을 시간-주파수 영역에서 해석하였다.

1. Introduction

For a long time vibration signals from a gear have been used for fault detection⁽¹⁾. These faults commonly manifest themselves by radiating impulsive signals due to impacting. However, it is very often difficult to detect these impulsive signals since they are

embedded in background noise, which may consist of harmonics of the rotation speed as well as broadband random process. These background noises hinder the early detection of faults in gear.

To aid fault detection, it is valuable to enhance the impulsive signals by suppressing this background noise prior to further processing. Such pre-processing can be based upon one of several signal processing paradigms. After successful pre-processing the signal has

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an increased Signal to Noise Ratio (SNR), which makes it more amenable to one of a gamut of signal processing tools which can be used to characterise the signal, including Auto-Regressive (AR) modelling⁽²⁾, kurtosis evaluation⁽³⁾, cepstrum analysis⁽⁴⁾, time-frequency analysis⁽⁵⁻⁷⁾, higher order spectra⁽⁸⁾ and higher time frequency analysis⁽⁹⁾. In addition to characterization of faults, the pattern recognition of faults in gearboxes can be classified by further analysis using time-frequency analysis and neuralnetworks⁽¹⁰⁾.

In order to remove fundamental frequency and harmonics of the shaft speed in gear, as pre-processing, time-averaging methods⁽¹¹⁾ have been proposed. This approach has the disadvantage of requiring a triggering signal at the rate of once per revolution. In this paper a two-stage ALE (Adaptive Line Enhancer)⁽¹²⁾ is applied for the enhancement of an impulsive signal embedded in background noise. This method does not require a reference signal and so can be simply applied. The first stage ALE is employed to remove the harmonics of rotational speed, and in the second stage, the impulsive signal is enhanced relative to the broadband random components. However successful application of a two-stage ALE to the enhancement of impulsive signal depends on careful selection of the parameters of the adaptive filter, such as filter length, the step-size and the decorrelation delay. In this paper the LMS algorithm is used for the first stage ALE and the QR-LSL (QR decomposition based on the Least Squared Lattice) algorithm is employed for the second stage. Conditions are presented for the choice of the parameters. For the problem discussed, the output of the adaptive scheme is then passed to the Choi-Williams distribution⁽¹³⁾. Besides simulation examples results are also presented from measured data due to faults in industrial gears.

2. Gear Vibration

The vibration signal in industrial gear depends

on a wide variety of factors such as geometry, stress loading, oil, resonance of gear box, rattle, etc. In the majority of cases, the dominant source is vibration caused by transmission errors (due to inaccurate geometry) introduced during manufacture. It is not possible to avoid manufacturing errors entirely therefore noise and vibration will always be present from this source. Consider gears which mesh under a constant load at constant speed, and assume initially that all the teeth on gear are identical and are equally spaced. The meshing vibration $x(t)$ of one of gears may be represented by a fundamental and harmonics of the tooth meshing frequency f_M :

$$x(t) = \sum_{n=0}^N X_n \cos(2\pi n f_M t + \phi_n) \quad (1)$$

where X_n is magnitude of n harmonic component and f_M is the fundamental frequency of gear meshing. If gear has a manufacturing error so the teeth are non-uniform in profile, then the pitch mesh vibration $x(t)$ includes amplitude and phase modulations and can be rewritten by

$$x(t) = \sum_{n=0}^N X_n (1 + a_n(t)) \cos(2\pi n f_M t + \phi_n + b_n(t)) \quad (2)$$

where $a_n(t)$ and $b_n(t)$ are amplitude and phase modulation functions⁽¹⁴⁾ respectively. They are represented by

$$a_n(t) = \sum_{m=0}^M A_{n,m} \cos(2\pi m f_R t + \phi_{n,m}) \quad (3)$$

$$b_n(t) = \sum_{m=0}^M B_{n,m} \cos(2\pi m f_R t + \theta_{n,m}) \quad (4)$$

where f_R is rotational speed of shaft. These modulation functions may vary from one meshing harmonic to the next, so the subscript n must be incorporated in equations (3 and 4). The derivation so far has concentrated on the case when the amplitude and phase modulation are small under normal condition of gear. However, when a local defect such as a fatigue crack is present in the gear, the modulations extend over a wide frequency range and form an impulsive signal in time domain. The modulations are sideband

frequencies of harmonic components of the gear meshing frequency in frequency domain. In this case, an exact mathematical expression is difficult. However, the vibration signal can be separated as regular signal $x_h(t)$ (carrier frequency) and residual signal $x_r(t)$ (sideband frequencies). The regular signal, which is harmonics of meshing frequency, consist of a sum of pure sine waves and the residual signal, which is amplitude and phase modulated signal, is an impulsive signal due to the fatigue crack. Therefore meshing vibration $s(t)$ of gear can be expressed as

$$s(t) = x_h(t) + x_r(t) + n(t) \quad (5)$$

where $n(t)$ is the broadband noise.

At the early faulty stage of gear, it is may difficult to detect the impulsive signal embedded in the background noise such as the fundamental and harmonic component of shaft rotation speed amplitude modulated signal and frequency modulation signal due to geometric errors. Our object is to enhance the impulsive signal and extract the background noises to detect the impulsive signal.

3. Two-Stage ALE (Adaptive Line Enhancer)

3.1 Principle of the Scheme

In order to enhance the impulsive signals we use a two-stage ALE, the block diagram of

which is shown in Fig. 1. The filter output signal $y_{k,1}$, in the 1st stage ALE, is the correlated signal between input signal x_k and its delayed version $y_{k-\Delta_1}$. Hence Δ_1 must be chosen to cause the impulsive signal to decorrelate with their delayed versions, whilst sinusoidal signals, such as the harmonics of the engine rotation speed, remain correlated under the action of the delay. The error signal $\epsilon_{k,1}$ at the first stage should contain the uncorrelated components, which consists of the impulsive signal and broadband noise. If the delay Δ_1 is too small, or the filter length L_1 is poorly chosen, then the impulsive signals are also attenuated. Later a discussion about how this choice can be made is presented. In the 2nd stage ALE, the error signal from the first stage is used as the input signal. The function of the 2nd stage ALE is to enhance the impulsive signals embedded in broadband noise. This is achieved by exploiting the local structure of the impulsive signals and the short correlation time of the noise. The ALE structure attempts to predict Δ_2 samples into the future. If Δ_2 is large enough then the ALE cannot predict the noise, whilst for the impulsive signals, assuming the filter can track the non-stationary behaviour, then these are predictable. The resulting filter output, $\epsilon_{k,2}$, should contain the impulsive signals at an enhanced SNR.

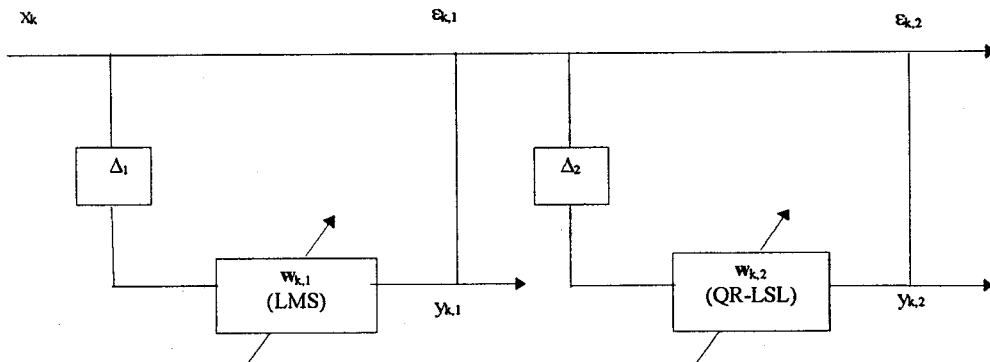


Fig. 1 Two-stage ALE for enhancing the impulse signals in background noise

3.2 Selection of Adaptive Algorithm

The goals of the two stages of this scheme shown in Fig. 1 are very different. The objective of the 1st stage is to remove the narrowband components from the input signal. To achieve this, long filters are required, not only to remove the interactions between each of the terms but also to increase the gain/attenuation of amplitude of the filter at the frequencies of tones. The LMS algorithm⁽¹⁵⁾ is a natural choice for such a problem since it has a low computational cost even for relatively long filter lengths. The objective of the 2nd stage ALE is to reduce the level of the broadband noise. To achieve this we require a filter which can react quickly and we can afford to sacrifice resolution (in the form of reducing the length of the filter) in order to achieve this. For this purpose we employ an exact least squares algorithm specifically the QR_LSL algorithm⁽¹⁶⁾. This is one of the class of exact least squares

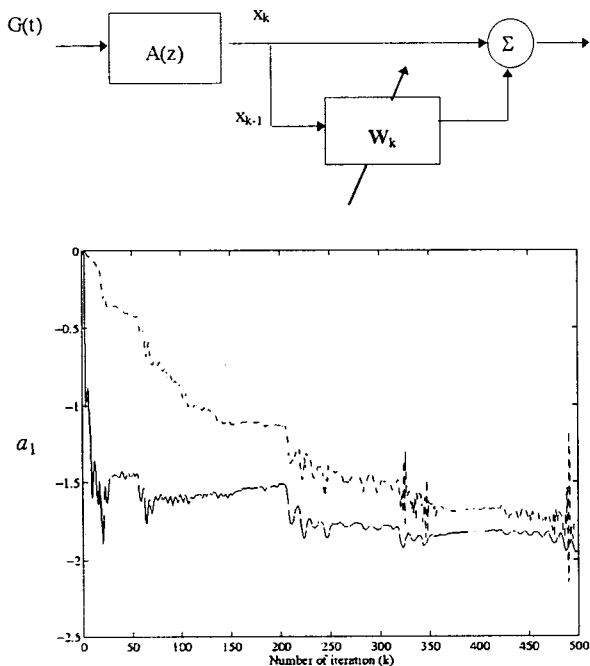


Fig. 2 Evolution of the coefficients of the adaptive filters (a) Model of system identification (b) Prediction of coefficient of the system $A(z)$: LMS: ---- QR_LSL:.....

algorithms which requires only $O(L_1)$ computations and appears to be numerically stable. In order to compare the capability of these adaptive algorithms to track time-varying parameters we constructed test data set by passing white noise through a second order all-pole filter $1/A(z) = 1/(a_0 + a_1z^{-1} + a_2z^{-2})$ with the following parameters:

$$A(z) = 1 - 1.6z^{-1} + 0.95z^{-2} \quad (6)$$

and after 200 time steps, the filter parameters abruptly changed to

$$A(z) = 1 - 1.9z^{-1} + 0.97z^{-2} \quad (7)$$

Thus a_1 is -1.6 before 200 time steps and then -1.9 after 200 time steps. The adaptive filter is used in identification of parameters in system $A(z)$ as shown in Fig. 2(a) in which $G(t)$ is Gaussian random noise. In this test a step size parameter for the LMS algorithm of $\mu = 0.005$ is chosen and a forgetting factor $\lambda = 0.99$ is employed in the QR_LSL algorithm. In both cases a filter length of $L_1 = 2$ is used. In the QR_LSL the prediction errors are initialised to $0.01\sigma^2$. Figure 2(b) shows the evolution of the filters first weight which initially should converge to -1.6, then after 200 samples converge to -1.9. It is evident from this that the QR_LSL converges more rapidly than the LMS algorithm. If we increase the step size in the LMS algorithm, it converges faster but with a greater variance of steady state filter coefficients, or alternatively diverges.

3.3 Conditions for Choosing the Filter Parameters

The periodic nature of the impulsive signal lays open the possibility that the 1st stage of the scheme will identify them with the narrowband components and in doing so attenuate them. To avoid such an eventuality then care over the choice of the parameters Δ_1 and L_1 must be exercised. For the LMS algorithm the update equation for the filter

weight vector \mathbf{w}_k can be written as follows:

$$\mathbf{w}_{k+1} = (\mathbf{I} - 2\mu \mathbf{x}_k \mathbf{x}_k^T) \mathbf{w}_k + 2\mu d_k \mathbf{x}_k \quad (8)$$

where \mathbf{w}_k is a column vector containing the L_1 most recent input samples. From equation (8), with $\mathbf{w}_0 = 0$, we note that if for all k either d_k or \mathbf{x}_k are zero the weight vector remains zero. Thus, if the condition⁽¹²⁾, $L_1 < T_p - 2\Delta_1$, is satisfied the impulse response of the adaptive filter is zero and the impulsive signal is not attenuated by the adaptive filter whilst \mathbf{w}_k is non-zero if the condition $L_1 < T_p - 2\Delta_1$ is met. Details for design of adaptive filter parameters refers to reference (7).

4. Detection of Impulsive signals In synthetic signal

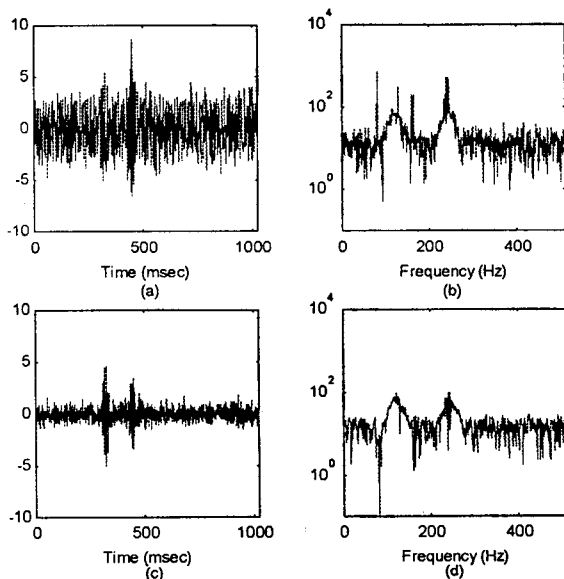


Fig. 3 Enhanced signal with two impulsive signals with centre frequencies 0.1kHz and 0.2 kHz, by Two-stage ALE ($c=0.2$): (a) Input noise (x_k) synthesised as shown in figure 2.4 (b) Fourier Transform of (x_k) (c) Output of ALE (d) Fourier Transform of output of ALE

The simulated signal, as seen in Fig. 3(a), aims to typify the vibration signature of an gear. The detection of the impulsive component is made difficult by the additional noise components. In this section we present the results of applying the two-stage adaptive filter to this simulated data. The parameters of the adaptive schemes are selected in accordance with the design criteria present earlier. In the first adaptive stage the step size μ is selected as $0.2/\lambda_{\max} = 0.1\mu_{\max} \cdot \lambda_{\max}$ is the maximum eigenvalue of autocorrelation matrix of input signal. The delay has to be sufficiently long so as to decorrelated the impulses, this can be achieved by making the delay at least the duration of an impulse, in this case we chose $\Delta_1=200$. The filter length must also satisfy $L_1 < T_p - 2\Delta_1$. In this case we have used $\Delta_1=600$. In the second stage, a suitable choice of filter length L_2 can be made by plotting the MMSE for various filter lengths (17) or using Singular Value Decomposition, in the example here a value of $L_2=8$ was selected. The decorrelation factor Δ_2 should be long enough to decorrelate the noise but not so long as to decorrelate the impulsive signals. For non-stationary signals, the forgetting factor λ is usually selected to satisfy $0.9 < \lambda < 0.99$, we use $\lambda=0.96$.

The result of applying the two-stage line enhancer to the synthetic signals for the gear vibration is depicted in Fig. 3(b). Thus if we assume that when crack of gear due to fatigue occurs the impulsive signals $X_i(t)$ is embedded in background noise as shown in Fig. 3(a) and its Fourier transform is as shown in Fig. 3(b). In here, our aim is to enhance only impulsive signals. The output of the ALE is shown in Fig. 3(c), the impulsive signals are clearly evident. The impulsive signals are also evident in the Fourier domain, Fig. 3(d).

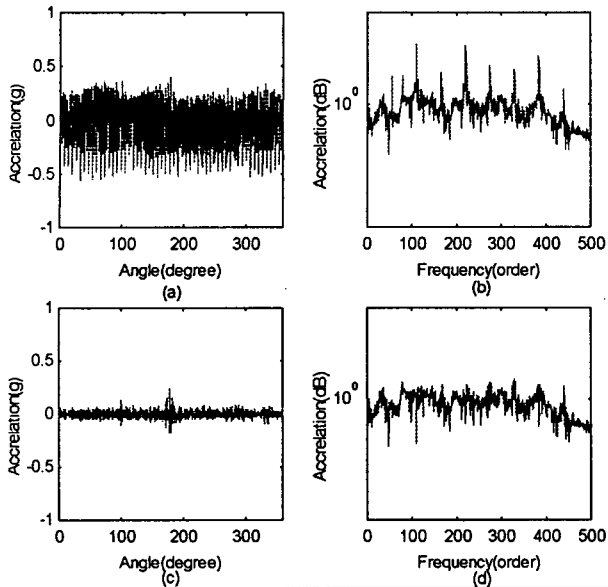


Fig. 4 Using two-stage ALE the detection of impulsive sound due to fault of industrial gear: (a) Vibration signal measured on the industrial gear box. It shows the meshing frequency and its harmonic components (b) FFT of raw signal (c) Impulsive signal after pre-processing using two-stage ALE. (d) FFT of processed signal

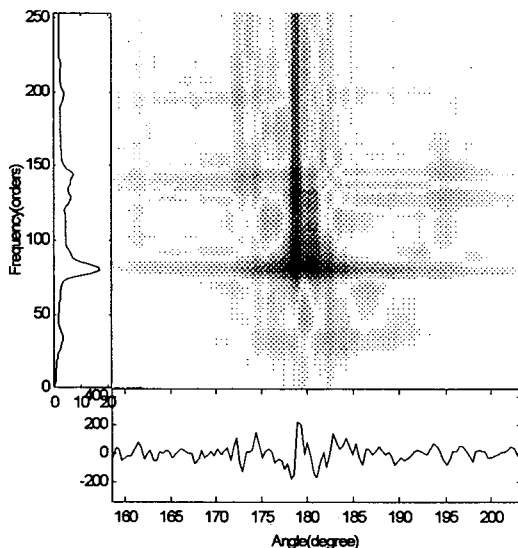


Fig. 5 Choi-Williams distribution for the measured vibration data in fault gear

5. Application

Significant research into the early detection of faults in industrial gears has been undertaken⁽⁹⁾. Figure 4(a) shows the vibration signal measured from a faulty gear⁽¹⁰⁾ whilst Fig. 4(b) shows its Fourier transform. From Fig. 4(b) we can not identify the frequency characteristics of the vibration of fault gear because they are masked by the fundamental and harmonic components of shaft rotation speed. This has 28 teeth and the gear has 56 teeth. Thus the 56th order and 110th orders are the fundamental frequency and the 2nd harmonic of the meshing frequency of the gear wheel. In order to remove the fundamental and harmonics of the tooth meshing frequency without a signal synchronised with drive shaft rotating speed, we employ the two-stage ALE. After processing using two-stage ALE, the impulsive signal is enhanced as shown in Fig. 4(c) and its Fourier transform in Fig. 4(d). Further analysis using Choi-Williams distribution⁽¹³⁾, which is time frequency analysis, gives much better information about the gear fault signal as shown in Fig. 5. The final result implies that the gear fault occurs at 180° shaft angle and at the 82nd orders of shaft rotation speed. This order is the sidebands of the 2nd order meshing frequency (the 110th order) on wheel gear.

6. Conclusion

The impulsive sound and vibrational signals in gear is useful tool for their fault diagnosis since the impulses often occur due to impacting. However detection of these impulsive signals is hindered by high levels of background noise such as fundamental frequency and harmonics of rotating speed and broadband noise. Under circumstances when a triggered signal is unavailable we have proposed a scheme, it called the two-stage ALE. It is

formed by two ALEs and each with different roles. This method does not require any reference signal but use a delayed version (phase shifted) of the original signal as a the reference signal. The output from this algorithm spectralized by time-frequency analysis to yield information about the time when impulsive occurs and simultaneously the frequency content of time signal.

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본 국제학술대회에서는 기조연설 3건과 진동, 제어, 소음분야의 논문 234편 등 총 237편이 발표되었다. 아울러 본 학술대회에서 발표된 논문들이 게재된 Proceeding (I, II)을 회원 여러분께 보급코자 하오니 필요한 회원께서는 학회로 연락 주시기 바랍니다.

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