Proceedings of the 17th World Congress
The International Federation of Automatic Control
Seoul, Korea, July 6-11, 2008

Diagnosis of Engine Misfiring Based on the Adaptive Line Enhancer

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Abstract: In an automotive engine impulsive sounds and vibration are induced by faults or design constraints which degrade the sound quality of the engine. Thus it is important for an NVH engineer to detect and analyse impulsive sound and vibration signals for both fault diagnosis and also for sound quality assessment. However it is often difficult to detect and identify impulsive signals because of interfering signals such as those due to engine firing, harmonics of crankshaft speed and broadband noise components. These interferences hinder the early detection of faults and improvement of sound quality. In order to overcome this difficulty we present a two-stage ALE (Adaptive Line Enhancer) which is capable of enhancing impulsive signals embedded in background noise.

1. INTRODUCTION

For a long time acoustic and vibration signals from a rotating machinery have been used for fault detection (Braun ,1986). These faults commonly manifest themselves by radiating impulsive signals due to irregular 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 rotational machinery. In order to remove fundamental frequency and harmonics of the shaft speed in rotating machinery, time-averaging methods (Braun, and Seth, 1979) 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) (Lee and Cho, 2006)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 time-frequency analyzer. Besides simulation examples results are also presented from measured data due to faults in automotive engines.

2. NOISE AND VIBRATION SIGNAL FOR AUTOMOTIVE ENGINE

2.1 Noise and Vibration Path

In an IC engine (internal combustion engine), typical vibration transfer paths within an internal combustion engine are shown in Fig.1. When the engine speed is constant, if T is the period of one revolution of the crankshaft, then the combustion forces can be expressed in a Fourier series as:

$$p(t) = \sum_{k=-\infty}^{\infty} C_k e^{2\pi jkt/T_p} \quad (1)$$

$$C_k = \frac{1}{T_p}\int_{-T_p/2}^{T_p/2} p(t)e^{-2\pi kt/T}dt, \quad k = 0,\pm1,\pm2 \ldots \quad (2)$$

where T_p is the period of the signal, which in the case of an in-line 4 cylinder IC engine, is 2T. 1/T is called the first order frequency of the crankshaft rotation; combustion forces generate half order components. The spectrum of the combustion force is typically distributed over a wide frequency range, with the energy decaying as a function of frequency, reaching negligible levels above 0.5kHz .

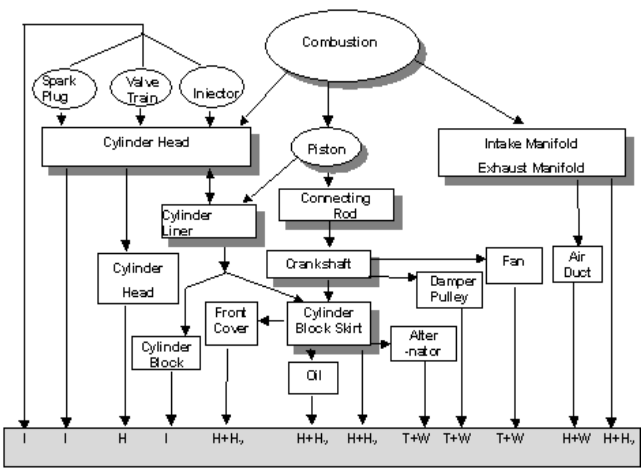


Fig. 1 Transfer path for noise and vibration of automotive engine

In the case of inertial forces the first order and second order components dominate the higher order terms (Priede,1968). From this discussion we can see that there are a great many sources of noise in the IC engine. However, they can largely be classified into five types. First, periodic vibrations with harmonics of the engine rotation speed denoted "H" in Fig.1. Second, vibrations generated by the resonant of structures, which have low damping, such as the vibration from the cylinder head cover, front cover, block skirt, intake/exhaust manifold and oil pan; these we refer to as "H+H_v" in Fig.1. Such signals are characterized by spectra which are band limited with discrete spectral lines. Third, are the high frequency tonal signals; these are generated by rotating components, such as the cooling fan and alternator, labelled "T" in Fig.1. The final group are the impulsive noises denoted by "I" in Fig.1, the sources of which include impacting due to the opening and closing of intake/exhaust valves. From the previous discussion, the signal model we exploit for engine vibration consists of an impulsive signal component, $S_M(t)$, two narrowband signals, $S_H(t)$ and $S_T(t)$, along with broad band noise, n(t). The signals $S_H(t)$ and $S_M(t)$ are assumed to be periodic with period T_p. The total signal x(t) can be modeled as the convolution of signal $S(t) = S_M(t) + S_H(t) + S_T(t)$ and a combination of delta functions separated by T_p plus background noise n(t) as follows:

$$x(t) = \sum_{k=-\infty}^{\infty} S(t) * \delta(t - kT_p) + n(t) \qquad (3)$$

These signals are sampled such that $x_k = x(\Delta k)$, where is the sampling interval..

2.2 Model Signal of Engine Noise

The synthetic signal used in this paper is depicted in Fig.2.

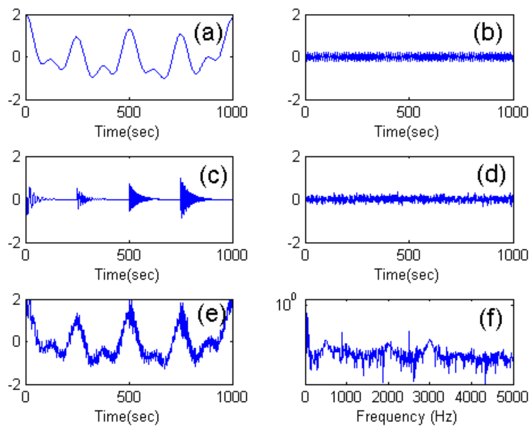


Fig.2 The model signal of internal combustion engine noise ; (a) Fundamental firing frequency and harmonics of crankshaft rotational speed (b) Pure tone noise at 0.9kHz and 2.5kHz (c) Four multiple impulsive sounds at 0.5kHz, 1kHz, 2.0kHz, 3kHz (d) Broadband Gaussian noise with variance σ^2= 0.2 (e) Model_2 signal of engine noise during one period T_p (2T) (f) Fourier Transform of model signal S(t).

The time histories show only one period (T_p). The high amplitude wave, shown in isolation in Fig.2 (a), with period two, is the second order harmonic noise signal $S_H(t)$ of the engine rotation speed. This synthetic signal also includes four impulsive signals $S_M(t)$ with center frequencies 0.5kHz, 1.5kHz, 2.0kHz and 3.0kHz plus two pure tone noise signals $S_T(t)$ (2.5kHz and 0.9kHz) individually shown in Fig.2 (b) and (c) respectively. Fig.2 (d) shows broadband noise signal n(t). Fig.2 (e) and (f) show the complete signals in the time and the frequency domains respectively. Our aim in this study is to extract the impulsive signals $S_M(t)$ from other corrupting noise signals.

3. DDETECTION OF IMPULSIVE SIGNALS IN AN AUTOMOTIVE ENGINE

3.1 Two-Stage ALE (Adaptive Line Enhancer)

In order to enhance the impulsive signals we use a two-stage ALE, the block diagram of which is shown in Fig. 3. 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 $x_{k-\Delta 1}$. 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 $\varepsilon_{k,1}$ 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.

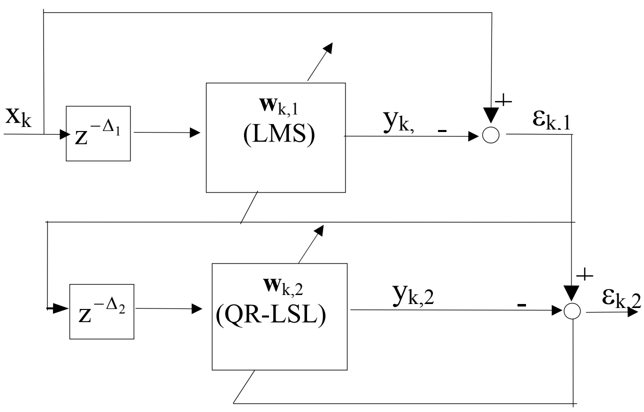


Fig.3 Two-Stage ALE for enhancing the impulse signals in background noise

The resulting filter output, $y_{k,2}$, should contain the impulsive signals at an enhanced SNR. The goals of the two stages of this scheme shown in Fig. 3 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 (Widrow, B. et al., 1975) 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 (Haykin,2001)). This is one of the class of exact least squares algorithms which requires only O(L) 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) with the following parameters:

$$A(z)=1-1.6z^{-1}+0.95z^{-2} \quad (4)$$

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

$$A(z)=1-1.9z-1+0.975z^{-2} \quad (5)$$

The adaptive filter is used in identification of parameters in system A(z) as shown in Fig. 4(a) in which G(t) is Gaussian random noise. In this test a step size parameter for the LMS algorithm of μ=0.005 is chosen and a forgetting factor λ=0.99 is employed in the QR_LSL algorithm. In both cases a filter length of L=2 is used. In the QR_LSL the prediction errors are initialised to $0.01\sigma^2$. Fig. 4(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. 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. To show how this decision can be made consider an impulsive signal, one period of which is shown in the uppermost frame of Fig. 5(a). For the LMS algorithm the update equation for the filter weight vector $\mathbf{w}_k$ can be written as follows:

$$\mathbf{w}_{k+1} = (I - 2\mu \mathbf{x}_k \mathbf{x}_k^T)\mathbf{w}_k + 2\mu d_k \mathbf{x}_k \quad (6)$$

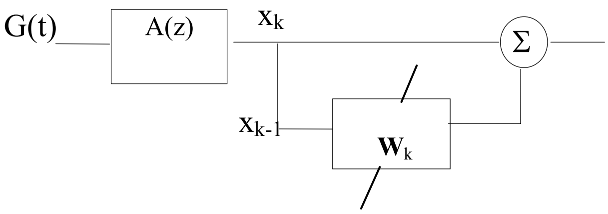
where $\mathbf{x}_k$ is a column vector containing the L most recent input samples. From Eq. (6), with $\mathbf{w}_o$=0, we note that if for all k either d_k or $\mathbf{x}_k$ are zero the weight vector remains zero. For the impulsive signal this condition is met when $L < T_p-2\Delta_1$ since under this assumption

1) when $0 \le k \le \Delta_1$,

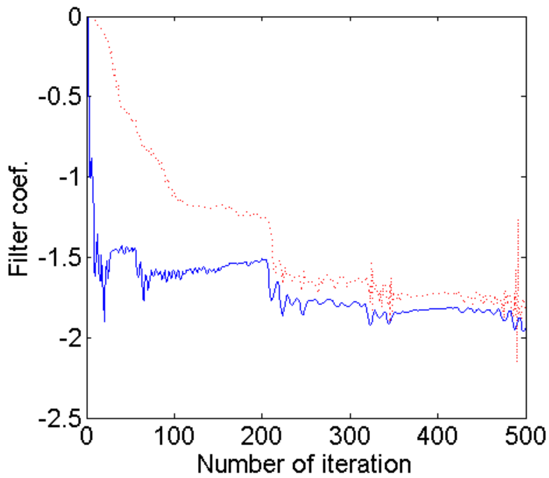
$$\left.\begin{array}{l} d_k \neq 0 \\ \mathbf{x}_k = \mathbf{0} \end{array}\right\} \quad (7)$$

2) and when $\Delta_1 < k < T_p$

$$\left.\begin{array}{l} d_k = 0 \\ \mathbf{x}_k \neq \mathbf{0} \end{array}\right\} \quad (8)$$



(a)



(b)

Fig. 4 Evolution of the coefficients of the adaptive filters LMS:········ QR_LSL: ———

From a statistical view point, if we take the expectation of Eq. (6), we obtain (assuming the independence of $\mathbf{x}_k$ and $\mathbf{w}_k$)

$$E[\mathbf{w}_{k+1}] = (I - 2\mu E[\mathbf{x}_k \mathbf{x}_k^T])E[\mathbf{w}_k] + 2\mu E[d_k \mathbf{x}_k] \quad (9)$$

Eq. (9) illustrates that under the previous assumptions then $E[\mathbf{w}_k]$=0, since $E[d_k\mathbf{x}_k]$ = 0, and $\mathbf{w}_o$ =0. Further assuming a slow adaptation rate also means that the level of misadjustment will be small (Haykin, 2001) implying that $\mathbf{w}_k \approx 0$. The above restriction on the length of the filter is at odds with the desire to increase resolution, and hence attenuation of the unwanted narrowband signals, which comes with a large filter length. Hence one aims to pick L_1 as close to $T_p-2\Delta_1$ without exceeding it (note that T_p is generally not known a priori).

3.2 Detection of Impulsive Signals in Synthetic Signal

The simulated signal, as seen in Fig. 2, aims to typify the acoustic and vibration signature of an IC engine. The detection of the impulsive component is made difficult by the additional noise components. Conventional spectral analysis, based on the FFT, as shown in Fig. 2(f), fails to clearly represent the impulsive 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 μ is selected as $0.2/\lambda_{max} = 0.1\mu_{max}$. 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 Δ_1=200. The filter length must also satisfy $L_1 < T_p - 2\Delta_1$. In this case we have used $L_1 = 600$. In the second stage, a suitable choice of filter length L_2 can be made by plotting the MMSE for various filter lengths (Sayed, 2003) 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. The forgetting factor λ has an equivalence to $1\text{-}2\mu_2$, where μ_2 is the step size in the LMS algorithm. For non-stationary signals λ is usually selected to satisfy $0.9 < \lambda < 0.99$, we use $\lambda = 0.96$. Table 1 details the choice of all the parameters in the adaptive scheme employed in here.

Table 1 The ALE parameter for two-stage ALE

	1st stage ALE	2nd stage ALE
Weight vector length (L)	600	8
Decorrelation factor (Δ)	200	1
Convergence step-size (μ)	0.2 μmax	
Forgetting factor (λ)		0.96
Period length (Tp)	1000	1000
Data length (N)	5000	4200

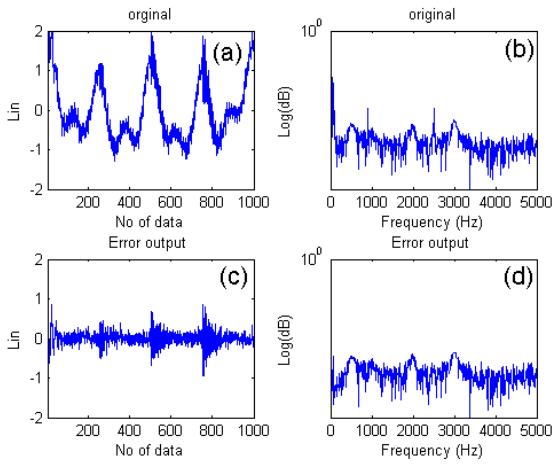


Fig. 5 Enhancement of multi- impulsive signals with centre frequencies 0.5kHz, 1.0kHz, 2.0kHz and 3.0kHz and interference tones at 0.9kHz and 2.5kHz (σ_s / σ_0 =0.0289) by using Two-Stage ALE (c=0.2,λ=0.96) : (a) input noise (x_k) (b) Fourier Transform of x_k (c) Filter output ($y_{k,2}$) in the 2[nd] ALE (d) Fourier Transform of $y_{k,2}$

The result of applying the Two-Stage Line Enhancer to the synthetic signals for the engine noise is depicted in Fig. 5.The impulsive components are difficult to discern in the unprocessed time series, Fig. 5(a), and in its Fourier transform, Fig. 5(b). The output of the two-stage ALE is shown in Fig.s 65(c) and (d), the narrow band components, both at low and mid frequencies, have been significantly attenuated. The impulsive signals are more clearly evident. From Fig. 5(c) and (d), one can identify the synthetic impulsive signals every $T_p/4$ secondss with center frequencies at 0.5kHz, 1.0kHz, 2.0kHz and 3.0kHz.

4. FAULT DIAGNOSIS FOR CAR ENGINE

The vehicle used in this test is an European passenger car with 2.0L in-line 4 cylinder engine. This engine has no fuel injection system. Engine was working at an idle speed around 890rpm, *i.e* the first order is 14.8Hz. The impulsive sounds in this car are induced by loosening a spark plug. This impulsive sound, cannot easily be identified by Fourier analysis because of the background noise. However it can be separated from the background noises by using the Two-Stage ALE. The positions of the microphones used in this test are shown in Fig. 6 ① and ②. The microphones were calibrated using a B&K pistonphone (94dBA). The measured analogue data was converted to digital form at a sampling rate of 10kHz. At site ② in Fig. 6 the impulsive sound from the test car was not clearly audible because of the relatively low SNR.

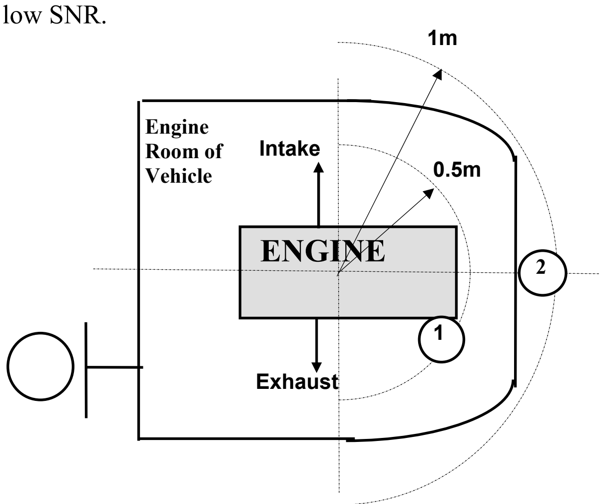


Fig. 6 Installation sites for microphone in the test vehicle.

Fig. 7 (a) and (b) show the time history and its Fourier transformed version for the one cycle of the raw signal measured from the test car. In the first stage ALE the harmonic noises at the engine rotation speed and pure tone noises are nearly cancelled and the impulsive sound is enhanced by the second stage ALE, as shown in Fig. 7 (c). From the Fourier representation of enhanced signal as shown in Fig. 7(d) it can be seen that the impulsive sound of test car engine has a peak at slightly less than 1.5kHz.

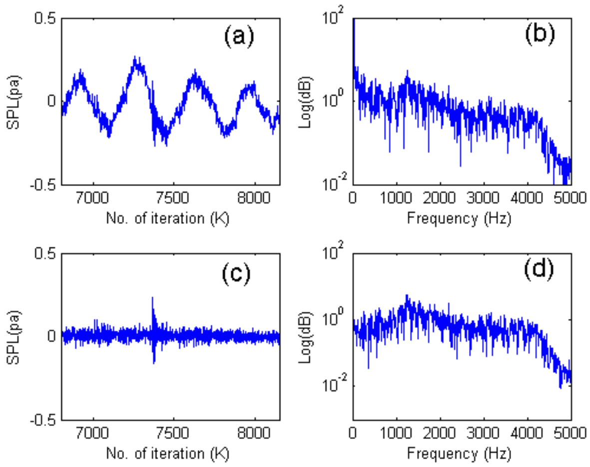


Fig. 7 Impulsive noise enhancement using Two-Stage ALE at point ① (:μ=0.00822 and λ = 0.99, L_1=600 L_2=8 Δ_1=300 Δ_2= 3); (a) Input noise (x_k) (b) Power Spectral Density of x_k (c) filter output (y_k^2) in 2nd ALE (d) Power Spectral Density of y_k^2

5. CONCLUSION

The impulsive sound signals in automotive engine is useful tool for their fault diagnosis since the impulses often occur due to irregular 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 Adaptive Line Enhancer. 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.

REFERENCES

Braun, S. (1986) *Mechanical Signature Analysis-Theory and Applications*, Academic Press New York

Braun, S. G. and Seth, B. B. (1979) On the extraction and filtering of signals acquired from rotating machines. *Journal of Sound and Vibration.*, **65(1)**, 37-50.

Lee, S. K and Cho B. O. (2006) Identification of Automotive Engine Based On Detection of Impact Vibration Using an Adaptive Filter. *Key Engineering Materials*, **306**, 229-234.

Priede, T. (1968) Origin of automotive engine and vehicle noise, ISVR, *University of Southampton, Memorandum*, No.263.

Widrow, B. et al. (1975) Adaptive noise cancelling: principles and application, *Proc. IEEE*, 63, 1692-1716.

Haykin, S. (2001) Adaptive filter theory, 4th Edition, Prantice Hall.

Sayed, A. H. (2003) Fundamentals of Adaptive Filtering. Wiley-IEEE Press.