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New Frequency-Domain Method for Shock Vibration Control*

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Abstract: Through analyzing the characteristic of shock vibration control and the limitation of former algorithm, a method of shock vibration control based on adaptive inverse control is raised to improve the accuracy and robust of the system. It can adjust the parameter of the controller according to the different load and can solve the problems of low resolution at low frequencies and overflow. In order to decrease the computation complexity and shorten the convergence time of the filter-X LMS algorithm, a fast block filter-X LMS algorithm is put forward. The practical result shows the algorithm performance is better than former algorithm.

Key words: adaptive inverse control; adaptive filtering; filter-X LMS algorithm; shock vibration control

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Shock testing is one of environment vibration testing. It is to assess the reliability and the adaptability of the specimen after the occurrence shock condition which is expected to encounter during operation or transport.

Shock vibration usually exhibits by the shock pulse, shock duration and peak accelerations of the pulse and the typical shock pulse contain half-sine, square, triangular, trapezoidal and etc. in form. Shock vibration control system which is based on vibration object (including specimen, electromagnetic vibration actuator, PA, acceleration sensor, etc.) properties, to generate control signals through the algorithms according to the setting parameters of shock pulse, encourages the electromagnetic actuator to get the reproduction of set-shaped shock pulse. Because of the big differences in the structure and weight of the specimen, the system model changed significantly, it is difficult to realize the shock pulse accurately. Based on this reason, the control algorithms are required to be adaptive to the load-varying system.

Traditional control method of Shock testing adopted the Frequency Domain Self-tuning Algorithm (FDSA). It begins by driving the controller with typical shock pulse that set previously. The difference between response spectrum and driving spectrum which computed in real time is feedback to revise the driving spectrum of the shock pulse, and then the time domain signal transformed from the driving spectrum is used to control the actuator. By continuous iterations, the response vibration waveform will converge to the reference shock waveform finally.

The frequency domain self-tuning algorithm serves well in random vibration control system, but it has obvious shortcomings in the shock vibration control. Figure 1 is the spectrum of a half sine with a pulse width $\tau = 2\text{ms}$, it shows that:

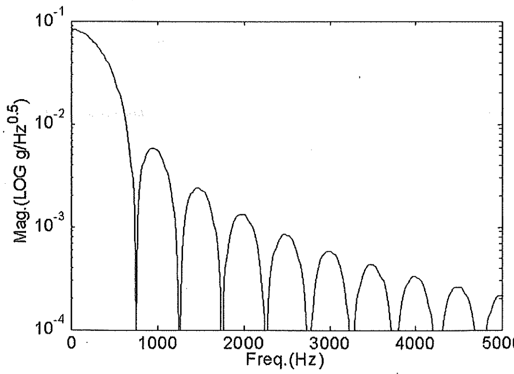

Fig. 1 Spectrum of a half sine

1) The energy of shocking signal mainly concentrates in the low-frequency band, and the frequency domain self-tuning algorithm is based on the FFT algo-

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rithm, so there is a contradiction between the relatively lower resolution of Fourier transform of a signal in low-frequency band and the general concentration of energy of a shocking signal in low-frequency band;

2) At the frequency of $(0.5 + n)/\tau (n > 0)$, Spectrum value tends to be zero, this brings to the FDSA a mathematical problem: the computing result of driving spectrum could be overflowed or very easy to be zeros, resulting in poor control accuracy.

Through analyzing the characteristic of shock vibration control and the limitation of FDSA, a method of shock vibration control based on adaptive inverse control is raised to improve the accuracy and robust of the system. It can adjust the parameter of the controller according to the different load and it can solve the problems about low resolution at low frequencies and overflow. In order to decrease the computational complexity and shorten convergence time of the filter-X LMS algorithm, a fast block filter-X LMS algorithm is put forward.

1 Adaptive Inverse Control of Shock Vibration Control

Adaptive control is a novel approach to the design of control system. The ideas that have been evolved suggest open-loop control of system dynamics by using a series controller whose transfer function characteristics are inverse to those of the object to be controlled. The controller is adaptive, and adjusts itself to optimize the overall dynamic response of the object and its controller. The system dynamic performance can control separately with the disturbance attenuating. Feedback is only used in the adaptive process.

$M(z)$ in Fig. 2 is the reference model which reaches the desired performance of the control system. The controller $C(z)$ can be seen as optimal filter with tap weights $w(n)$. The taps $w(n)$ is adjusted by the adaptive algorithms in order to minimize the error $e(n)$. After the inverse controller $C(z)$ converges the dynamic characteristics of it cascaded with the object transfer function $P(z)$ will equal to the reference model $M(z)$, that is:

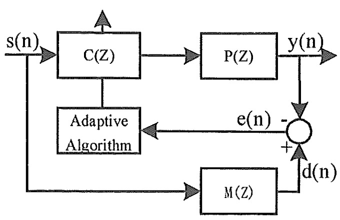


Fig. 2 Schematic of adaptive inverse control

$$Y(z)/S(z) = C(z)P(z) = M(z) \qquad (1)$$

Although Shannon-Bode method in evaluating optimal adaptive filter is for IIR filter, FIR filter has no poles and there are no stability problems, then it could be proved, that the pulse response of optimal FIR model for the IIR object can match the characteristics of the object in the duration of its response, because once the limited duration of model impulse response covers one point of the object impulse response, then behind this point the object impulse response would attenuate to a negligible level. So in general the adaptive inverse controller $C(z)$ can be realized by the FIR model.

In inverse object modeling, when the target is the non-minimum phase system, some of its poles which are outside the unit circle cause the inverse controller instability. To resolve this problem, the Wiener optimal solution is obtained by adapting the inverse model to the delayed input signal. This will bring some error, but it can achieve adequate control accuracy as the system.

1.1 Filter-X LMS Algorithm

The block diagram of filter-X LMS adaptive inverse control algorithm is shown in Fig. 3, in which $P(z)$, $\hat{P}(z)$ and $\hat{C}(z)$ are the vibration system, the transfer function of the vibration system and the inverse controller model respectively. Tests show that the electromagnetic vibration systems are non-minimum phase systems, so the inverse controller model $\hat{C}(z)$ feed by the time delay Δ of the input signal.

The filter-X LMS algorithms have a good characteristic that the error of the system model $\hat{P}(z)$ has no effect on the Wiener solution of the inverse controller model $\hat{C}(z)$, that means the system is insensitive to the error of controller model. Even if the system model has errors, the system can satisfy Eq. 1, the only purpose of $\hat{P}(z)$ which get similar to the $P(z)$ is to ensure the $\hat{C}(z)$ to be converged.

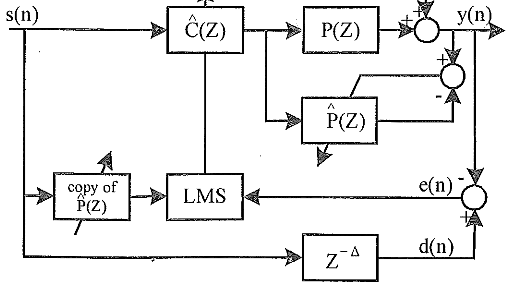


Fig. 3 Filter-X LMS algorithm

However the convergence performances of LMS algorithms are not satisfactory. First, it has a slow convergence rate because the shocking signal is a colored signal, and the convergence rate is strongly dependent on the correlation of the shocking signal; second, it is so complex in computation to process in real time. So a fast block filter-X LMS algorithm is put forward to satisfy the shocking vibration system performances.

1.2 Fast Block Filter-X LMS Algorithm

In the Filter-X LMS algorithm, when the impact response has a long duration, the computation complexity will increase significantly, and this brings difficulties to the shock vibration control in real-time. The performance of the filter-X LMS algorithm is strongly dependent on the auto-correlation of the input signal. So it has a slow convergence rate for the application when the eigenvalue spread of the correlation matrix of the input signal dispersed considerably. In order to overcome the above drawbacks, a fast block Filter-X LMS adaptive inverse control algorithm is raised which is shown in Fig. 4. It uses block computing and the FFT algorithm to improve the real-time character: Fist it reduces the relativity by transforming the input signal to frequency domain by discrete Fourier transform, and then normalizes the signal' s energy to force the eigenvalue of auto-correlation matrix concentrated in the vicinity of 1. All of the improvements accelerate the convergence rate of the adaptive filter.

As shown in Fig. 4, the inverse controller is structured by the FIR filter with tap weights $w(n)$ for $n=1, \cdots, M$. In the new shock vibration control algorithm the data to be processed sectioned into block by the number M. S_{in} and S_{out} are the linear convolution with 50% overlap-save method. S_{in} is $2M$ length of data blocks, in which the former M data are preserved from the last data block and the data After M is current data block. However $S_{\text{in}}^{'}$ in has a slightly different that the former M data are zeros, caused by removing of the former M data of S_{out}. The Fourier transform of $w(n)$ added by M zeros is:

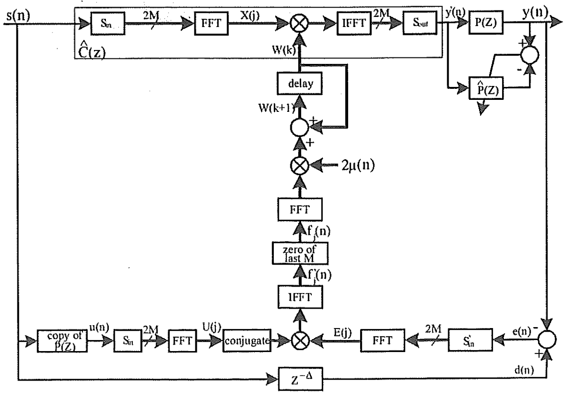


Fig. 4　Fast block filter-X LMS algorithm

$$W^T(k) = \mathrm{FFT}[w(0), \ldots, w(M-1), 0, \ldots, 0] \tag{2}$$

Supposing the jth block data is processed currently, $U(k)$ is the diagonal matrix of the Fourier transform of $u(n)$ composed by $(j-1)$ th and jth block data. It is written as follows:

$$U(j) = \mathrm{diag}\{\mathrm{FFT}[u(jM-M), \ldots, u(jM-1), u(jM), \ldots, u(jM+M-1)]\} \tag{3}$$

The system error signal of jth block data is produced by comparing the reference model output against the object output, its Fourier transform is

$$E(j) = \mathrm{FFT}[0, \ldots, 0, e(jM), \ldots, e(jM+M-1)]^T \tag{4}$$

Then the correlation coefficients of $u(n)$ and $e(n)$ for the jth block data are the first M elements of

$$\varphi_j^{'}(n) = \mathrm{IFFT}[U^*(j)E(j)] \tag{5}$$

Finally, the equation to update tap-weights vector of the inverse controller in frequency domain is

$$W(j+1) = W(j) + 2\mu(j) \, FFT[\varphi_j(0), \ldots, \varphi_j(M-1), 0, \ldots, 0]^T \tag{6}$$

Where

$$\mu(j) = \mu \times \mathrm{diag}\{U_0(j), \ldots, U_{2M-1}(j)\} \tag{7}$$

Where $U_i(j)$ is the ith diagonal element of the matrix $U(j)$, μ is the step-size parameter of the tap-weights vector, which satisfies:

$$0 < \mu < 1 \tag{8}$$

In Fig. 4, the linear convolution of the inverse controller tap weights $w(n)$ and the input signal $s(n)$ applies 50% overlap-save method by fast Fourier transforming algorithm. When the inverse controller is converged it yields

$$\hat{C}(z)P(z) \approx M(z) = z^{-\Delta} \tag{9}$$

That is

$$|\hat{C}(z)P(z)| \approx 1 \tag{10}$$

This realized $y(n)$ is the Δ - delay one of the $s(n)$.

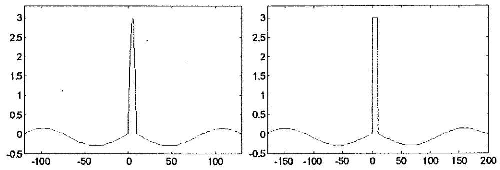
In Eq. (6) the tap weights of the inverse controller update once every a block data time, and remain unchanged in a block data, so it can see that the block LMS algorithm uses a more accurate estimate of the gradient vector because the time averaging, with the estimation accuracy increasing as the block size M is increased.

It can be seen that the fast block filter-X LMS algorithm accelerates the convergence rate of the adaptive inverse controller with the following method: first, contrasting to the standard method which computing in time domain, it computes the convolution and correlation operations in frequency domain by orthogonal transform, so the correlation of the input signal of adaptive algorithm is reduced; second, by using the block mechanism and the FFT algorithm it greatly reduced an operation complexity.

2 Practical Results

A multi-processor based vibration control system, powered by the digital signal processor TMS320C6713 and industrial PC, has been developed to realize the FDSA and the fast block filter-X LMS adaptive inverse control algorithm. Both half-sinusoidal and rectangular shock pulse are selected to check the performances of the two algorithms.

In the shocking vibration control system, in order to truly reflect the shock pulse shape, the system sampling rate is adjusted by the shocking-pulse width to ensure 100 sampling points in the effective shocking duration, so to process different shocking pulse the amount of sampled data is almost same. Here the width and the acceleration of the shock pulse are 10 ms and 3 g respectively. In practical, in order to ensure the consecutive of the acceleration, velocity and displacement, it is requested all of them to be zeros at the beginning and the end. This will maximally power the dynamic performance of the electromagnetic vibration actuator, so it needs to compensate the pulse. The compensated shocking pulse is shown in Figure 5.



a. half-sine b. rectangular

Fig. 5 Compensated shock pulse

Through analyzing the system model $\hat{P}(z)$, the vibration system $P(z)$ is non-minimum phase system. To ensure the inverse controller $\hat{C}(z)$ stability, the time delay Δ of the reference model can be selected by the rule, that the impulse response of the inverse controller has attenuated negligible levels outside the $w(n)$ coverage. In the inverse controller the M, length of the FIR filter is 1024 and the reference model delay Δ is 800. Fig. 6 is the system response in frequency domain by applying random signal after the inverse controller has been converged and Fig. 7 is the one that achieved by HP35670A using sweep-frequency method, it can be seen that the effect of the adaptive convergence is excellent.

To compare the performance of the fast block filter-X LMS adaptive inverse control algorithm with the conventional FDSA, here the half-sine and rectangular shock pulse with the width and the acceleration are

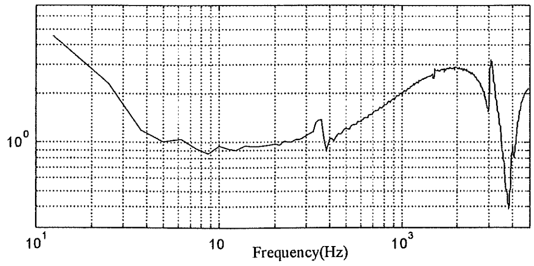


Fig. 6 Frequency response of inverse controller

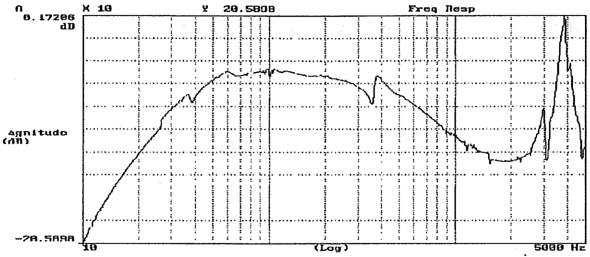


Fig. 7 Frequency response of electrodynamic actuator

10 ms and 3 g respectively applied to the vibration system, the vibration-control result are shown in Fig. 8 and Fig. 9. The results illustrate that the performance of fast block filter-X LMS adaptive inverse control algorithm at the low-frequency domain is greatly improved and the smoothness property of response shock wave is enhanced markedly.

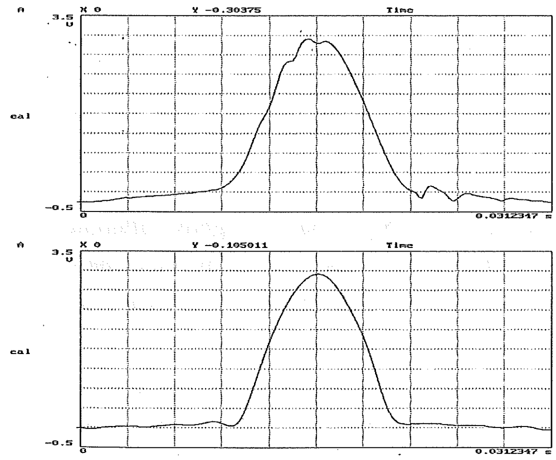


Fig. 8 Comparision of practical result of half-sine pulse

3 Conclusions

The adaptive inverse vibration control algorithm is a novel method to design the shocking vibration controller. To reduce the computational complexity and improve the convergence rate in the conventional filter-X LMS algorithm, a new shock vibration control method based on the fast block filter-X LMS adaptive inverse control algorithm is raised. It can adjust the parameter

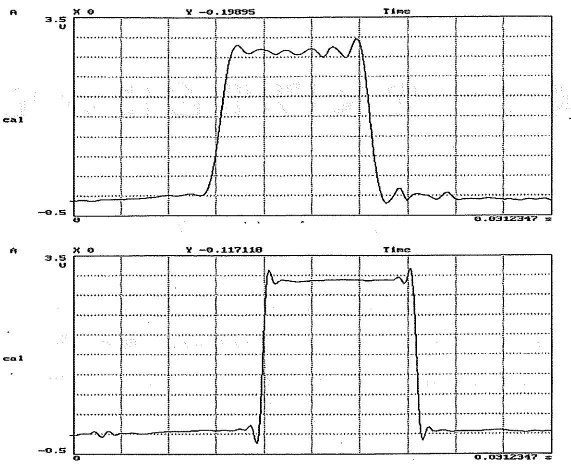


Fig. 9　Comparision of practical result of rectangular pulse

of the controller according to the different load and the convergence process weakly depends on the auto-correlation matrix of input signal. The practical result shows the algorithm gets excellent performance in real-time behavior and convergence rate. Comparing to the traditional algorithm, it solves the problems of overflow and low resolution at low frequencies, and greatly improves the shocking accuracy.

References:

[1] WIDROW B, WALACH E. Adaptive inverse control[M]. Xi'an: Publishing House of Xi'an Jiaotong University, 2005.

[2] PAGE M. Signal processing for vibration control[A]. IEE Colloquium on DSP in Instrumentation, London, 1992.

[3] LIU Aiping, ZOU Lihua, DA Suliang. Study on Wavelet-pack de-noising treatment for vibration control[J]. Journal of Vibration and shock, 2008, 27(2): 95 - 99.

[4] WIDROW B, PLETT G. L, FERREIRA E, et al. Adaptive inverse control based on nonlinear adaptive filtering. Proceedings of 5th IFAC workshop Algorithms arthitectures for real-time control. AARTC'98, Cancun, Mexico. 1998: 247 - 252.

[5] PLETT G L. Adaptive inverse control of unknown stable SISO and MIMO linear systems. Int. J. Adaptive control. Signal Processing. 2002, 16(4): 243 - 272.

[6] SHYNK J. Frequency-domain and multirate adaptive filtering[J]. IEEE Transactions on Signal Processing, 1992, 9(1): 14 - 37.

[7] YU Ping, HAN Pu, ZE Dong, et al. Volterra Basic Function Network Based Adaptive Controller, 2004 International Conference on Machine Learning and Cybernetics, Shanghai, China 2004.

[8] PLETT G L. Adaptive inverse control of linear and nonlinear systems using dynamic neural networks[J]. IEEE transactions on neural networks, 2003, 14(2): 360 - 376.

[9] HAYKIN S. Adaptive filter theory[M]. Forth Edition, Beijing: Publishing House of Electronics Industry, 2002.

[10] YU Huamin, ZHU Haichao, SHI Yin. Active noise control FXLMS Algorithm based on Non-minimum phase adaptive inverse control[J]. Journal of vibration Engineering. 2003, 16(3): 331 - 336.

[11] FARHANG B B, CHAN K S. Analysis of the Frequency-Domain Block LMS Algorithm[J]. IEEE Transactions on Signal Processing, 2000, 48(8): 2332 - 2342.

基于自适应逆控制的冲击振动控制新方法

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摘　要：在分析冲击振动控制特性以及已有控制算法的缺陷基础上，提出了一种新的冲击振动自适应逆控制方法，该方法可以根据负载特性的不同自适应地调整逆控制器参数，提高系统控制精度及鲁棒性，完全解决了传统频域控制方法中低频分辨率低、易产生溢出的问题。同时，为了减少算法计算复杂度及缩短算法的收敛时间，提出了一种快速 X 滤波 LMS 算法，运用批处理技术，使运算效率得到提高。最后对算法进行了实验验证，结果表明文中提出的控制方法精度较高、算法计算量小，明显优于已有的控制方法。

关键词：自适应逆控制；自适应滤波；X 滤波最小均方算法；冲击振动控制

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