SIMULATION AND EXPERIMENTAL RESEARCH ON UNBALANCE VIBRATION CONTROL OF AMB SYSTEM

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ABSTRACT

Two kinds of processing method for the unbalance vibration control are simulated and experimented in a AMBs-spindle system. One is called Force Free Control (FFC), and the other is Open Loop Feedforward Control (OLFC), which is called as Force Controlled Control (FCC) also in this paper. The simulation is done with MATLAB. Experiments were done on above methods based on the ADSP2181 DSP system. With FFC method, the maximum vibration amplitude at 780Hz is less than 35 (μm), and with the FCC method the vibration at 500Hz is reduced from 20(μm) to 1(μm).

INTRODUCTION

The synchronous vibration caused by the unbalance of a rotor suspended by AMBs cannot be avoided eventually. In most application, the speed of the rotor is very high, the synchronous vibration may limit the performance. Sometimes, the vibration of a rotor increases due to the current output of controller saturates near some special speed and causes the amplitude of the stiff critical frequency (such as 200~300Hz in our experiment set) to increases very fast, and causes the stability of the rotor to decay.

By FFC, a compensated signal which is generated as the same amplitude and phase of the input displacement signal can be used to subtract the synchronous component of the vibration signal from input of controller[1], and no control current is inputted to the coils. There exists two realizing ways to generate such compensate signal: adaptive LMS filter and direct Fourier coefficients calculate. Experiment proved that only the synchronous part is affected exactly. However the steady error of compensation of a LMS algorithm can be influenced by the step size. Comparing to the LMS algorithm, there are the close measurement results with the direct Fourier coefficient calculation. Analyses and experiment also show that the Fourier method will generate uncontinuous compensating signal.

The further discussion focuses on reducing the vibration amplitude of the rotor displacement at the operating speed by FCC. To reduce the vibration amplitude usually requires the full information of system parameter to derive the parameter for convergent control. There are many ways to obtain the system parameter, but, because the Active Magnetic Bearings system parameters may shift seriously with the operating point, the robustness of the convergent control is very important. A nonlinear convergent control is proposed by Nonami[2], which does not require the full information of the transfer function, and the algorithm uses a try-adjust method to find the correct direction of actual transfer function. However, the convergence cannot be proved theoretically, and the vibration may enhance temporarily at the procession of finding the right direction. Simulation for this method turned out that if the transfer function is not estimated exactly, the convergence speed is very slow at some cases. The iteration procession tends to change direction quickly, and cannot find a good direction. Sometimes, the system tends to oscillation.

A modified method in this paper is proposed to improve the convergent rate. The idea is to make the matrix in the Nonami's nonlinear adaptive law equation full of four parameters, and then adjust the weight to another two parameters, if oscillation happened.

All methods mentioned above were tested on a AMBs-spindle system. With FFC method, the maximum vibration amplitude at 780Hz is less than 35(μm), and in the FCC method the vibration at 500Hz is reduced from 20(μm) to 1(μm).

FORCE FREE CONTROL

The basic idea of force free control is to generate a compensate signal with the same amplitude and phase of the input displacement signal and to send it to controller, the controller will not response to the synchronous disturbance. The synthesizer has a structure as shown in Fig. 1.

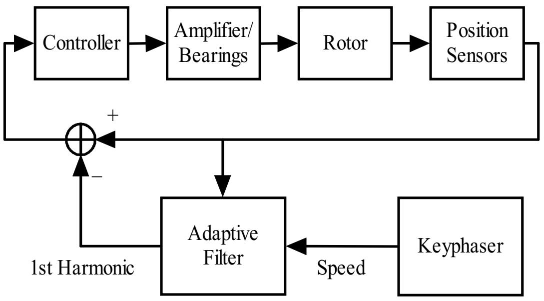


FIGURE 1: Force Free Control

1. LMS Algorithm

LMS algorithm modifies the gain parameters every sample time by using a momentary gradient method, and has the advantage of simple structure of algorithm, The synthesizer has a structure as shown in Fig.2.

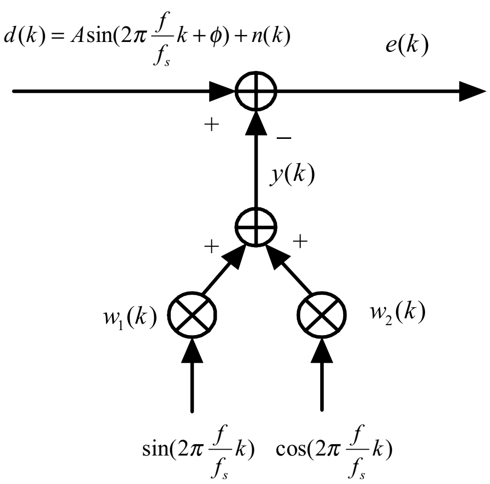


FIGURE 2: Input Cancel Synthesizer

The object function is $J(\underline{w})=e^2(k)$.and

$$e(k) = d(k) - \underline{w}'(k)\underline{x}(k) \tag{1}$$

$$\underline{w}(k) = \begin{bmatrix} w_1(k) \\ w_2(k) \end{bmatrix}$$

$$\underline{x}(k) = \begin{bmatrix} \sin(2\pi \frac{f}{f_s} k) \\ \cos(2\pi \frac{f}{f_s} k) \end{bmatrix}$$

where, $e(k)$: error after compensation;
$y(k)$: compensating signal;
$d(k)$: vibration signal;
$w_1(k)$, $w_2(k)$: Fourier coefficients;
f: rotating speed in Hz;.
f_s: sample frequency;
k: sample sequence,
w': the transpose of w.
Modifying the parameter by using temporary gradient,

$$\underline{w}(k+1) = \underline{w}(k) - c\nabla_{\underline{w}}\{J(\underline{w})\} = \underline{w}(k) + 2ce(k)\underline{x}(k) \tag{2}$$

Where the c is a factor for adjusting. The equation (1) and (2) is the LMS method.

2. Fourier Coefficient

First, in order to get the Fourier coefficient, let the $w_1(k)$ $w_2(k)$ equal the coefficient respectively by using correlation method. This method has the advantage of reject non-synchronous sine waves as well as noise.

$$\begin{aligned} E\{d(k)\sin(2\pi \frac{f}{f_s} k)\} &= \frac{1}{2} A\cos\phi \\ E\{d(k)\cos(2\pi \frac{f}{f_s} k)\} &= \frac{1}{2} A\sin\phi \end{aligned} \tag{3}$$

In order to reduce the calculation for a fixed point DSP system, the coefficients can be obtained from equation (4) based on the facts that 1st harmonic is the main part of signal.

$$\begin{aligned} E\{d(k)\text{sign}[\sin(2\pi \frac{f}{f_s} k)]\} &= \frac{2}{\pi} A\cos\phi \\ E\{d(k)\text{sign}[\cos(2\pi \frac{f}{f_s} k)]\} &= \frac{2}{\pi} A\sin\phi \end{aligned} \tag{4}$$

Choosing $J(\underline{w})=E\{e^2(k)\}$, then the Mean Square Error cost function.

$$\begin{aligned} J(\underline{w}) &= E\{e^2(k)\} \\ &= E\{d^2(k)\} + \underline{w}'E\{\underline{x}(k)\underline{x}'(k)\}\underline{w} - 2\underline{w}'E\{d(k)\underline{x}(k)\} \end{aligned} \tag{5}$$

That is the steepest descent method, and then,

$$R = E\{\underline{x}(k)\underline{x}'(k)\} = diag(\frac{1}{2}, \frac{1}{2});$$

$$\underline{p} = E\{d(k)\underline{x}(k)\} = \frac{A}{2}\begin{bmatrix} \cos\phi \\ \sin\phi \end{bmatrix}$$

$$\begin{aligned} \underline{w}(k+1) &= \underline{w}(k) - c\nabla_{\underline{w}}\{J(\underline{w})\} \\ &= (I - 2cR)\underline{w}(k) + 2c\underline{p} \end{aligned} \tag{6}$$

This method has a low pass filter effect to the error of calculating the Fourier coefficient. But its adaptation speed is slow especially with a small value of c. When c in the LMS and the steepest method is small, the adaptive process is not fast and not sensitive to the change of the signal amplitude. And when c is large the process is sensitive to noise and tends to unstable. Both the two methods converge are not guaranteed for any step size of c, and upper limit of c is related with the eigenvalue of R. The LMS requires a small c for it uses a momentary gradient, and the steepest can use a large value of c.

FORCE CONTROLLED CONTROL

Fig.3 is the Force Controlled Control scheme block diagram. For simplifying the system, only synchronous loop, (a quasi open loop) has been taken into account, which is shown in Fig.4.

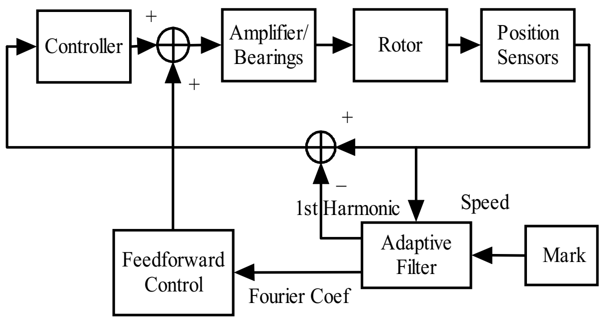


FIGURE 3: Force Controlled Control

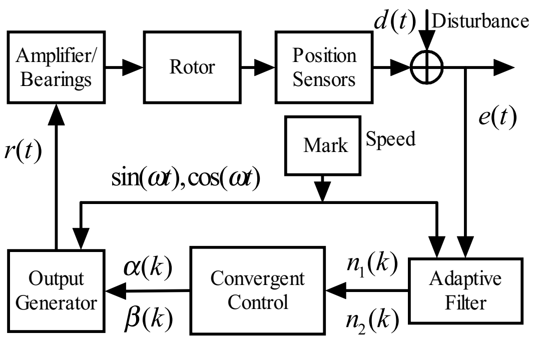


FIGURE 4: System Rejecting Synchronous Disturbance

Assuming that the unbalance disturbance is sinusoidal and describing as:

$$d(t) = \alpha_d \sin(\omega t) + \beta_d \cos(\omega t) \qquad (7)$$

The transfer function of the plant at rotor speed is $G(j\omega)=G_\omega e^{j\theta}$, the plant includes amplifiers/bearings, rotor and sensors. Constructing the synchronous compensation signal of the following form

$$r(t) = \alpha(t)\sin(\omega t) + \beta(t)\cos(\omega t) \qquad (8)$$

Then the steady output with disturbance is expressed as

$$\begin{aligned} e(t) = {} & G_\omega(\alpha(t)\sin(\omega t+\theta) + \beta(t)\cos(\omega t+\theta)) \\ & + \alpha_d \sin(\omega t) + \beta_d \cos(\omega t) \end{aligned} \qquad (9)$$

Presuming that the correction signal does not change rapidly or wait enough time till the output reaches the steady state. So the correlation coefficient output can be written as the follows,

$$\begin{aligned} \begin{bmatrix} n_1(k) \\ n_2(k) \end{bmatrix} = {} & 0.5G_\omega \begin{bmatrix} \cos(\theta) & -\sin(\theta) \\ \sin(\theta) & \cos(\theta) \end{bmatrix} \begin{bmatrix} \alpha(k) \\ \beta(k) \end{bmatrix} \\ & + 0.5 \begin{bmatrix} \alpha_d \\ \beta_d \end{bmatrix} \end{aligned} \qquad (10)$$

Paper [2] gives a nonlinear adaptive law which independents of the θ

$$\begin{aligned} \begin{bmatrix} \alpha(k+1) \\ \beta(k+1) \end{bmatrix} = {} & \begin{bmatrix} \alpha(k) \\ \beta(k) \end{bmatrix} - \begin{bmatrix} \mu_{11}(k) & 0 \\ 0 & \mu_{22}(k) \end{bmatrix} \\ & \begin{bmatrix} n_1(k) \\ n_2(k) \end{bmatrix} \end{aligned} \qquad (11)$$

The step size is determined by the following equation.

$$\begin{bmatrix} \mu_{11}(k+1) \\ \mu_{22}(k+1) \end{bmatrix} = \begin{bmatrix} \mu_{11}(k)\mathrm{sign}(n_1^2(k) - n_1^2(k+1)) \\ \mu_{22}(k)\mathrm{sign}(n_2^2(k) - n_2^2(k+1)) \end{bmatrix} \qquad (12)$$

If the differential is regard as approximate derivative, then:

$$\begin{aligned} \begin{bmatrix} \dot{n}_1(k) \\ n_2(k) \end{bmatrix} & = +0.5G_\omega \begin{bmatrix} \cos(\theta) & -\sin(\theta) \\ \sin(\theta) & \cos(\theta) \end{bmatrix} \begin{bmatrix} \dot{\alpha}(k) \\ \beta(k) \end{bmatrix} \\ & = -0.5G_\omega \begin{bmatrix} \cos(\theta) & -\sin(\theta) \\ \sin(\theta) & \cos(\theta) \end{bmatrix} \begin{bmatrix} \mu_{11}(k) & 0 \\ 0 & \mu_{22}(k) \end{bmatrix} \\ & \quad \begin{bmatrix} n_1(k) \\ n_2(k) \end{bmatrix} \end{aligned} \qquad (13)$$

The idea of this method is to change the sign of coefficients to find the right direction of the output at which amplitude decreases. The convergence of the algorithm requires the knowledge of upper bound of the system gain. Simulation turned out that if the gain is not known exactly the convergence speed is very slow at some cases. If a big margin about the gain estimate happens to appear, a small step size has to be used. For example, giving one tenth of gain $|\mu_{ii}(k)|=0.1/G_\omega$. From the simulation results which shown in Fig. 5. As it is seen that if the θ is near $\pi/2$, the iteration process tend to be change direction quickly, and cannot find a good direction. From equation (13), when the θ is near $\pi/2$, the system tends to oscillation.

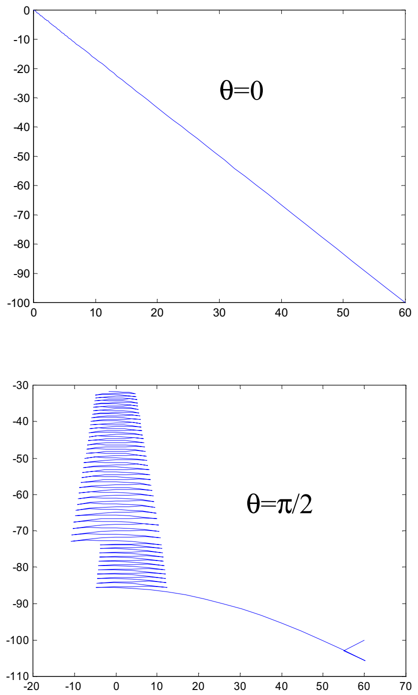


FIGURE 5: Phase Trajectories of Nonlinear System

For avoiding oscillation, a modified method is proposed to improve the convergent rate. First, making the matrix in equation (11) full of four parameters as (14), and then adjusting the weight to another two parameters, if oscillation happened.

$$\begin{bmatrix}\alpha(k+1)\\ \beta(k+1)\end{bmatrix}=\begin{bmatrix}\alpha(k)\\ \beta(k)\end{bmatrix}-\begin{bmatrix}\mu_{11}(k) & \mu_{12}(k)\\ \mu_{21}(k) & \mu_{22}(k)\end{bmatrix}\begin{bmatrix}n_1(k)\\ n_2(k)\end{bmatrix} \quad (14)$$

Firstly, trying with

$$|\mu_{11}(k)| = |\mu_{22}(k)| = 0.1/G_\omega; \quad |\mu_{12}(k)| = |\mu_{21}(k)| = 0,$$

When the sign change of parameters exceeds 25 in the first 50 steps of calculation, let

$$|\mu_{11}(k)| = |\mu_{22}(k)|=0; \quad |\mu_{12}(k)| = |\mu_{21}(k)| =0.1/G_\omega,$$

The parameter of equation (14) is modified as the following rule.

$$\begin{bmatrix}\mu_{11}(k)\\ \mu_{21}(k)\\ \mu_{12}(k)\\ \mu_{22}(k)\end{bmatrix}=\begin{bmatrix}\mu_{11}(k)\mathrm{sign}(n_1^2(k)-n_1^2(k+1))\\ \mu_{21}(k)\mathrm{sign}(n_1^2(k)-n_1^2(k+1))\\ \mu_{12}(k)\mathrm{sign}(n_2^2(k)-n_2^2(k+1))\\ \mu_{22}(k)\mathrm{sign}(n_2^2(k)-n_2^2(k+1))\end{bmatrix} \quad (15)$$

Simulation shows the modifying method can work well even without any knowledge of the transfer function, the results are shown in Fig.6.

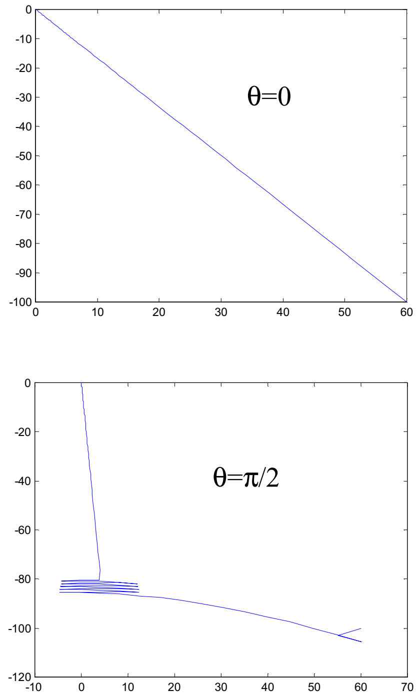


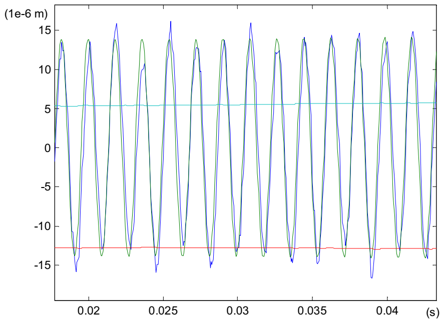
FIGURE 6: Simulation Results of Fast Convergence

EXPERIMENTS RESULTS

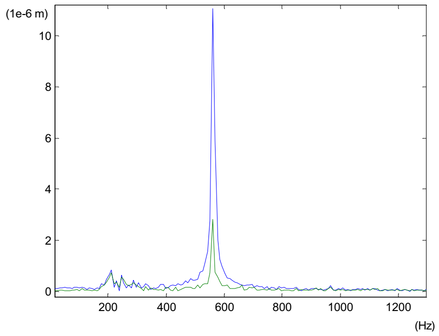
1. LMS Method Experiments

Fig.7 shows a compensation of input and cancel signal generator by LMS algorithm experiment, the cancel signal can accord with the input signal exactly. After compensating, the synchronous vibration decreases 10dB. Its spectrum shows that only the synchronous part is affected

Fig.8 shows LMS's influence to the control current at 570Hz. Without LMS control, the current is stronger (Fig.8a), with LMS control, the synchronous current is reduced very much (Fig.8b) which is almost the same to the static state. Of cause, because of not controlling the synchronous vibration, the magnitude is higher.

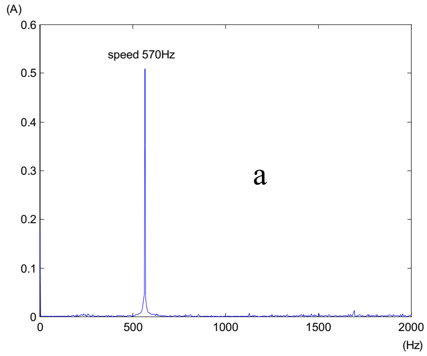


Input displacement signal and cancel signal, and the two coefficients of LMS algorithm

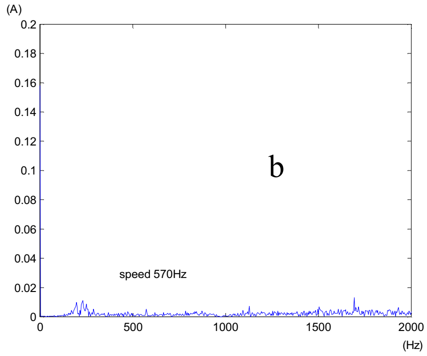


The input and error signal spectrum

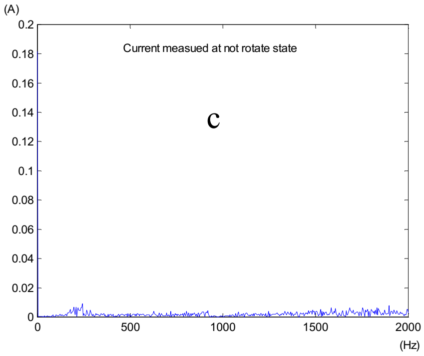
FIGURE 7: LMS Force Free Algorithm Result



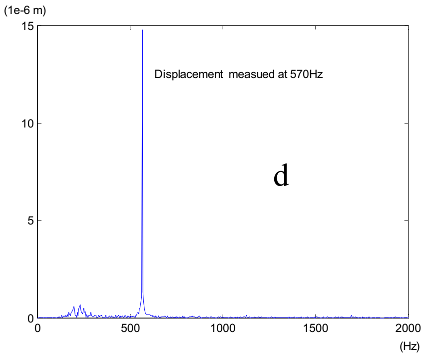
Current spectrum without LMS control



Current spectrum with LMS control



Static current not rotating

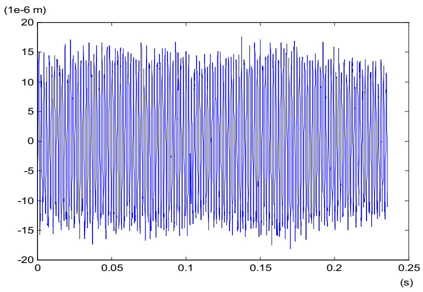


Vibration spectrum at 570Hz

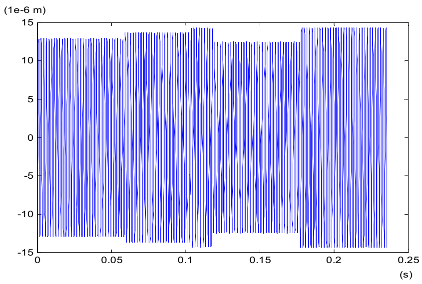
FIGURE8: LMS Force Free Algorithm influence

2. Fourier Coefficient

Fig.9 shows the experiment results. It is clear that Fourier coefficient method will generate uncontinuous compensating signal.



Displacement measured



Compensating signal.

FIGURE 9: Correlation Method

3. Force Controlled Control

Fig.10 shows experiment results at a rotor speed of 570Hz by force controlled control, when the current bias is 1.25 A. Because α is limited by program, the n_1 is not reach zero as n_2 does. As it is shown, the algorithm is convergence.

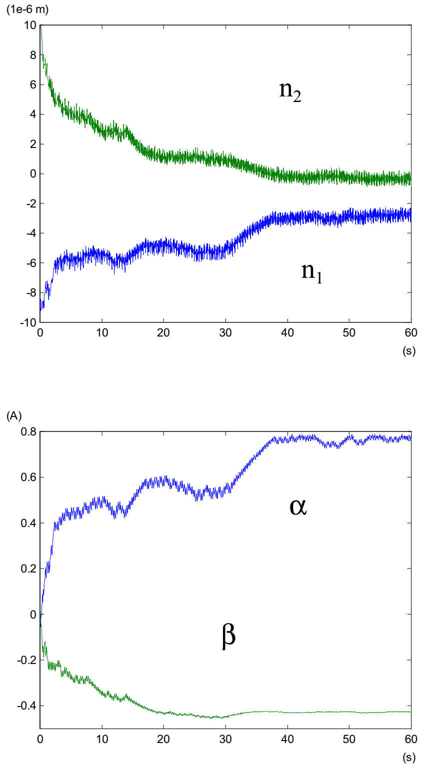


FIGURE 10: Experiment of Nonlinear Convergence Control

SUMMARY AND CONCLUSION

For different applications of AMBs-rotor system, high performance of the system can be achieved by unbalance vibration controlling by the FFC and FCC methods. which are demonstrated by the simulation and experiment results. The LMS algorithm in a fix point DSP, the step size will affect steadies error of compensation. When the step size increases, the steady error decreases; their product remains a constant. Especially, by the FCC method, the higher the speed, the more power need supplying.

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