耳機之主動控制實現
Implementation of Active Noise Control on the Headphone

陳國在 [1]、周俊宏 [2]、陳志煌 [3]

[1] 台灣大學工程科學及海洋工程學系教授

[2] 工業技術研究院機械與系統研究所研究員

[3] 台灣大學工程科學及海洋工程學系碩士

摘　要

　　本文利用FIR(Finite Impulse Response)數位濾波器，作為本論文之控制系統的主體架構，並以LMS(Least Mean Square)演算法來即時修正數位濾波器之係數，以求得最佳的控制效果。

　　在實驗方面，控制的頻率範圍在 600Hz 以下，適應性主動控制針對不同的噪音源(單頻，雙頻及白噪音)進行消音控制，此實驗起初是針對單邊耳機來進行消音，在單頻，雙頻及窄頻部分皆有 10~20dB 的消音效果。另外，雙邊耳機消音實驗中，因為聲波到達右耳和左耳存在時間差，因此兩支麥克風量到的聲音壓力不相同，最終導致控制後的結果差異。其中，單頻部分 300Hz,400Hz 及 500Hz 右耳和左耳控制後的音壓位準差值大約在 5~8dB。

關鍵字：主動噪音控制、有限脈衝響應、耳機

Abstract

　　This article utilizes the FIR (Finite Impulse Response) digital filter as the main structure of the control system and uses LMS (Least Mean Square) to modify in real time weighting coefficient of the algorithm to feedback the digital filter for trying to get the best control effectiveness.

　　As regard to the experiment, the frequencies under control are all below 600Hz and the noise sources to be adaptively controlled include those ones dealing with mono, dual, and white noise sources. The experiment of the study is firstly to try to attenuate the sound at the unilateral headphone, the result of noise reduction by 10~20dB for mono, dual, and white noises are obtained.

　　Then, we continue to set up the active attenuation on the bilateral headphone. Due to the different travel times for sound waves to arrive the right and the left ears, the sound pressures measured at the associated microphones are quite different, and so did the resulting noise reductions. Therefore, the distinction of control effectiveness by 5~8dB between two ears is obtained.

Keywords：Active Noise Control, Finite Impulse Response, Headphone

1. INTRODUCTION

Noise is an unwanted sound. Exposing in it too much time will seriously cause physical body to be unhealthy. Thus, how to reduce the exposure to high level noise had been a very important event to human life. Acoustic waves ranging from 20 to 20 KHz are audible to human ears and noise control is of course to be concentrated in the region as talked. Generally speaking, two methods including one passive and one another active control techniques are involved. Between them, passive technique uses barriers, absorption materials or their combination on the transmission path of sound wave to using acoustic isolation or absorption effectively reduce the associated level at high frequencies. Otherwise for the active control technique, a secondary source is used to produce a copied sound similar to the primary source to cause a destructively interaction between the two fields involved. This study mainly concentrates on the active control on the sound transmission through headphone from outside sources of various types.

Regarding to the past studies dealing with the active control concept, original idea was proposed by P. Leug in his 1936 US patent [1]. Further, earlier studies on ANC were concentrated upon signal processing situation to adaptive filters. Widrow proposed LMS (Least Mean Square) adaptive algorithm in 1970 [2] to be able to more accurately evaluate output error, in which the acoustic feedback from the secondary source and the error path were not concerned. Many similar works were continually worked out in several later years. To solve acoustic feedback and the effect resulting from error path, in 1987 Ericksson [3] proposed IIR filter to construct a FULMS adaptive algorithm by combining with RLS algorithm. In acoustic laboratory of NTU, Tsai took acoustic feedback into account to study the active control on the acoustic transmission in duct in 2003 [4] and got not bad effectiveness. Back to the past studies involved in this study, Lin adopted CFxLMS (weighted- constrained FxLMS) algorithm to more strongly protected the interference from microphone and then to reach the cancellation of outside noise to headphone in 2005[5]. In 2007, Shau [6] used combined type feedback active noise control arrangement to reduce the outside noise to headphone, and further renew the control parameters by matching the control system involved with Filtered-X least mean square (FXLMS) algorithm.

The motivation in this study is to design a headphone with background noise protection when someone would like to enjoy MP3 music quietly. To simplify the complexity of topics, regular background noise outside to the headphone is taken as source in experiment. Meanwhile, a usual open space and a reverberant space are both used as the sites for experiment. From the results of study, phase of measured error path spectrum shows linear characteristics ranging from 200 through 600 Hz. The control effectiveness corresponding to single, dual and finite bandwidth frequency noises are 10-20 dB, 10-15 dB and 10 dB respectively.

2. THEORETICAL CONSIDERATION ON ACTIVE CONTROL

2.1 Principle of active control:

As proposed by Leug in 1936, the mechanism of active control of sound results from the destructively cancellation between acoustic waves, which can be shown in Fig. 1 [4, 7]. In this figure, three signals dealing with the primary, the secondary sources and the

residuary one between them should play major roles in the active control process. In order to reach better cancellation of noise, two signals produced separately by primary and secondary sources in the region to be controlled must have the same magnitude but 180º out of phase.

2.2 Sampling theorem for continuous signal:

All natural signals in physical world such as electric pressure, temperature, wave patterns, etc., are analog. They have to be converted to be digital using A/D converter before being processed in computer system. To avoid aliasing of sampled signal, we must follow Nyquist theorem [8], that's to say the sampling frequency $F_{sampling}$ must greater than twice that of the maximum analytical frequency F_m of the measured signal, i.e. $F_{sampling} > 2F_m$.

2.3 Digital Filter:

As discussed in digital filtering theory [8], a digital filter is an algorithm to transfer an input sequence to another output sequence, and is also the core of the control system to be involved. Accordingly, the relation between the input and output sequences in discrete time domain can be expressed in difference equation as:

$$y(k) = \sum_{i=0}^{M} a_i x(k-i) + \sum_{j=1}^{N} b_j y(k-j) \qquad (1)$$

Where a_i, b_j are the characteristics of system . If $N = 0$ in eq.(1), the output $y(k)$ is the weighting average of input sequence and the filter involved is called the FIR filter(finite response filter) or moving average filter, which is abbreviated to be MA filter as well. Otherwise, if $M = 0$ in eq.(1), the resulting output is the auto-regression of its earlier one, and the resulting system is called the auto-regress model or abbreviated as AR model. Further , if M, N are not both zero, the resulting filter is called the IIR (infinite impulse response) filter or ARMA model. Fig. 2, 3 show the arrangements for FIR and IIR digital filters, respectively. Taking Z transform to the output response of IIR filter can obtain the corresponding transfer function as:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{\sum_{i=0}^{M} a_i z^{-i}}{1 - \sum_{j=1}^{N} b_j z^{-j}} = \frac{A(z)}{1 - B(z)} \qquad (2)$$

The pole in eqn(2) will drive IIR digital system unstable. Thus to avoid the instability of system involved, a FIR model is selected in this study.

2.4 LMS algorithm:

An adaptive digital filter mainly consists of two parts, which are a digital filter to process the desired output signal and an algorithm to modify the weighting parameters of the above filter. Fig.4 shows the prototype for a digital filter, in which d_k is the desired response , x_k is the reference input signal, y_k is the actual output , and error signal e_k is

the difference signal between the actual signal y_k and the desired signal d_k. Accordingly, the function for the adaptive algorithm as discussed is used to continually modulate the parameters of the adaptive filter involved to drive error signal to reach minimum. Fig. 5 shows a FIR filter of order L, the associated reference input signal in time k can be expressed in vector form as:

$$X_k = \begin{bmatrix} x_k & x_{k-1} & \ldots & x_{k-(L-1)} \end{bmatrix}^T \tag{3}$$

And the relation between output and input is

$$y_k = \sum_{l=0}^{L-1} w_{lk} x_{k-l} \tag{4}$$

The weighting coefficients in eqn(4) can be expressed in weighting vector of order L as:

$$W_k = \begin{bmatrix} w_{0k} & w_{1k} & \ldots & w_{(L-1)k} \end{bmatrix}^T \tag{5}$$

In eqn(5), T is the transpose operation of matrix. Combination of eqn(3) with eqn(5) gives

$$y_k = W_k^T X_k = X_k^T W_k \tag{6}$$

The corresponding error signal in time k is

$$e_k = d_k - W_k^T X_k = d_k - X_k^T W_k \tag{7}$$

Taking square of eqn(7) can obtain the energy of the error signal as:

$$e_k^2 = d_k^2 + W_k^T X_k X_k^T W_k - 2d_k X_k^T W_k \tag{8}$$

In eqn(8), it always assumes that the desired response and the reference input signal are all statistically stationary signal such that a filter with optimum weighting vector are required to drive the mean square error signal to be minimum. Following the result as obtained in Chen's study [9], using method of steepest descent can yield the iterative formula for the optimum weighting vector as

$$\begin{aligned} W_{k+1} &= W_k - \frac{1}{2}\mu\nabla\hat{\xi} \\ &= W_k + \mu e_k X_k \end{aligned} \tag{9}$$

Eqn(9) is the so-called LMS rule, in which μ is the step side parameter for every time iteration. From the text written by S. Haykin [10], the criteria of μ for eqn(9) to be stably convergent is

$$0 < \mu < \frac{2}{\lambda_{max}} \tag{10}$$

Where λ_{max} is the greatest eigen value of the auto-correlate matrix for the input reference signal.

3. RADIATION EFFECT OF HUMAN EARS
3.1 Direction perception of source

The masking effect of two ears on sound source might cause inconsistent sound pressure measurements at their error microphones when making the experiment involved in progress. At higher frequencies, the difference of acoustic intensity between two ears will become more apparently to produce great influence on the control performance.

Regarding to the direction perception of source is deterministically influenced by both

the inter-aural intensity difference (IID) and the inter-aural time difference (ITD) of the propagating sound from source between two ears. Summarily to say, you will feel sound source behind your right back when large intensity received at right ear. Meanwhile due to smaller diffraction at high frequency, a little change of orientation for sound source will produce greater intensity difference between two ears. Accordingly, direction perception of sound source will be determined by ITD at low frequency but determined by IID at high frequency, which are shown in Fig.6. The conclusion as discussed is called the duplex theory and was proposed by Lord Rayleigh in 1907[11].

To explain the detail as discussed, Fig.7 clearly shows acoustic wave from source reaches right ear at first and then reaches left ear next, the resulting path difference is $a\theta + a\sin\theta$, which corresponds to inter-aural time difference:

$$ITD = \frac{a}{c}(\theta + \sin\theta) \ , \ -90^o \le \theta \le 90^o \tag{11}$$

Eqn(11) clearly reveals ITD is zero when the source is directly ahead to human head ($\theta = 0$) and reaches maximum when $\theta = 90$ or $\theta = -90$. Accordingly the masking effect from human head becomes apparent at high frequency and IID determines the perception of sound at ear. Otherwise, the diffraction effect from human head is apparent at low frequency and ITD determines the corresponding perception of sound from source at human's ears. Whatsoever, ITD and IID provide very important information for the directional perception of sound source to human's ears.

3.2 Directivity of source

The radiation of any sound source is not uniform and has some directivity. Accordingly the sound pressure level produced by a directive source will not be uniform in different direction. Generally speaking, most sources are not directive at low frequency, and get more directivity when frequency gets higher. For a baffled piston source, as shown in Fig. 8, the acoustic pressure at far field along axial direction can following the derivation in Chen's study [9] be expressed as

$$\therefore \quad p(r,\theta,t) = j\frac{\rho_0 c}{2}U_0\frac{a}{r}ka\ e^{j(\omega t - kr)}\left[\frac{2J_1(ka\sin\theta)}{ka\sin\theta}\right] \tag{12}$$

Where the direction factor can be expressed as

$$H(\theta) = \left|\frac{2J_1(ka\sin\theta)}{ka\sin\theta}\right| = \left|\frac{2J_1(v)}{v}\right| \tag{13}$$

Looking into the tendency of direction factor with respect to θ for various ka can find that stronger directivity exists when ka gets greater. The comparison of two results as shown in Fig.9 and Fig.10 for $ka = 1.0, 10$ can clearly reveal the above conclusion.

4. SET UP FOR EXPERIMENT
4.1 Experimental equipments:

To achieve the acoustic cancellation in human headphones, a loudspeaker enclosed by a cubic box of each side by sixteen centimeters is used as the primary source. Meanwhile, all equipments involved in the experiment are chosen by following Chen's study [9] and

can be simply illustrated as follows: (1) Function generators include HP3314A and B&K 2035 for respectively producing pure and dual tone signals. (2) Sunlight 160 is used to be a power amplifier to provide enough output for the primary source. (3) A disk type loudspeaker of 9 cm in diameter enclosed by a cubic box is used as the primary source. (4) A common headphone is purchased in general market. (5) Another TERK 601C power amplifier is used to amplify by 100 times the output from the controllable secondary source. (6) A capacitance type B&K 4190 microphone of 1/2 inch in diameter is used as an error microphone to take error signal from the field to be controlled and then is delivered to PC for control feedback. (7) A B&K 2633 preamplifier is used to amplify 50 times the signal received at microphone. (8) A B&K 2804 power supply is used to provide powers required by preamplifier and microphone. (9) RION SA-59 one-third octave band analyzer is used to filter out noise signal at high frequency. (10) Intel E6600 dual-core central processor with memory of 1024Mbytes is used as the processor of the control system involved in experiment.(11) A PCI-1712 AD/ DA converter, developed by ENHUA Co., is used to transfer an analog signal to a digital one or reversely. (12) B&K 2035 dual channel signal analyzer is used in both time and frequency domains to examine whether desired reduction of acoustic field to be controlled is reached or not. (13) Borland C++ version 3.0 programming language is used to construct digital control system and is used as the applied software for algorithm.

4.2 Experimental setup and procedure involved:

All equipments involved in experiment are arranged as shown in Fig. 9. By this, we can use HP3314A function generator to produce a pure-tone signal of frequency range from 200 to 500Hz and then using power amplifier to generate the required noise from the primary source. Meanwhile in PC, we use Borland C++ program language to compile control software, and then use PCI-1712 AD/DA card as the interface between input and output. Therefore, use of A/D converter can transfer both the reference signal from the primary source and the signal received at error microphone to digital ones for the processing in computer, and further using D/A converter can transfer digital output from computer to become analog signal for the control on the field to be the desired level in headphone. Fig. 10 shows the flow chart for the adaptive active control experiment involved in this study. Some more studies dealing with the effect on system by error path, optimum noise cancellation of headphone and the sampling simulation of measurement were discussed in detail in Chen's study [9] and will not involved in this article.

5. RESULTS AND DISCUSSIONS
5.1 Experimental results in reverberation room

In this study, experiments for both one ear and two ear headphones were made in reverberation room. Among them, some single frequencies 200, 300, 400, 500 Hz, some dual tone frequencies (200Hz, 300 Hz), (300 Hz, 400 Hz), and central frequencies 300 Hz , 400 Hz with some same bandwidths of 50 Hz, 100 Hz , and 200 Hz are selected for one ear headphone in experiment. Meanwhile, some frequencies such as 300, 400 and 500 Hz are chosen in experiment for two ear headphones. As regards to the order of the transfer function and the associated step side parameter for the FIR filter to be used, after some trials and errors the best choice for two items as talked are $L=10$ and

$\mu = 0.001$ respectively. Some typical results at 300 Hz, (300 Hz, 400 Hz), and 300 Hz with bandwidth 50 Hz for one ear headphone, and the other ones at 400 Hz, 500 Hz and low pass at central frequency 400 Hz for either right or left ear are respectively shown in Fig. 11 through Fig.18.

5.2 Experimental results in open space

Following the same procedure of experiment as that in reverberation room can get overall results for that in open space. Some typical results of the same experimental situations as that in former case are shown in Fig. 19 through Fig. 21.

5.3 Discussions

Comparing the experimental results for those obtained in both reverberation room and in open space will obviously reveal almost all measurements have the same effects at the main central frequencies except having greater reduction of pressure levels around those central frequencies in reverberant room. Among them, the phase spectrums are linear from 200 Hz through 600 Hz and the better reductions of active cancellation on one ear headphone are around 10dB to 20 dB for either pure, dual tone or narrow band noise, and 10 dB for low pass noise respectively. Regarding the control effectiveness for right or left ear headphone, there is some discrepancy between them due to different inter-aural time difference from source and different scattering effects by head on incident waves. Furthermore, from the phase spectrum of error path involved, there is instability, nonlinearity at low frequencies below 200 Hz to cause ineffective control in this frequency zone.

6. Conclusions

Upon the discussions on the last paragraph, the following conclusions for this study are obtained:

1. Instability and nonlinearity of error path at low frequency below 200 Hz will cause ineffective cancellation of noise in headphone.
2. Due to linear characteristics of phase spectrum of the error path transfer function at frequencies from 200 through 600 Hz for the adaptive FIR control system used in this study, the best control effectiveness is obtained.
3. Due to different inter-aural time difference from noise source to left and right ears and different scattering effect on incident waves by human head there is not small discrepancy of active cancellation on noise in headphones of two ears.
4. Except having different reduction in sound pressure level in headphones around main central frequency to be measured, all measurements for both in reverberant and open spaces almost have the same control effectiveness.
5. Due to essential inter-aural time difference from noise source between two ears, it is impossible to completely cancel noise in headphone, which is also to drive to have better effectiveness for more stable noise to be controlled.

References

[1] P. Lueg, Process of Silencing Sound Oscillation, U.S. Patent No.2, 043,416, 1936.
[2] B Widrow and E. Walach, Adaptive inverse control ,Prentice Hall,Upper Saddle river, NJ,1996.

[3] Eriksson, L.J., M.C. Allie, and R.A. Greiner, the selection and application of an IIR adaptive filter for use in active sound attenuation, IEEE Transaction on acoustics, Sppech, and Signal Processing, Vol. ASSP-35, No.4, pp.433-437, April 1987.

[4] Tsai, Application of acoustic feedback separation on the active control of sound transmission in duct, master thesis of NAOE, NTU, 2001.

[5] C.Y. Lin, Z. H. Hwang, and A. Z. Shou, Active control of noise in headphone by using the restraining of weighting function, master thesis, EE, NCNU, 2005.

[6] J. H. Shou, S. Y. Lu, Implementation on the development of active headphone by using digital signal processing, master thesis, ME, NTUST, 2007.

[7] K.W. So, Active control on sound in duct by using a controllable plate source, master thesis, ESOE, NTU, 2007.

[8] A. V. Openheim, A. S. Willsky, and S. H. Nawab, Signals & systems, 2nd Ed., Prentice-Hall, 1997.

[9] Z. H. Chen, Implementation of active control on the headphones, master thesis, ESOE, NTU, 2008.

[10] S. Haykin, Adaptive filter theory, 4th Ed., Prentice-Hall, NJ, 2002.

Illustrations
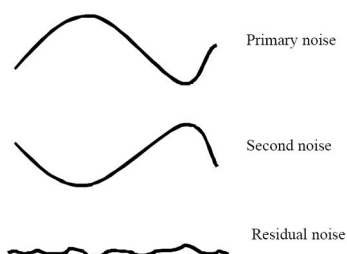
Fig.1 Mechanism of active noise control
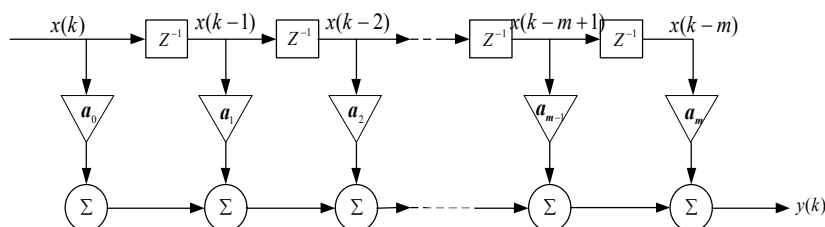
Fig.2 Structure of a direct type digital FIR filter
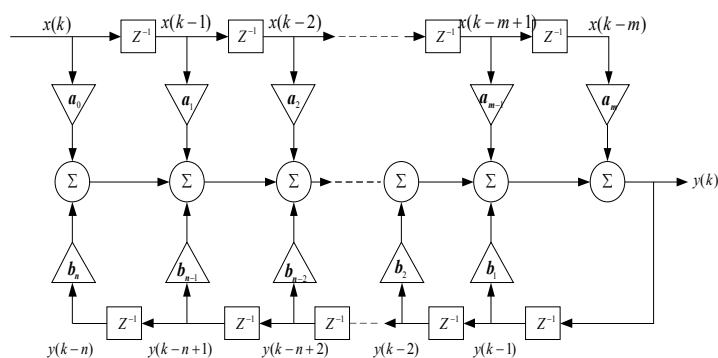
Fig.3 Structure of a direct type digital IIR filter
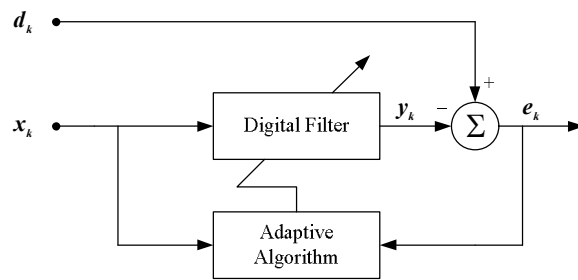
Fig.4 Block diagram of an adaptive digital filter
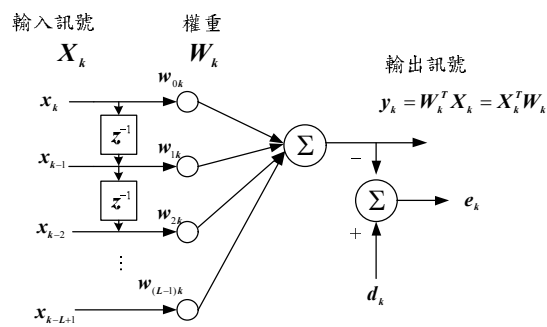
Fig.5 A digital FIR filter of order L
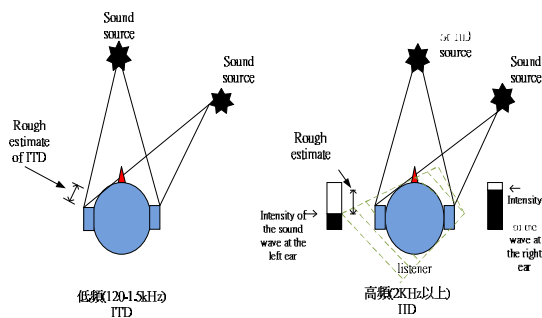
Fig.6 Illustration to masking effect by human head
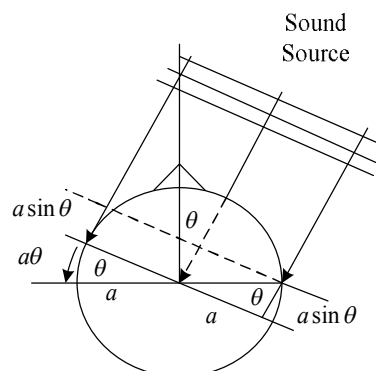
Fig. 7 Arrival path difference to two ears due to diffraction
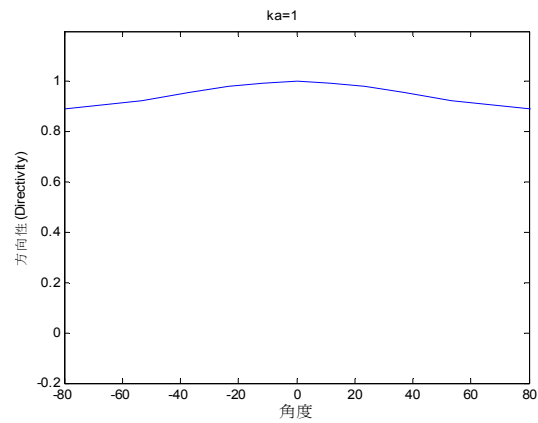
Fig. 8 Directivity factor of source versus angle of inclination when $ka = 1$.
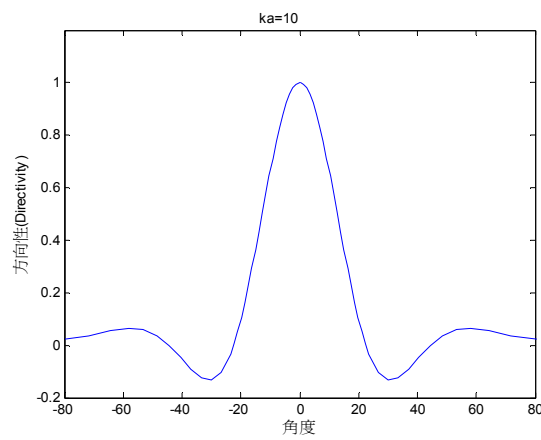
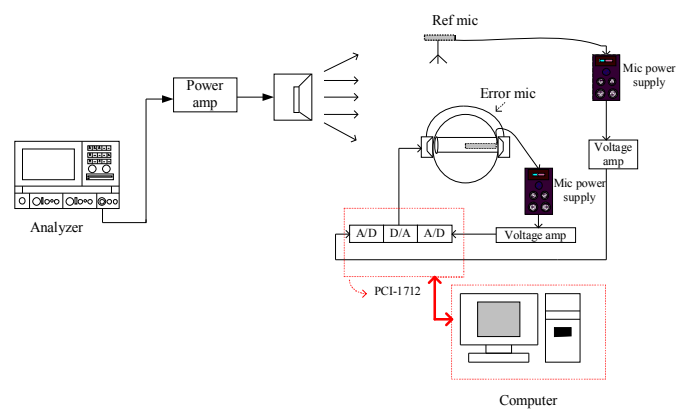Fig. 9 Directivity factor of source versus angle of inclination when $ka = 10$

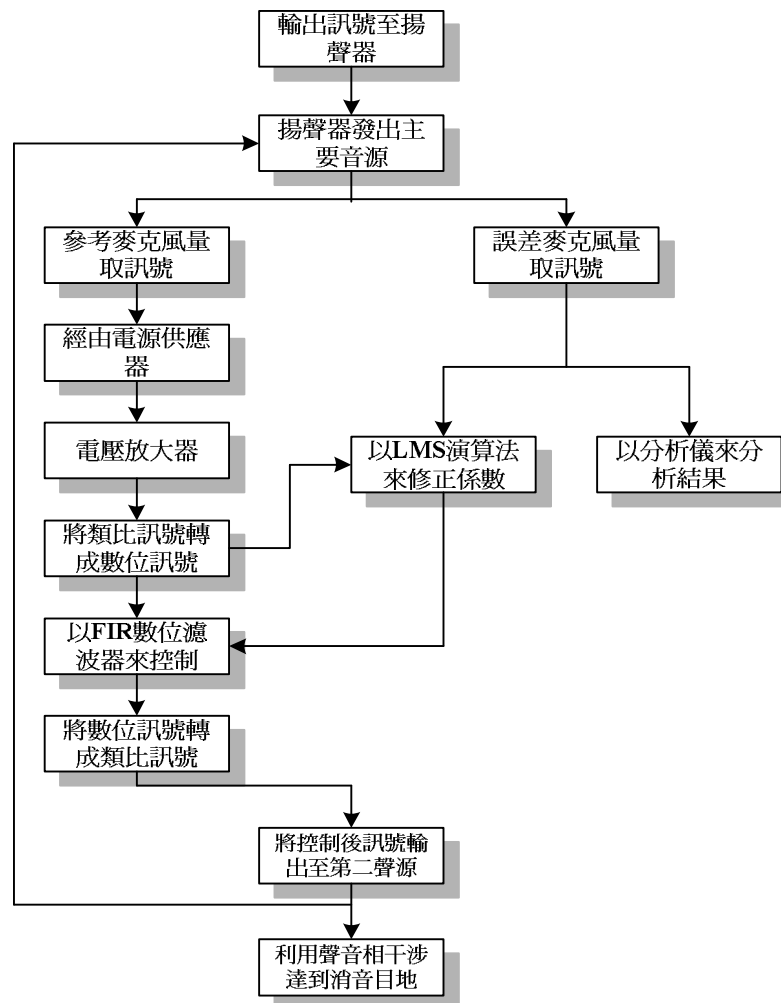Fig.10 Equipments arrangement for experiment setup
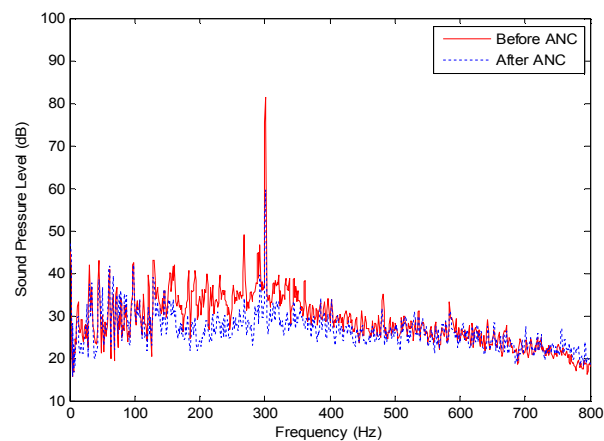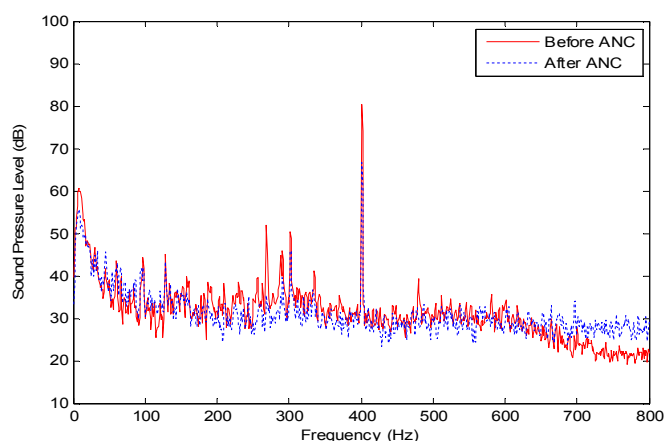
Fig.10 Flow chart for the adaptively active control experiment
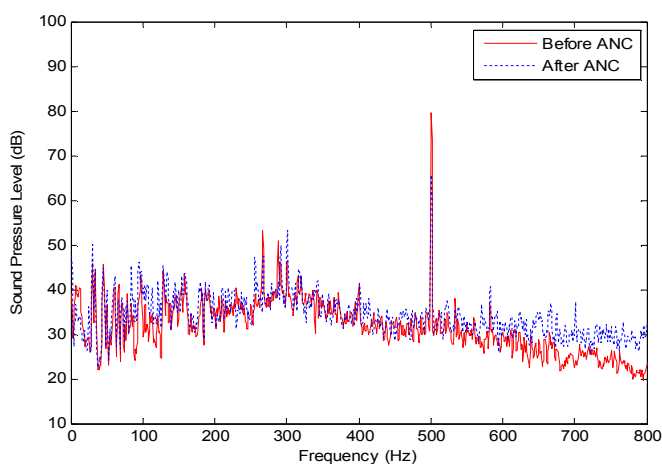

Fig.11 Sound pressure at 300 Hz in one headphone before and after active control
(in reverberation room)
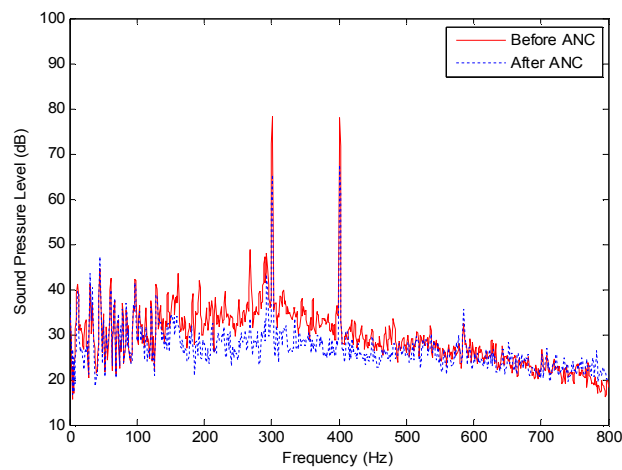
Fig.12 Sound pressure levels at dual-tone (300 Hz, 400 Hz) in one headphone before and after active control. (in reverberation room)
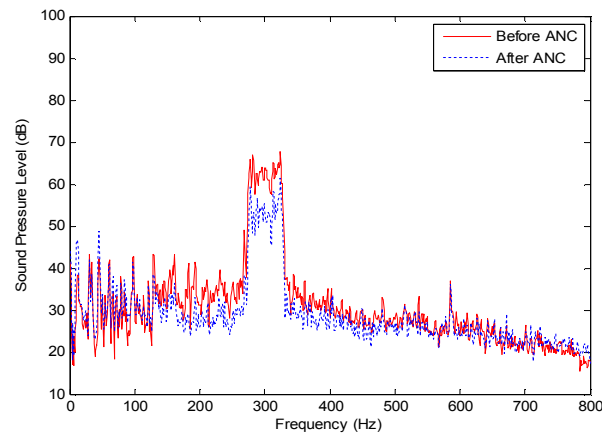
Fig.13 Sound pressure levels at central frequency 300 Hz, with bandwidth 50 Hz in one headphone before and after active control. (in reverberation room)
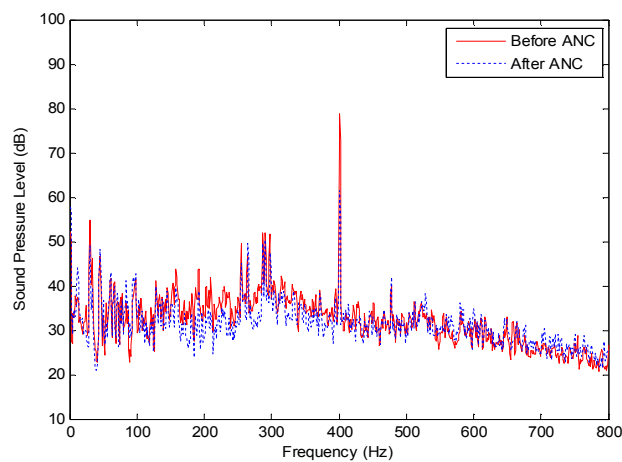
Fig.14 Sound pressure levels at 400 Hz in right headphone before and after active control. (in reverberation room)

Fig.15 Sound pressure levels at 400 Hz in left headphone before and after active control. (in reverberation room)

Fig.16 Sound pressure levels at 500 Hz in right headphone before and after active control. (in reverberation room)
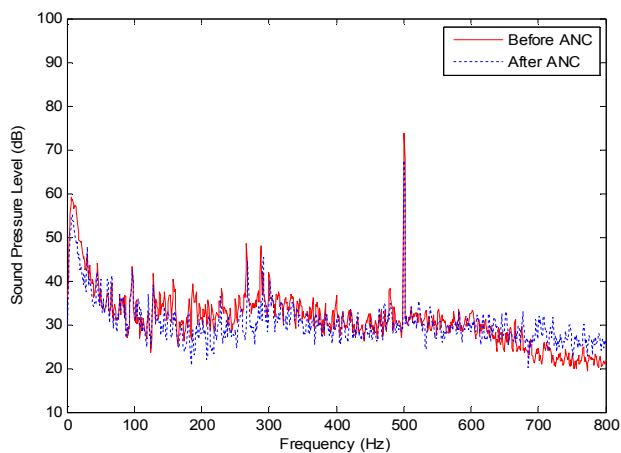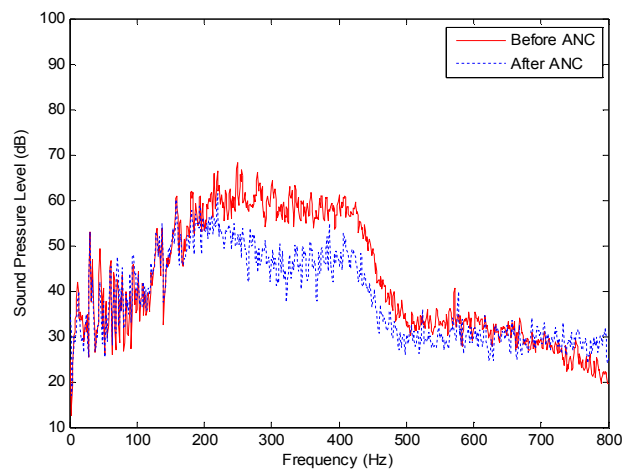

Fig.17 Sound pressure levels at 500 Hz in left headphone before and after active control. (in reverberation room)

Fig.18 Sound pressure levels of low pass noise at central frequency 400 Hz in two ears headphones before and after active control. (in reverberation room)
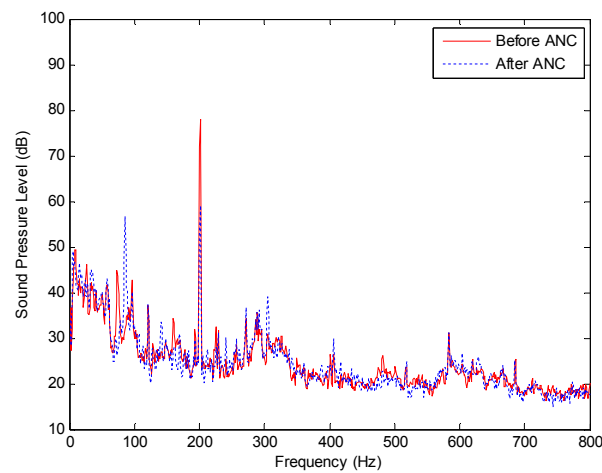

Fig.19 Sound pressure at 200 Hz in one headphone before and after active control (in open space)
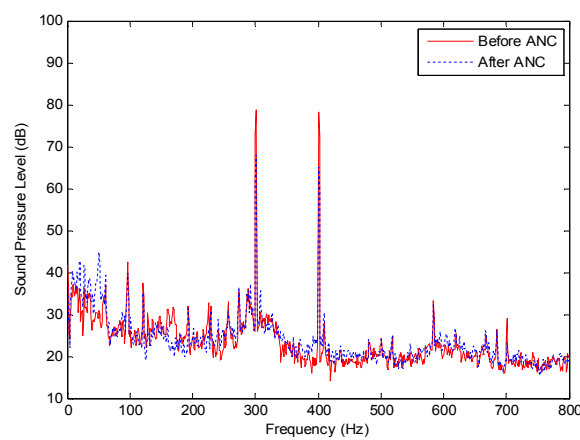

Fig.20 Sound pressures at dual-tone (300 Hz, 400 Hz) in one headphone before and after active control (in open space).
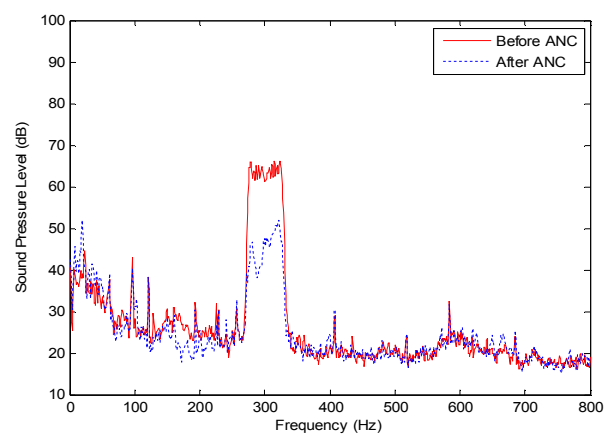
Fig.21 Sound pressures of narrow band noise at 500 Hz with bandwidth 50 Hz in one headphone before and after active control (in open space)