Development of Integrated Headphone Communication system

Chien-Wun Chiu
Department of Electronic Engineering,
Lunghwa University of Science and Technology,
Postgraduate
jc0804@ethome.com.tw

Chun-Cheng Lin
Department of Electronic Engineering,
Lunghwa University of Science and Technology,
Assistant Professor
cclin@mail.lhu.edu.tw

摘要

在現行的通訊裝置中，耳機可以帶來使用上的便利，也可以降低電磁波對於人體的潛在影響。再者，在吵雜的環境中，如果可以利用耳機產生反噪音訊號，則可以進一步提升雙向通訊的品質。據此，本研究將發展整合型適應性回授式主動噪音控制(IAFANC)通訊系統應用於手機耳機，以降低耳朵聽到的噪音強度，以及避免近端噪音傳送至遠端。本研究將使用人工合成之弦波噪音、引擎低速與高速運轉時之噪音作為近端的噪音訊號，以測試 IAFANC 的性能。實驗結果顯示對於訊號雜訊比-20dB 的人工合成之弦波噪音、引擎低速與高速運轉的噪音，噪音衰減量分別為達到 62.8dB、21.8dB 與 29.6dB。

關鍵字：最小均方演算法、主動式噪音控、制適應性濾波器

Abstract

Nowadays headsets commonly found in currently widely used communication devices indeed provide convenience in use and also reduce potential impacts of electromagnetic wave to human body. Further, if it is possible to use a headset to generate anti-noise signals when exposed in noise surrounding, then it is possible to enhance the quality of two-way communication. In view of this premises, this study aims at developing an integrated adaptive feedback active noise control (IAFANC) communication system for applications in cell phone headsets for the purposes of reducing the noise level received by human ears and preventing near-end noise from being sent to the far end. Furthermore, in this study only artificial synthesized sine noises and noise emitted from an engine running at high or low rpm are to be adapted as near-end noises for better testing the performance of the IAFANC communication system in question. The lab experiments results demonstrate that the IAFANC communication system has achieved noise reduction at 62.8dB, 21.8dB and 29.6dB respectively in its application to artificially synthesized sine noise with signal noise ratio -20dB, noise from engine low speed operation noise and engine high speed operation noise.

Keyword: LMS, ANC, Adaptive Filter

1. Introduction

A headset provides convenience in application and reduces potential impact of electromagnetic wave to human body when it serves as an element in a hand-held audio device such as MP3 player, cell phone, PDA, or radio communication device, etc.. A headset promotes reception quality straightly especially in noisy environment. However reception quality of a near-end speech voice, or in some cases, even far-end signal, is seriously affected upon occurrence of any noise interference problem from industrial equipment in use such as engine, fan, transformer, compressor, motorcycle or automobile, etc.. To reduce noise impact it is possible to reduce noise of higher frequencies with a conventional passive method a headset with higher shielding capacity, however the passive method proves to be poor in attenuating noise of lower frequencies and entirely helpless in promoting far-end signal reception quality. Hence it is desperate to develop a headset type active noise attenuation system which can cope with both signal reception and transmission. Active noise control (ANC) system achieves noise attenuation by offsetting a noise signal with an anti-noise signal of the same amplitude but the opposite phase.

In the latest research, feedforward technology is applied to ANC communication headsets [2]-[4]. However, the independent sensor used in feedforward technology invites a problem involving near-end noise signal interference. Furthermore, the independent sensor of excessively high price results in increase of product cost, therefore, it becomes desperate to develop and design a more appropriate system and IAFANC is the answer. This paper falls in two parts: The first part aims at establishing an adaptive feedback active noise control (AFANC) system, evaluating the secondary path transfer function, and discussing the design of IAFANC system. The second part deals with performance evaluation on computerized simulation of noise signal ratio between different near-end noises and other different noises.

2. IAFANC System

This section describes the Filtered-X LMS algorithm, i.e. FXLMS algorithm, in AFANC system and the overview of estimating the secondary path transfer function, and discussing the design of IAFANC system in applications in cell phone headsets.

2.1 AFANC System

In an adaptive feedback active noise control (AFANC) system, the main noise signal $d(n)$ is ineffective, since it has been offset by the adjacent anti-noise signal $y(n)$ already. AFANC is different from adaptive feedback control (ANC) system in that the reference signal $x(n)$ is from the source of the independent sensor. AFANC system needs to synthesize the reference signal $x(n)$ by using estimation techniques in order to change the weight in the adaptive filter $w(n)$.

Fig. 1 is the illustration of the FXLMS algorithm [1] in AFANC system, where $\hat{s}(z)$ is the compensatory secondary path transfer function. The reference signal $x(n)$ serves as estimation to $d(n)$, represented as:

$$x(n) \equiv \hat{d}(n) = e(n) + \sum_{m=0}^{M-1} \hat{s}_m y(n-m) \qquad (1)$$

where $\hat{s}_m$, $m = 0,1,2,3,4 \cdots M-1$, is the parameter of M order Finite Impulse Response (FIR) Filter for estimating the secondary path. Anti-noise signal $y(n)$ is represented as:

$$y(n) = \sum_{l=0}^{L-1} w_l(n) x(n-l) \qquad (2)$$

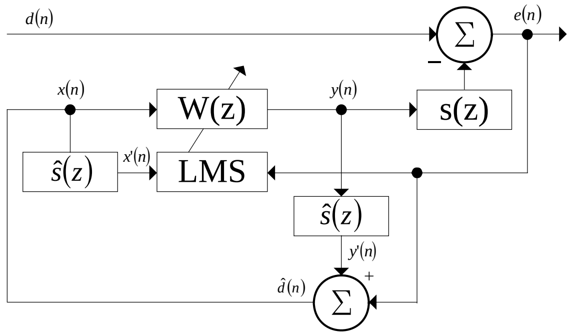

Fig 1. AFANC System

where $w_l(n)$, $l = 0,1,2,3,4 \cdots L-1$, is the parameter of L order FIR filter when $W(z)$ is at time n. The Filter parameter through FXLMS algorithm is updated as the follows:

$$w_l(n+1) = w_l(n) + \mu x'(n-1) e(n) \qquad (3)$$

where μ is the step value. $x'(n)$ is the reference signal after being filtered, and is represented as:

$$x'(n) = \sum_{m=0}^{M-1} \hat{s}_m y(n-m) \qquad (4)$$

2.2 Estimation of the secondary path transfer function

Following Section 2.1, the so called secondary path $s(n)$ denotes the anti-noise signal $y(n)$ and the feedback path formed by an array of devices including the headset, microphone and other hardware equipment the said signal travels through. To eliminate the influence from $s(z)$ it is a must to conduct estimation and update with FXLMS algorithm. The $s(z)$ estimation normally adapts off-line estimation technology which recovers and completes the earlier training signal and inputs the signal jointly to the adaptive filter $\hat{s}(z)$ and the real secondary path $s(z)$, and such a structure serves as reference for identifying conventional adaptive systems.

Selecting the continuous emitting training signal for imitating broadband secondary path is crucial and vital. In view of the uniform distribution of spectrum density of broadband bandlimited white noises, the said noises can server as very ideal broadband training signal for

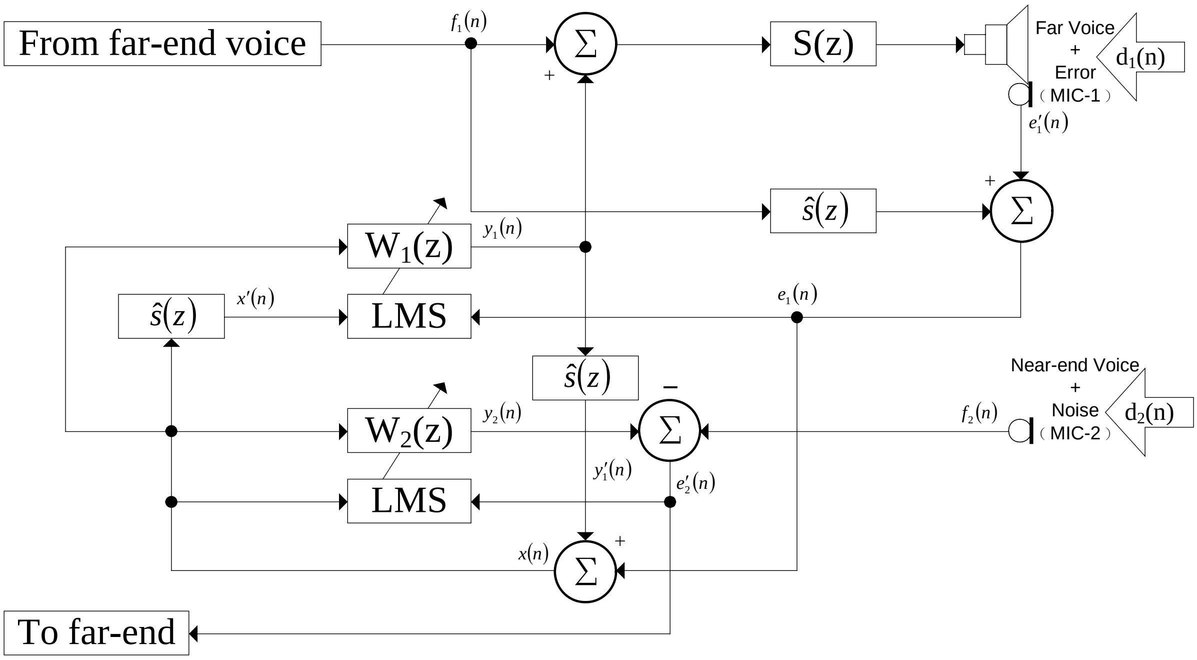

Fig 2. Block diagram of cell phone headset with AFANC system

system identification. However in the application of ANC headsets, any signal of off-line estimation is heard upon an AFANC headset is triggered, therefore, it is necessary to adapt a quieter signal (for example: bird singing) reasonably to serve as training signal.

2.3 Integrated adaptive feedback active noise control (IAFANC) system

Fig 2. The filter $W_1(z)$ further integrates the AFANC system mentioned in 2.1 and the far-end speech voice it received thus forms the IAFANC systems [5]-[6] ideal for cell phone headset application. As shown in Fig 2, the residual near-end noise and Far-end speech voice are obtained through microphone MIC-1, next, minus the far speech voice signal synthesized reference signal $x'(n)$, at last, update the adaptive filter parameter $W_1(z)$ by using FXLMS algorithm. Fig 2 demonstrates the integration of currently available hand-held audio devices such as MP3 player, PDA and radio communication, etc.; the microphone MIC-1 is placed inside of the headset to obtain the residual noise signal and Far-end speech voice. However, components of far-end speech voice is likely to interfere the estimation performed by adaptive filter $W_1(z)$. Assuming $S(z)=\hat{S}(z)$ and $B_1(z)=B_2(z)=X(z)$, microphone MIC-1 output signal $e_1'(n)$ including the residual noise (or called error single) $e_1(n)$ plus Far-end speech voice ,can be presented as:

$$E_1'(z)=B_1(z)-(F_1(z)+Y_1(z))S(z) \tag{5}$$

$$E_1(z)=E_1'(z)+\hat{S}(z)F_1(z) \tag{6}$$

Sum (5) and (6) up to obtain the result (7), where $b_1(n)$ is near-end noise also known as background noise, $y_1(n)$ is the anti-noise signal estimated against $b_1(n)$ by adaptive filter $W_1(z)$.

$$E_1(z)=B_1(z)-F_1(z)S(z)-Y_1(z)S(z)+F_1(z)\hat{S}(z) \tag{7}$$

After further simplification (7), the follows are derived:

$$E_1(z)=B_1(z)-Y_1(z)\hat{S}(z) \tag{8}$$

$$X'(z)=E_1(z)+Y_1(z)\hat{S}(z) \tag{9}$$

Where $x(n)$ is reference signal, the comparison between (8) and (9) proves that both adaptive filters $W_1(z)$ and $W_2(z)$ obtain their precise reference signals.

For quality speech voice communication, near-end speech voice must be such treated that no near-end noise is left before it is transmitted to the far-end. Fig. 2 microphone MIC-2 I use for obtaining mainly the near-end speech voice and near-end noise, and adaptive filter $W_2(z)$ can be used for removing near-end noise. The reference noise input $x(n)$ is used for training the noise canceling Filter $W_2(z)$ and its LMS algorithm is derived as:

$$N_l(n+1) = N_l(n) + \mu x(n) e_2(n) \tag{10}$$

where $l = 0,1,2,3,4 \cdots L-1$, is the parameter of L order FIR Filter $W_2(z)$.

3. Performance evaluation

Computerized simulation is conducted for the time delay of about 8 units in the secondary path transfer function and the pre-recorded near-end speech voice, far-end speech voice and near-end noise with sample ratio of 8KHz are input into the computer for simulation process. For the part of near-end noise, artificial synthesized sine noise, and engine low speed operation noise as well as engine high speed operation noise are adapted as the near-end noises. The adaptive filters $W_1(z)$ and $W_2(z)$ adapt order 192 FIR filters, and their step values are about 97×10^{-6} and 100×10^{-6} respectively.

Fig 3.a and Fig 3.b illustrate the far-end speech voice and near-end speech voice interfered by sine noise, where the components of synthesized sine noise are demonstrated below as in (11)

$$\begin{aligned} b_1(n) = b_2(n) = 1.5\cos(200\pi t) \\ + \sin(120\pi t) + 2\cos(400\pi t) \end{aligned} \tag{11}$$

signal noise ration SNR (dB) is shown below as in (12)

$$SNR(dB) = 10\log_{10}\frac{E[f^2(n)]}{E[x^2(n)]} \tag{12}$$

Fig 3.c and Fig 3.d illustrate the result of anti-noise signals $y_1(n)$ and $y_2(n)$ after cancellation of sine noise and the result include the error signal $e_1'(n)$ obtained from far-end speech voice minus far-end voice signal $f(n)$ in wave shape as shown in Fig 3.e. Fig. 3f is the result obtained by deducting pre-recorded near-end voice signal from the result of anti-noise signal $y_2(n)$ being offset by sine noise.

Noise reduction R(dB) is derived by the formula (13) and the noise reduction under different SNR can be found in Table 1.

$$R(dB) = -10\log_{10}\frac{E[e_1^{\,2}(n)]}{E[x^2(n)]} \tag{13}$$

Fig 4 and Fig 5 illustrate the computerized simulation results taking engine low speed operation noise and engine high speed operation noise as samples; further, the noise reduction of each of the 3 main frequencies under different SNR can be found in Table 2 and Table 3. Figs 6 to 8 illustrate the noise reduction spectrum analyses of the three different near-end noises when SNR = -10dB and -20dB respectively.

Table 1. Noise reduction of receiving far-end speech voice interfered by sine noise

SNR(dB)	R(dB)		
	100HZ	60HZ	200HZ
0	30.6	28.2	13.4
-10	52	64	53
-20	62.8	65	64.4

Table 2. Noise reduction of sending near-end speech voice interfered by Sine noise

SNR(dB)	R(dB)		
	100HZ	60HZ	200HZ
0	51.7	68.8	54.2
-10	67.7	78.6	67.6
-20	70.3	72.8	69.6

Table 3. Noise reduction of receiving far-end speech voice interfered by engine low speed operation noise

SNR(dB)	R(dB)		
	143HZ	183HZ	196HZ
0	20	12.6	11.6
-10	22	14.6	12.4
-20	21.8	15	13

Table 4. Noise reduction of sending near-end speech voice interfered by engine low speed operation noise

SNR(dB)	R(dB)		
	143HZ	183HZ	196HZ
0	14.7	19.3	15.2
-10	23.1	25	19.2
-20	26.8	25.1	20.5

Table 5. Noise reduction of receiving far-end speech voice interfered by engine high speed operation noise

SNR(dB)	R(dB)		
	199HZ	239HZ	279HZ
0	20	6	7
-10	27.8	17	15.6
-20	29.6	18.2	16.6

Table 6. Noise reduction of sending near-end speech voice interfered by engine high speed operation noise

SNR(dB)	R(dB)		
	199HZ	239HZ	279HZ
0	21.6	3.5	-16.5
-10	28.1	11.2	-4.4
-20	30.9	20.5	7.6

4. Discussion

Among the three different IAFANC communication systems for testing different near-end noises, adaptive filter $W_1(z)$ proves profoundly influenced by the secondary path. Even the after having taken the influence into consideration, the error signal wave amplitude $e_1(n)$ is promoted inevitably, even far more clearly seen when compared to $e_2(n)$ which is free from influence of the secondary path.

Taking an overview over Figures 6 thru 12, both the noise from engine running at high rpm and the artificial synthesized sine noise demonstrate no significant change over time, therefore, hold the stationary characteristic statistically, consequently, IAFANC communication systems expect relatively exceptional attenuation effect to the said near-end noises. While, to low rpm engine noise signal characterized with the low rpm engine's relatively large frequency change over time, the LMS methods [7] and [8] based IAFANC communication systems proven to be unable to quickly and effectively fetch the very changeable signals, consequently, attenuate noise far less compared to the formers.

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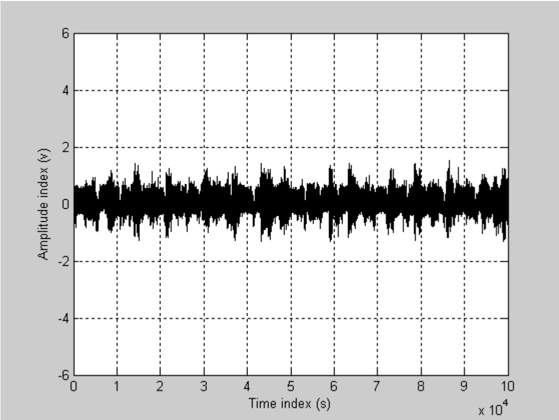
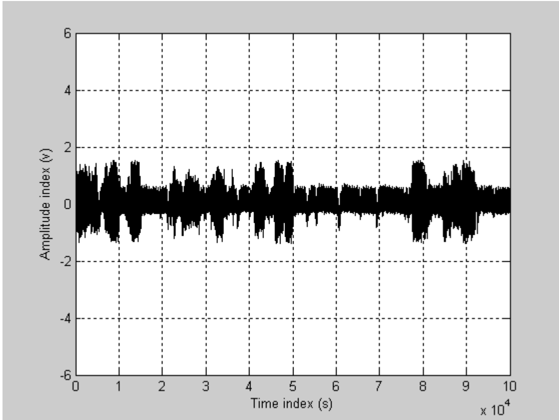
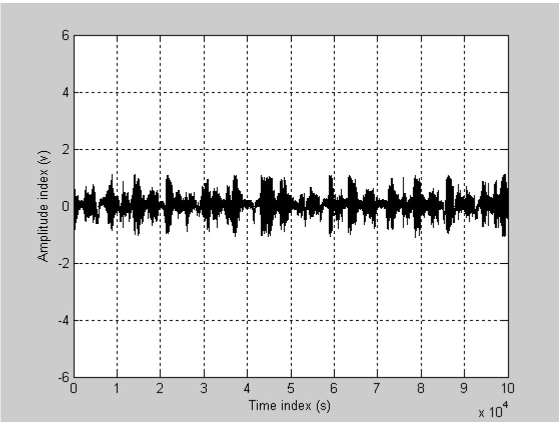


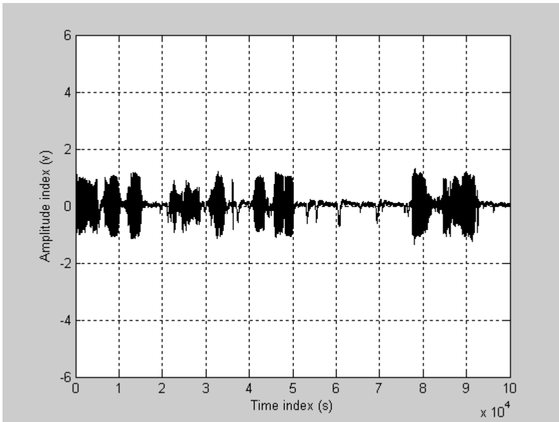
a. Far-end speech voice plus near-end noise



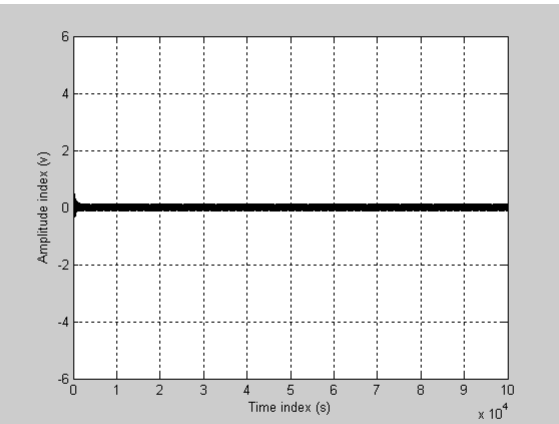
b. Near-end speech voice plus near-end noise



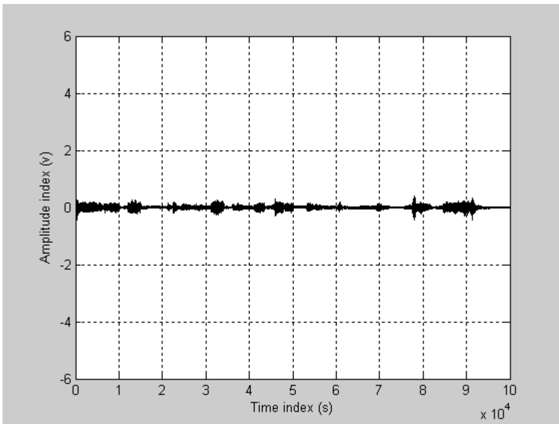
c. $e_1'(n)$



d. $e_2'(n)$

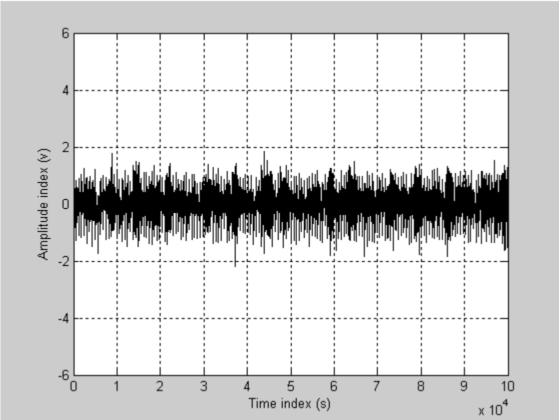


e. $e_1(n)$

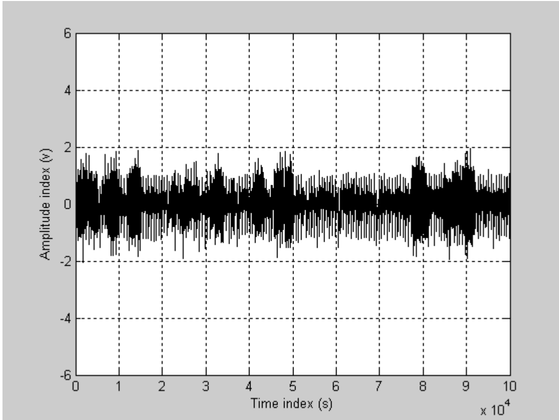


f. $e_2(n) - f_2(n)$

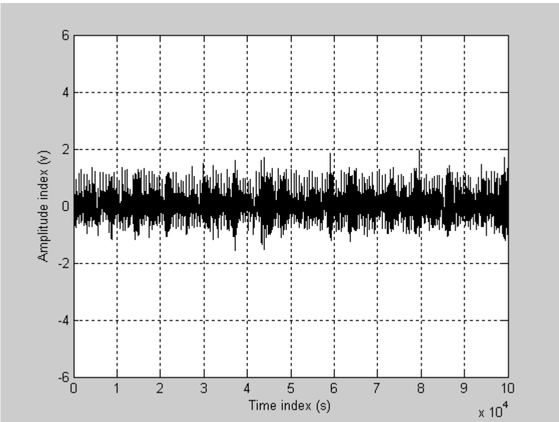
Fig. 3 The IAFANC system performance appraisal wave chart of near-end noise in form of sine noise, when SNR = 0dB



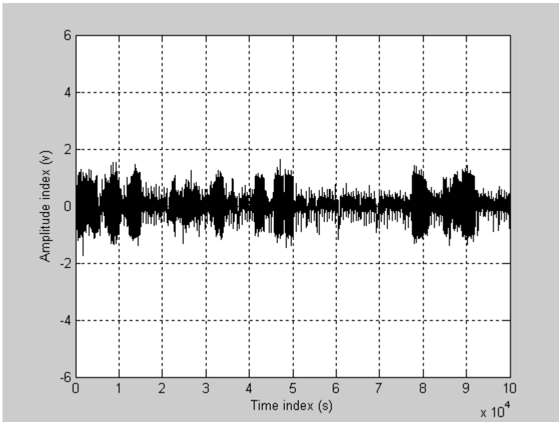
a. Far-end speech voice plus near-end Noise



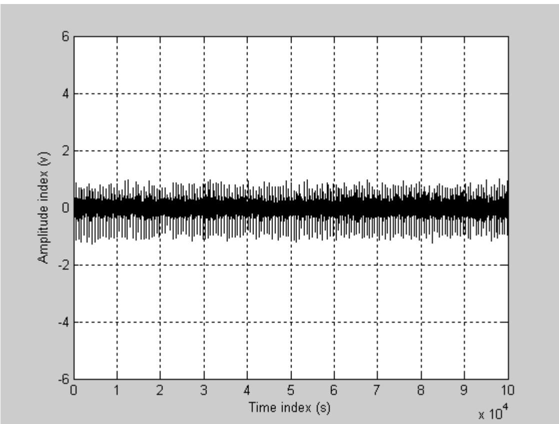
b. Near-end speech voice plus near-end noise



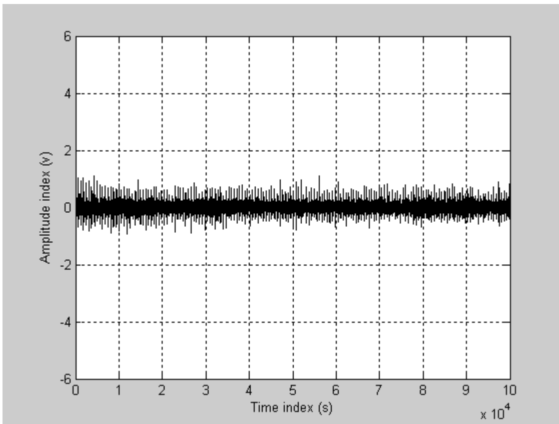
c. $e_1'(n)$



d. $e_2'(n)$

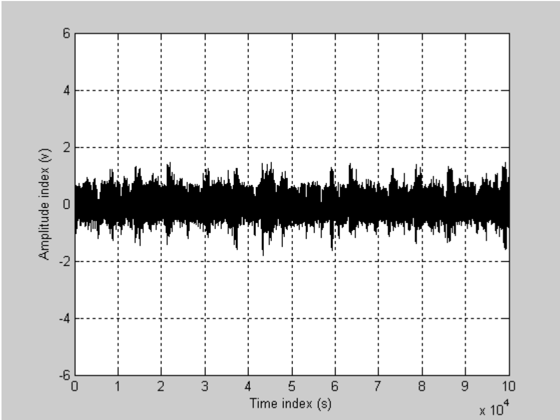


e. $e_1(n)$

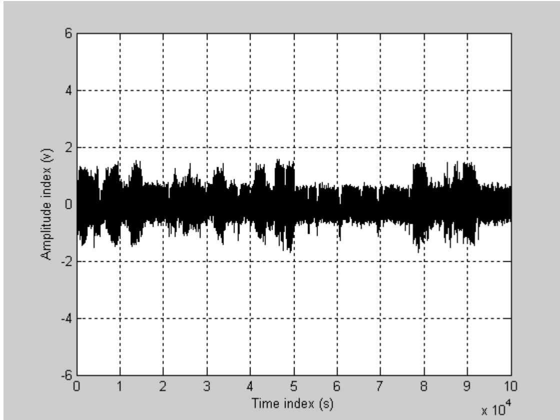


f. $e_2(n) - f_2(n)$

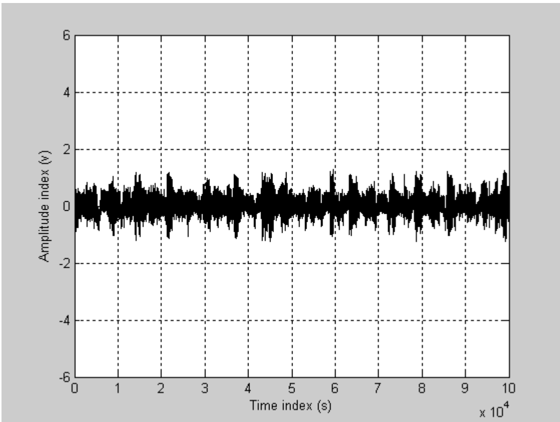
Fig. 4 The IAFANC system performance appraisal wave chart of near-end noise from engine low speed operation noise, when SNR = 0dB



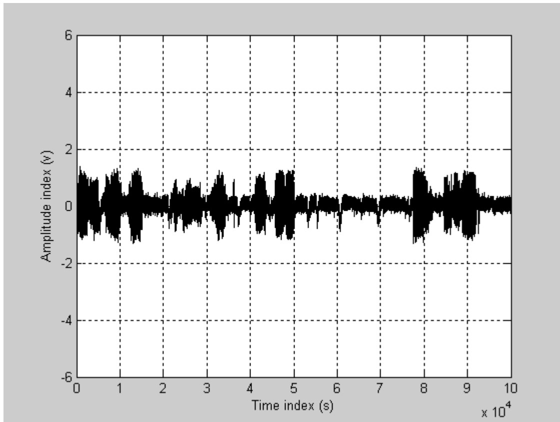
a. Far-end speech voice plus near-end noise



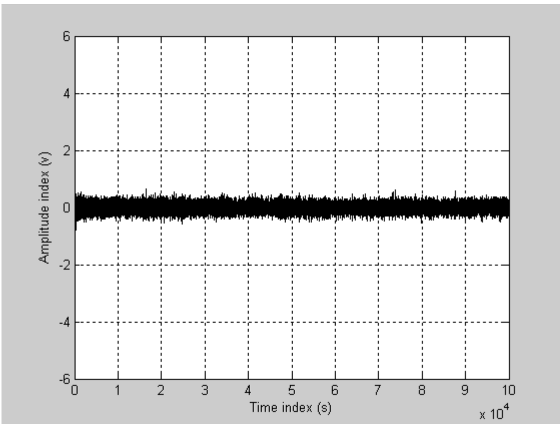
b. Near-end speech voice plus near-end noise



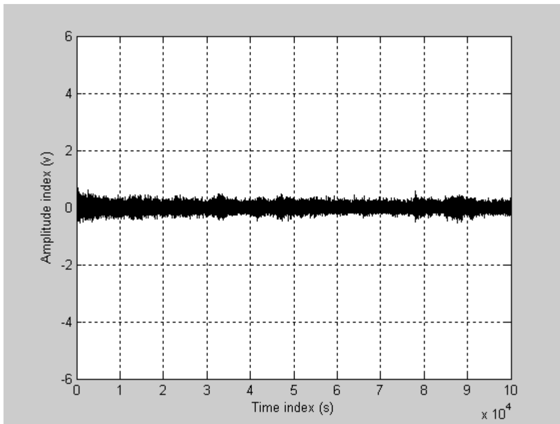
c. $e_1'(n)$



d. $e_2'(n)$

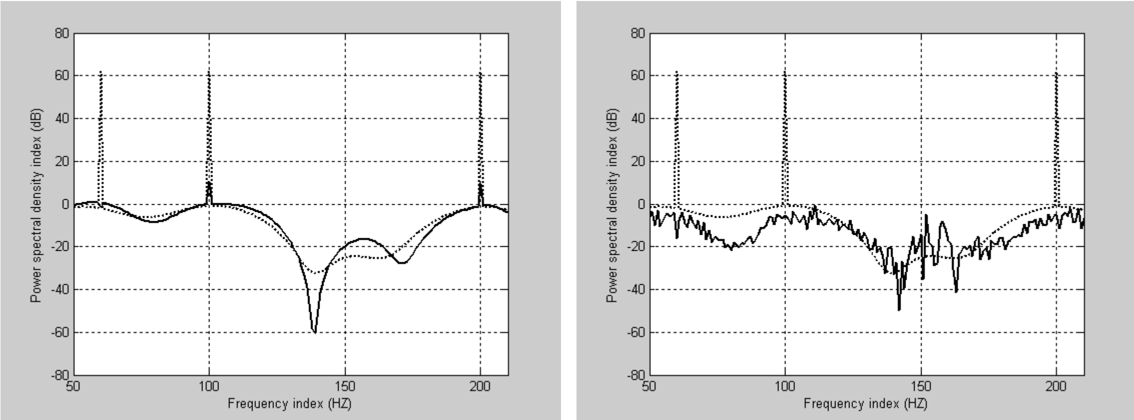


e. $e_1(n)$



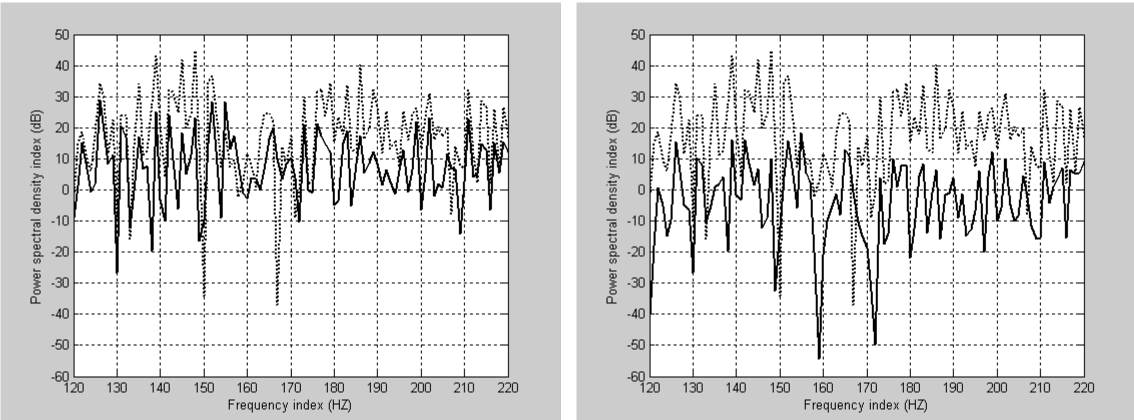
f. $e_2(n) - f_2(n)$

Fig. 5 The IAFANC system performance appraisal wave chart of near-end noise from engine high speed operation noise, when SNR = 0dB



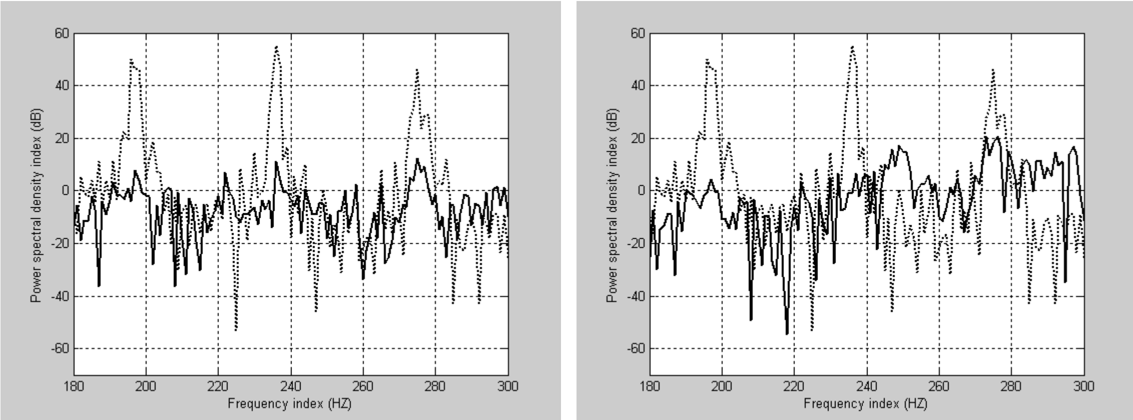
a. R(dB) in receiving far-end speech voice b. R(dB) in sending near-end speech voice

Fig 6. The frequency spectrum analysis chart of reduction in receiving & sending far-end speech voice R(dB) (noise reduction) and of near-end noise in form of sine noises respectively, where solid line indicates R(dB) while dot line indicates sine noise, when SNR = -10dB



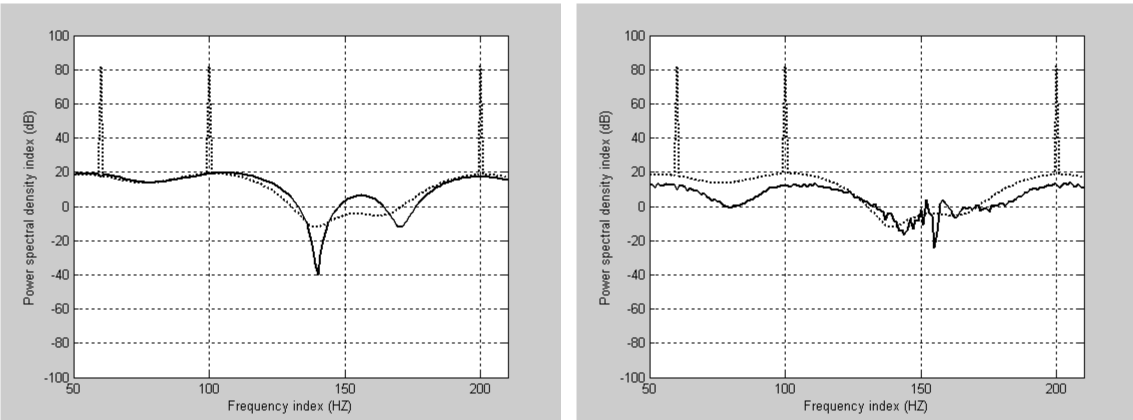
a. R(dB) in receiving far-end speech voice b. R(dB) in sending near-end speech voice

Fig 7. The frequency spectrum analysis chart of reduction in receiving & sending far-end speech voice R(dB) and of near-end noise in form of engine low speed operation noise, where solid line indicates R(dB) while dot line indicates engine low speed operation noise, when SNR = -10dB



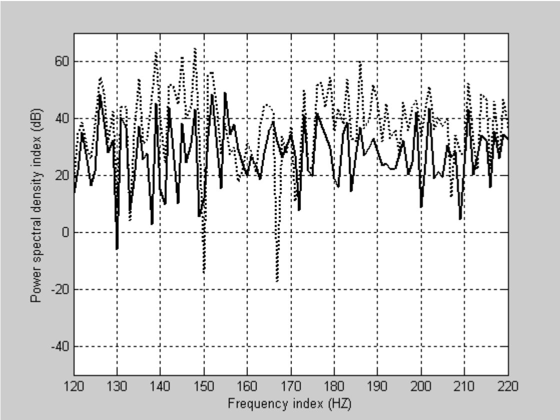
a. R(dB) in receiving far-end speech voice **b. R(dB) in sending near-end speech voice**

Fig 8. The frequency spectrum analysis chart of reduction in receiving & sending far-end speech voice R(dB) (noise reduction)and of near-end noise in form of engine high speed operation noise, where solid line indicates R(dB) while dot line indicates engine high speed operation noise, when SNR = -10dB

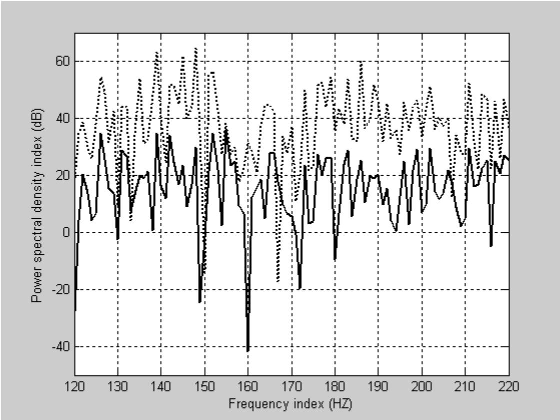


a. R(dB) in receiving far-end speech voice **b. R(dB) in sending near-end speech voice**

Fig 9. The frequency spectrum analysis chart of Reduction in receiving & sending far-end speech voice R(dB) and of near-end noise in form of sine noises respectively, where solid line indicates R(dB) while dot line indicates sine noise, when SNR = -20dB

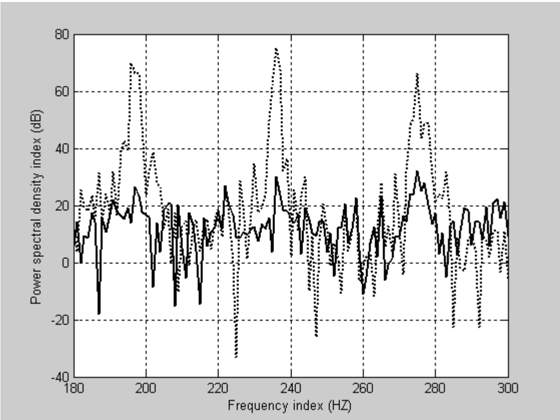


a. R(dB) in receiving far-end speech voice

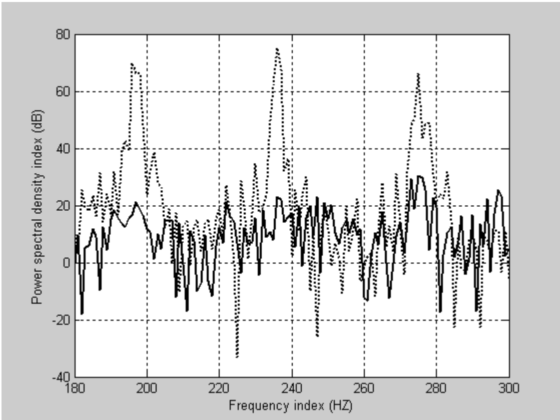


b. R(dB) in sending near-end speech voice

Fig 10. The frequency spectrum analysis chart of reduction in receiving & sending far-end speech voice R(dB) (noise reduction) and of near-end noise in form of engine low speed operation noise, where solid line indicates R(dB) while dot line indicates engine low speed operation noise, when SNR = -20dB



a. R(dB) in receiving far-end speech voice



b. R(dB) in sending near-end speech voice

Fig 11. The frequency spectrum analysis chart of reduction in receiving & sending far-end speech voice R(dB) and of near-end noise in form of engine high speed operation noise, where solid line indicates R(dB) while dot line indicates engine high speed operation noise, when SNR = -20dB