ABSTRACT

Enhanced Multi-channel Active Noise Control System and Real-Time Implementation for Infant Incubator

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The active noise control (ANC) system is implemented in the infant incubator and confirmed to improve environment effectively for infants. However, the traditional ANC systems cannot be applied directly to incubators without modification due to complicated noise models in the neonatal intensive care unit (NICU). It has multiple unexpected noise sources both inside and outside the incubators. The thesis proposed a system which combines the feed-forward and feedback ANC systems with multiple reference microphones, generates a larger zone of silence, offers deeper noise attenuation, and operates in complex noise environments, i.e. multiple unknown various noise sources. Moreover, the on-line secondary-path modeling algorithm is also designed and implemented to solve the time-varying secondary-path problem.

This thesis implemented the enhanced active noise system for infant incubators with noise source detection for improving the ANC system's performance. After detecting the noise source direction, selecting the relative closer reference microphones and preprocessing the received signal can solve the unknown noise source problem. The hybrid ANC algorithm can cancel both inside and outside noises for incubator. In order to track the time-varying secondary path, on-line

secondary-path training is also designed and implemented. In infant incubator application, a single-channel hybrid active noise control (MHANC) system is implemented for such hospital noise environment. A real time MHANC system implementation is being executed with a TMS6713 quad processor board from the HIRATSUKA Engineering Co., Ltd.

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ENHANCED MULTI-CHANNEL ACTIVE NOISE CONTROL SYSTEM AND REAL-TIME IMPLEMENTATION FOR INFANCT INCUBATOR

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CHAPTER 1

INRODUCTION OF ANC SYSTEMS

1.1 Introduction

Recently, noise problems have become more and more severe since industrial development. Noise does not only come from nature, such as snoring husband or loud people but also from equipment such as engines, fans, transformers and compressors which are common in our normal life. People always want to avoid the undesired noise and generate many methods to reduce them. The traditional noise control sets blockers in noise propagation path by passive methods, which includes using earplugs, barriers and silences to reduce the noise. Although these passive ways are effective in a high frequency range, they are expensive, huge and ineffective for low-frequency noise components.

The active noise control (ANC) involves an electroacoustic or electromechanical system that cancels the primary (undesired) noise based on the idea of superposition [1, 2]. Generating an anti-noise with same amplitude and opposite phase is combined with primary noise; therefore, they will be cancelled by each other. The design of acoustic of ANC utilizing a microphone and an electronically driven loudspeaker to generate a canceling sound was first proposed in 1936 by Lueg [3]. The ANC technique has been kept developing in the past decades. It depends on the development of the digital signal processing (DSP) system which can analyze and process the interested signal adaptively. Because ANC system is based on an adaptive filter which can adjust its coefficients to minimize an error signal, the general adaptive filter uses the least mean square

(LMS) algorithm. The application of ANC system applies LMS algorithm to process digitized noise signal.

1.2 Current Application

ANC system has many advantages in noise control area, such as: low cost, smaller size, and capability to cancel low-frequency noise. It can be implemented in healthcare equipment, appliances, industry and transportation. ANC system is set up in infant incubators for reducing noise from medical equipment. The system generates a quiet environment for newborn babies and provides other functions such as appeasing babies and crying recognition [4]. In appliances area, ANC system can be implemented for reducing noise from air-conditioning ducts, kitchen exhaust fans and washing machines and so on. Engine noise cannot be avoided in industrial and transportation field. Feed back ANC system has a good performance to cancel the regularity noise.

The thesis is organized as follows: Chapter 1 introduces popular ANC system structures with the classical least mean square (LMS) algorithm. Chapter 2 studies the simulation results which are based on real incubator noise. Then, Chapter 3 investigates multi-channel ANC application for infant incubators. The reference microphone detection will be discussed in this chapter. The determination of reference microphone or combination of reference microphone is also presented and analyzed in this chapter. The multiple-channel FXLMS algorithm will be used in this system. Chapter 4 includes implementation of the enhanced ANC system for infant incubators based on TMS32C6713 DSK platform. Summary of this thesis is at the end of this thesis.

1.3 Introduction of Feed-Forward ANC System

The broad-band ANC systems have a reference micro-phone, a louder speaker, and an error microphone. Figure 1-1 shows its structure. Reference microphones near the noise source for picking up reference input which is processed by ANC system. ANC system generates an anti-noise signal which is played by louder speaker. The unwanted noise will be canceled around the error micro-phone. Error microphone is used to monitor the performance of ANC system. The objective of this system is to minimize the sound picked up by the error micro-phone. In this application, we assumed that the reference microphone does not pick up the anti-noise which may affect reference signal for updating the adaptive filer. Feedback neutralization and adaptive IIR filter give some solution for this kind of problem [1].

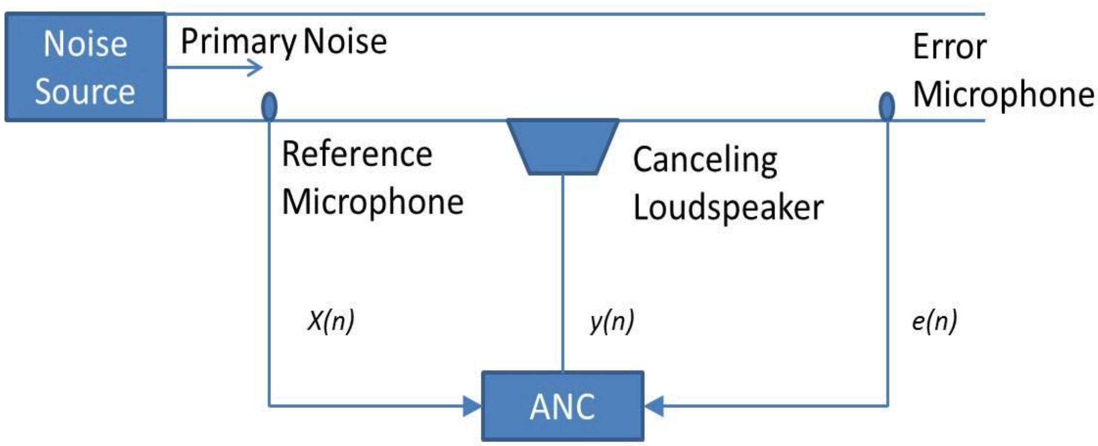

Fig.1-1 The diagram of broad-band feed-forward ANC system

The single-channel narrow-band feed-forward ANC System is used to cancel the periodical noise. In industrial and transportation field, many noises, which come from engine compressors, motors, and fans, are periodical signals. We can get driving information of mechanical motion directly by using an appropriate sensor. The fundamental frequency and all the harmonics of primary noise are included in driving information. Figure 1-2 is the system of the narrow-band feed-forward ANC system.

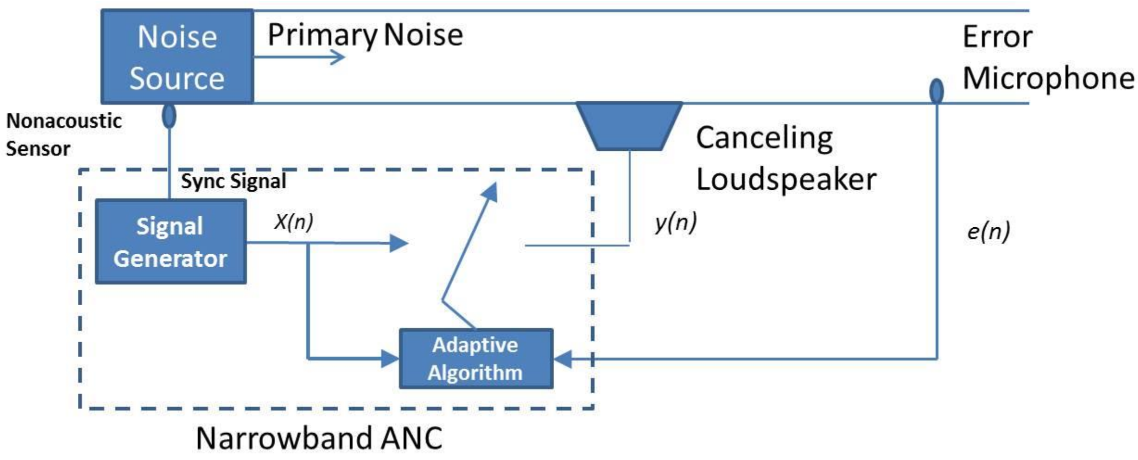

Fig.1-2 The diagram of narrow-band feed-forward ANC system

Comparing Fig 1-1 and Fig 1-2, the difference between the narrow-band ANC systems and the broad-band ANC system is the reference input signal. The narrow-band ANC system uses a signal generator to make a reference signal. For broad-band ANC system, the input signal is sampled by acoustic sensor. This technique is only effective for periodic noise since the fundamental frequency component is the reference information available. However, narrow-band feed-forward ANC system can be widely used since it has some advantages: 1) the input signal can be modeled by designing the harmonic reference signal; 2) it solved the acoustic feedback problem; 3) the filter order amount will reduce since it just has to model the plant transfer function over frequencies in range around the harmonic tones [1, 2].

Generally, there are two type of reference signals are used in narrow-band ANC systems: 1) an impulse train with a period equal to the inverse of fundamental frequency for the periodic noise; 2) the sinusoidal waves that have the same frequency components as the fundamental driving frequency. The first technique is called the waveform synthesis method [5]. The second technique is associated with adaptive notch filter [6].

1.4 Introduction of Feedback ANC System

Feedback ANC system is only based on the adaptive filter output and error signal. The following Figure 1-3 shows the system structure of feedback ANC system. The feedback ANC system does not have any reference signal. The error signal and the anti-noise signal are used to reconstruct the reference signal. Therefore, the feedback ANC system contains one error microphone and one loudspeaker. Because feedback ANC system predicts the information of the future values of the primary noise, it can be combined with feed-forward ANC system for reducing the predictable components of the primary noise that are not observed by reference microphone in hybrid ANC systems.

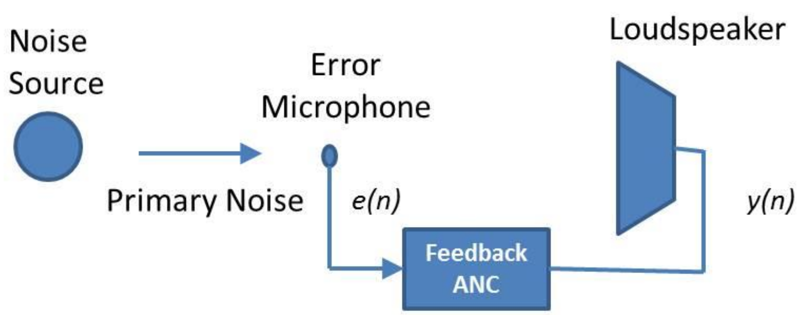

Figure.1-3 The diagram of broad-band feed-forward ANC system

1.5 Introduction of Hybrid ANC System (HANC)

The hybrid ANC system combines feed-forward ANC system and feedback ANC system (Figure 1-4). For feed-forward ANC system, the reference microphone picks up the primary noise to be canceled and the error microphone gets residual noise signal to update the filter and make ANC system work. Feedback ANC system uses an error microphone and cancels the predictable noise components of primary noise. In Chapter 4, we discuss more details about hybrid ANC system.

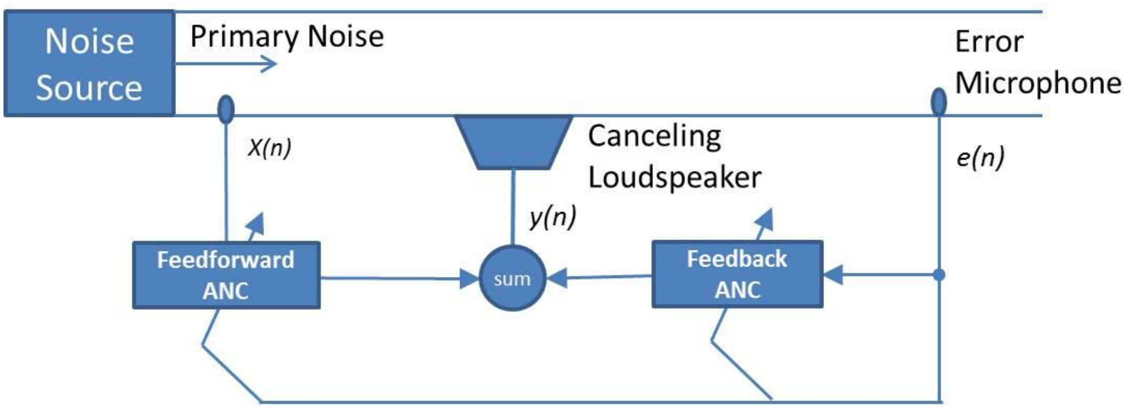

Figure.1-4 The diagram of hybrid ANC system

1.6 Introduction of Multi-channel ANC System (MCANC)

The multi-channel ANC system can be applied to build a large quiet dimension. Considering the noise sources come from different directions and positions, multi-channel ANC systems need several reference sensors and error sensors. Some of the best known applications are the control exhaust “boom” noise in automobiles, earth-moving machines, and the control of propeller-induced noise in flight cabin interiors [1]. Because the whole system should be causal system, the distance from the primary to the secondary source must be less than a quarter wavelength of the highest frequency [2]. Figure 1-5 shows the structure of a multiple-channel ANC system.

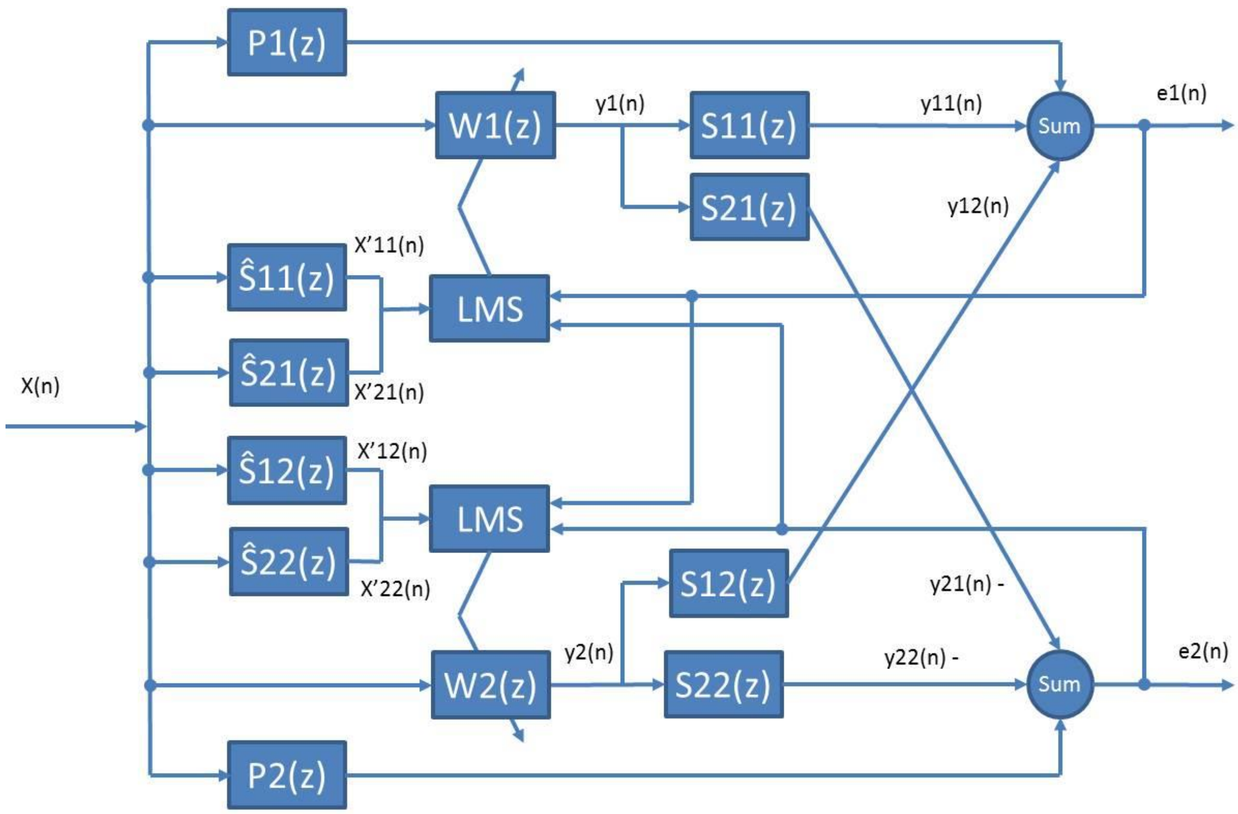

Figure.1-5 The diagram of 1×2×2 multi-channel feed-forward ANC system

1.7 Summary

This chapter simply introduced some applications of the active noise control, such as the popular system structure of active noise control and the basic principles. The development of multi-channel can improve the performance of ANC system.

CHAPTER 2

SECONDARY-PATH MODELING OF ACTIVE NOISE SYSTEM

2.1 Introduction of LMS Algorithm

Active noise control is described in an adaptive system structure which is defined in Figure 2-1. An adaptive algorithm can track the change of input signal, and adjust filter's parameters automatically based on certain objectives. In a real application, the plant is dynamic; adaptive algorithm has a task for continuously modeling the plant. In Figure 2-1, the objective of the adaptive filter $W(z)$ is to minimize the error signal $e(n)$. The noise source $x(n)$ across the unknown plant $P(z)$ becomes the primary noise $d(n)$. The digital adaptive filter's output $y(n)$ is driven by the input $x(n)$; the error sensor picks up the error signal $e(n)$ and sends it to the ANC system as feedback for updating the parameters of $W(z)$. The system achieves the minimum mean square of error signal $e(n)$. So, the system is updating the adaptive filer parameters dynamically to minimize the mean square of the error signal [7].

Finite impulse response (FIR) digital filter is commonly used in active noise control application because the FIR filter structure is always stable and has linear phase [1, 2]. In Figure 2-1, the input signal is described by a vector $x(n)=[x(n),x(n-1),\ldots,x(n-N+1)]^T$. The output signal is calculated by $y(n) = \sum_{i=0}^{N-1} w(i)x(n-i)$. The error signal is calculated by $e(n) = d(n) - y(n)$.

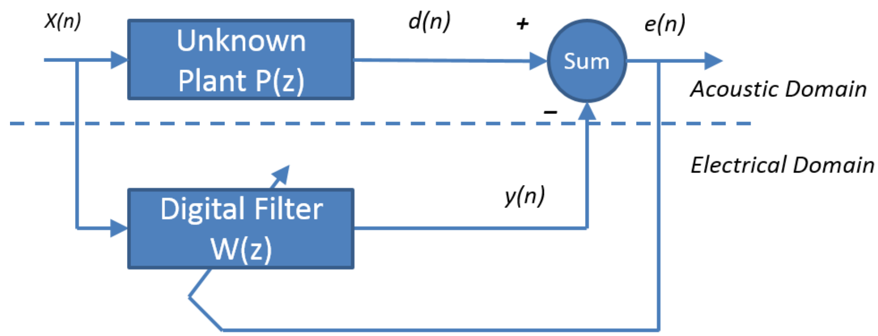

Fig 2-1 The system diagram of adaptive filter

Least mean squares (LMS) algorithm is one of the most popular algorithms for adaptive filter. The adaptive filter model mimics a desired filter by finding coefficients which produce the least mean squares of error signal ($E[e^2(n)]$).

The derivation of LMS algorithm is shown in the following [7]:

$$\mathrm{E}[\mathrm{e}^2(\mathrm{n})] = \mathrm{E}\left[\left(\mathrm{d}(\mathrm{n}) - \mathrm{y}(\mathrm{n})\right)^2\right] = \mathrm{E}[\mathrm{d}^2(\mathrm{n}) - 2\mathrm{d}(\mathrm{n})\mathrm{y}(\mathrm{n}) + \mathrm{y}^2(\mathrm{n})]$$

$$= \mathrm{E}[\mathrm{d}^2(\mathrm{n})] - 2\mathrm{E}[\mathrm{d}(\mathrm{n})\mathbf{x}^{\mathbf{T}}(\mathbf{n})\mathbf{w}(\mathbf{n})] + \mathrm{E}[\mathbf{w}^{\mathbf{T}}(\mathbf{n})\mathbf{x}(\mathbf{n})\mathbf{x}^{\mathbf{T}}(\mathbf{n})\mathbf{w}(\mathbf{n})]$$

$$= \mathrm{E}[\mathrm{d}^2(\mathrm{n})] - \mathbf{2P^T w(n)} + \mathbf{w^T(n) R w(n)} \quad \text{(2-1)}$$

where P=E[d(n)x(n)] is the cross correlation vector of desired signal and the input signal, and $\mathrm{R} = \mathrm{E}[\mathrm{x}(\mathrm{n})\mathbf{x}^{\mathbf{T}}(\mathbf{n})]$ is the autocorrelation matrix of the input signal. The vector of $\boldsymbol{w}(n)$ is the adaptive filter coefficient is related to the mean square error $E[e^2(n)]$. From the equation (2-1), the required $\boldsymbol{w}(n)$ is realizes the optimal transfer function as following:

$$-2\mathrm{P} + 2\mathrm{Rw}(\mathrm{n}) = 0$$

$$\mathrm{w}^0 = \mathrm{R}^{-1}\mathrm{P} \quad \text{(2-2)}$$

The steepest descent algorithm is described in the following equations:

$$w(n+1) = w(n) - \frac{\mu}{2}\nabla(E(e^2(n))) \text{ (2-3)}$$

$$w(n+1) = w(n) + \mu(P - Rw(n)) \text{ (2-4)}$$

Here μ is the step size of the algorithm. One can control the convergence speed by adjusting the step size of the algorithm. Assume the instantaneous squared error $e^2(n)$ is a reasonable estimate of the mean square error [7]. Based on this assumption, the previous equations can be modified to the following forms:

$$\nabla\left(E\left(e^2(n)\right)\right) \approx \nabla\left(e^2(n)\right)$$

$$e(n) = d(n) - \boldsymbol{w^T(n)x(n)}$$

$$\nabla\left(e(n)\right) = -x(n)$$

$$w(n+1) = w(n) + \mu e(n)x(n) \text{ (2-5)}$$

For normalized least mean square (NLMS), μ is the inverse power estimate of the reference signal$|x^2(n)|$. For LMS and NLMS, both of them enjoy low computation complexity for updating the adaptive filter coefficient.

2.2 Filtered-XLMS Algorithm

In real ANC application, the source noise and anti-noise represent the acoustic superposition in the acoustic channel from the loudspeaker to the error microphone, where the primary noise $x(n)$ is combined with the anti-noise $y(n)$ generated by the adaptive filter. Therefore, it needs to consider that the secondary-path transfer function $S(z)$ from $y(n)$ to $e(n)$. It includes the digital-to-analog (D/A) converter, reconstruction filter, power amplifier, loudspeaker, acoustic path from loudspeaker to error microphone and etc. So, Figure 2-1 represents the actual system.

From Figure 2-1 the Z-transform of the error signal is

$$\mathrm{E(z) = [P(z) - S(z)W(z)]X(z)} \qquad (2\text{-}6)$$

The residual error is limited by the coherence of the reference signal $x(n)$. We assume that the residual error is equal to zero after the adaptive filter converge. Then, equation (2-6) can be simplified to

$$\mathrm{W(z) = \frac{P(z)}{S(z)}}. \qquad (2\text{-}7)$$

According to Equation (2-7), the adaptive filter has to model the primary path $P(z)$ and the inverse of the secondary-path $S(z)$. But the system needs a high-order FIR filter to approach the rational function of $1/S(z)$, and the system performance largely depends by the secondary-path transfer function $S(z)$ [1, 2].

The secondary-path transfer function $S(z)$ is used in the standard LMS algorithm. The error signal is not correlated with the input reference signal, due to the presence of $S(z)$. Kuo and Morgan [2] gave two solutions for this problem. One is to place an inverse filter $1/S(z)$ for removing its effect. The second solution is to place an identical filter in the path from reference signal to the LMS algorithm, which is called filtered-X LMS (FXLMS) algorithm [1, 2]. The system is modified to the following diagram in Figure 2-2.

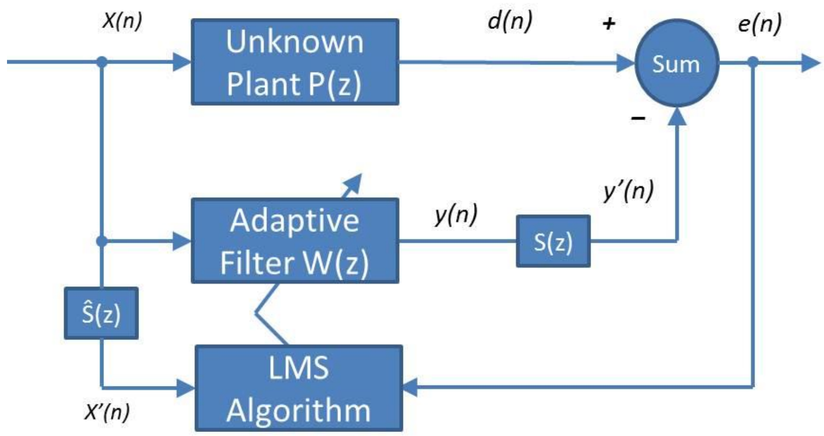

Fig 2-2 The system diagram of FXLMS algorithm

The derivation of the FXLMS algorithm is:

$$e(n) = d(n) - s(n) * [\boldsymbol{w^T}\,(\boldsymbol{n})\boldsymbol{x}(\boldsymbol{n})] \quad \text{(2-8)}$$

where n is the time index, $s(n)$ is the impulse response of secondary path, $S(z)$ denotes linear convolution, and $w(n)$ is coefficient of FIR filter.

$$\left(E\left(e^2(n)\right)\right) \approx \nabla\left(e^2(n)\right)$$

$$\nabla\left(e(n)\right) = -\boldsymbol{x}(\boldsymbol{n}) * s(n)$$

$$w(n+1) = w(n) + \mu e(n)(\boldsymbol{x}(\boldsymbol{n}) * s(n)$$

$$w(n+1) = w(n) + \mu e(n)\boldsymbol{x'}(\boldsymbol{n}) \quad \text{(2-9)}$$

$$\boldsymbol{x}'(\boldsymbol{n}) = \boldsymbol{x}(\boldsymbol{n}) * s(n)$$

In practical ANC application, $S(z)$ is always unknown and must be estimated by an additional filter $\hat{S}(z)$. So the reference signal passes the estimated secondary-path to get the filtered reference signal, which updates the coefficients of adaptive filter.

2.3 Secondary-path Modeling Simulation

There are two methods to model the second path transfer function: off-line and on-line modeling. Off-line modeling can be estimated by training sound (normally white noise) passing through the loudspeaker to error microphone with LMS algorithm while primary noise does not exist (Figure 2-3). After the error signal converges, the filler coefficients can be a good estimation of the secondary-path transfer function. On-line modeling is applied to model secondary-path in a noisy environment, which means training signal may be mixed with background noise. It requires modeling secondary-path and canceling the unwanted noise simultaneously.

2.3.1 Off-Line Modeling Technique

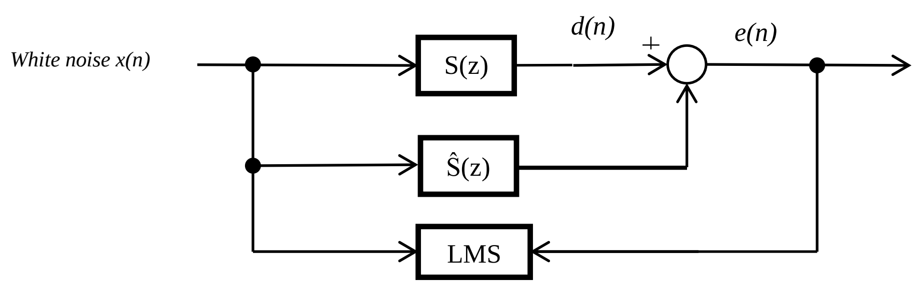

Fig 2-3 ANC system off-line training modeling with white noise

Figure 2-4 shows the performance of secondary-path modeling by using the white noise off-line training. Figure 2-4(left) shows the time domain of $d(n)$ and $e(n)$. Figure 2-4 (right) shows the frequency response of $S(z)$ and $\hat{S}(z)$. From Figure 2-4, since $\hat{S}(z)$ overlaps with $S(z)$, the secondary-path is modeled by off-line training system very well.

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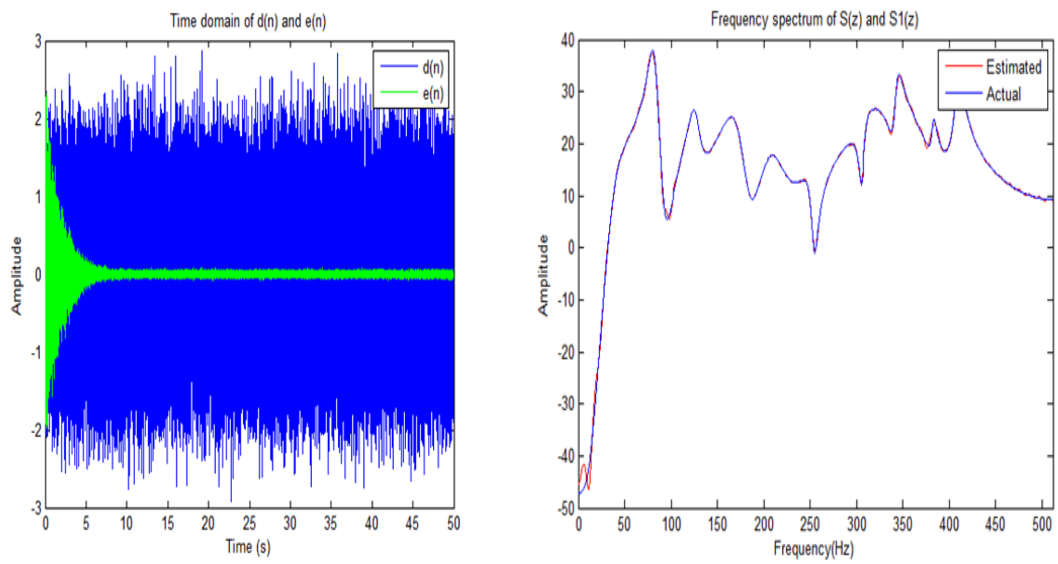

Fig 2-4 Time domain of error signal and desired signal (left) and secondary-path modeling result (right)

2.3.2 On-Line Modeling Technique

On-line modeling technique for active noise control system is implemented in the environment produced by different extreme situations such as un-stop power equipment. Based on the feed-forward ANC system, it implements the three blocks of diagram, as shown in Figure 2-5, 2-7, 4-7. One can also use audio source instead of the random noise to train second-path modeling. However, the interference of audio source affects the cancellation of the unwanted noise. We try to implement the three block diagrams to reduce the interference and cancel the undesired noise. We use another broad-band, uncorrelated noise as training signal, and the estimated secondary-path is closer to the real one [4].

We implement Widro and Starrns's algorithm to do on-line secondary-path modeling [2] and obtain the simulation result. If it trains the second path very well, spectra of *S(*z*)* and Ŝ(z*)* approximately overlap. Since original noise does not suit to be training signal, it cannot model secondary paths effectively. Fig 2-5 shows the block diagram. Fig 2-6 shows the frequency response of *S*(*z*) and Ŝ(*z*). It is shown that this algorithm cannot provide good estimation of the secondary path.

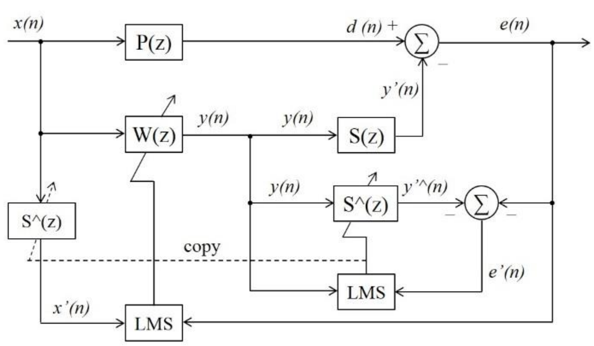

Figure 2-5 Block diagram of on-line secondary-path modeling [1]

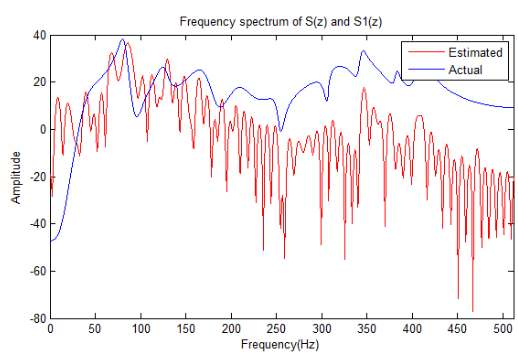

Figure 2-6 Frequency spectrum of *S*(*z*) and *Ŝ*(*z*) for on-line training

Audio-integrated ANC algorithm adds the white noise to on-line training system. Sampling frequency of neonatal intensive care unit (NICU) noise is 4000 Hz as the un-stop noise source. We expected that the interference of original noise can be reduced by using the on-line secondary-path modeling as shown in Figure 2-7. One can adjust value of α to control energy from the white noise.

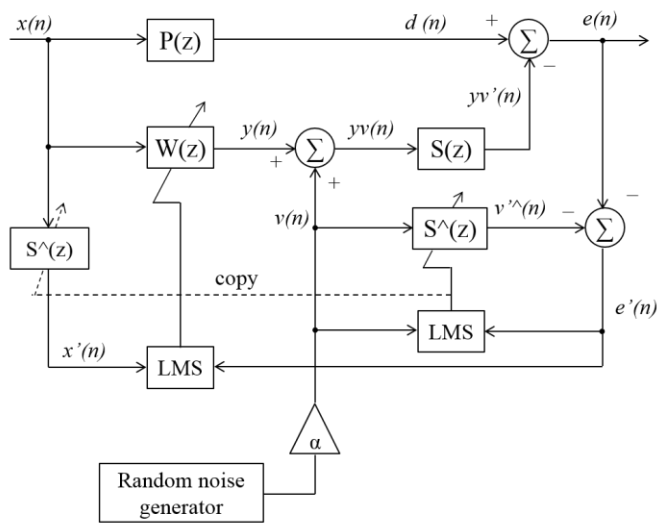

Figure 2-7 Block diagram of on-line secondary-path modeling using additive random noise

In this section, we adjust different α to obtain the performance of frequency spectrum of $S(z)$ and $\hat{S}(z)$. Let α equal 0.2, 0.5, and 0.8, 1 respectively. Figure 2-8 shows the frequency spectrum of $S(z)$ and $\hat{S}(z)$. The increased energy of white noise leads the coefficients close to the true value. When α equals 1, ANC system model secondary-path coefficients very well. However, the increased energy of white noise makes the convergence worse.

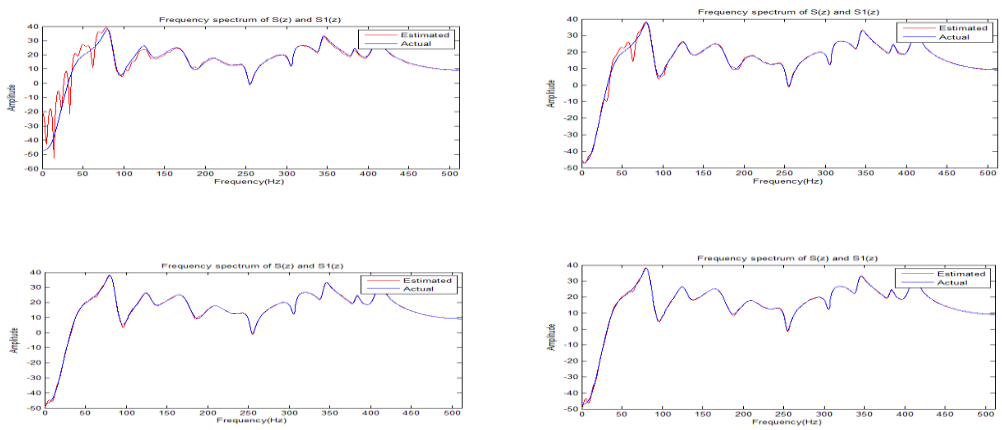

Figure 2-8 Frequency Spectrum of $S(z)$ and $\hat{S}(z)$ a=0.2,0.5,0.8 and 1

2.4 Summary

This chapter introduces the LMS algorithm and the filtered-X least mean square (FXLMS) algorithm. Several secondary-path modeling algorithms and their simulation results are presented. Implementing ANC system with on-line second-path modeling using random noise is studied. From the simulation results, we find that the performance of the algorithm is related with the energy of training signal. Generally, when energy of training signal is closer to the similar level of original noise, the modeling becomes better. However, the performance of ANC system becomes worse. Therefore, the energy of additive signal needs to be carefully chosen in the real-time experiment and real application.

CHAPTER 3

EFFECTS OF COMBINED REFERENCE FOR MULTIPLY REFERENCED ANC SYSTEM

The research for using multiple reference signal microphones in the multi-channel ANC system structure is studied in this chapter. Assume a typical multi-channel filtered-X least mean square (LMS) system has J reference sensors, K secondary sources and M error sensors: then we have a $J \times K \times M$ system. The multi-channel ANC system we discussed in the previous chapter uses only one reference microphone (J=1) to pick up a signal from one direction. However, when there are multiple unknown noise sources from various directions. The traditional multi-channel ANC system with one reference microphone cannot reduce the undesired noise effectively. The detection of the direction of noise source is discussed for improving the performance of active noise control in NICU [8]. For multi-channel ANC system with J reference microphones, each microphone picks up a reference signal $x_j(n)$. The canceling signals can be generated by processing the multiple reference signals simultaneously; we have a $J \times K \times M$ system. We can also pre-process the received multiple reference signals and obtain a combined reference signal; then we have a combined reference (CR) $J \times K \times M$ system. The objective of this chapter is to investigate the multi-reference ANC system performance while multiple noise sources from different directions exist.

3.1 1×2×2 ANC system

The 1×2×2 ANC system is included in the multiple-channel ANC system. The single-channel 1×1×1 ANC system can cancel the wide-band noise, but the quiet zone size is limited.

When we need larger quiet zone, the real system needs many loudspeakers and sensors to measure the noise direction and information; the multiple-channel ANC system becomes more complex. The control of complicated noise fields requires both the exploration and development of optimum strategies and construction of an adequate multi-channel controller. [9] The $1\times2\times2$ ANC system is a relatively simple model compared with other ANC systems that can be applied in infant incubator.

In this section, the $1\times2\times2$ broad-band feed-forward ANC system using FXLMS algorithm has been implemented. An example for broad-band feed-forward ANC system is illustrated in Figure 3-1. The reference microphone collects the audio signal to generate x(n). $P_1(z)$ and $P_2(z)$ are transfer functions of primary path from reference microphone to two error microphones. The residual signals $e_1(n)$ and $e_2(n)$ are picked up by the two error microphones which continuously collect the residual signals and send them to the system. The system uses two adaptive filters $W_1(z)$ and $W_2(z)$ to generate anti-noise signal $y_1(n)$ and $y_2(n)$ which is played by the two loudspeakers. According to the structure of the ANC system, we have four secondary-path transfer functions: $S_{11}(z)$, $S_{21}(z)$, $S_{12}(z)$ and $S_{22}(z)$. The $S_{11}(z)$ and $S_{21}(z)$ are the secondary paths from loudspeaker #1 to the error microphone #1 and #2 respectively.

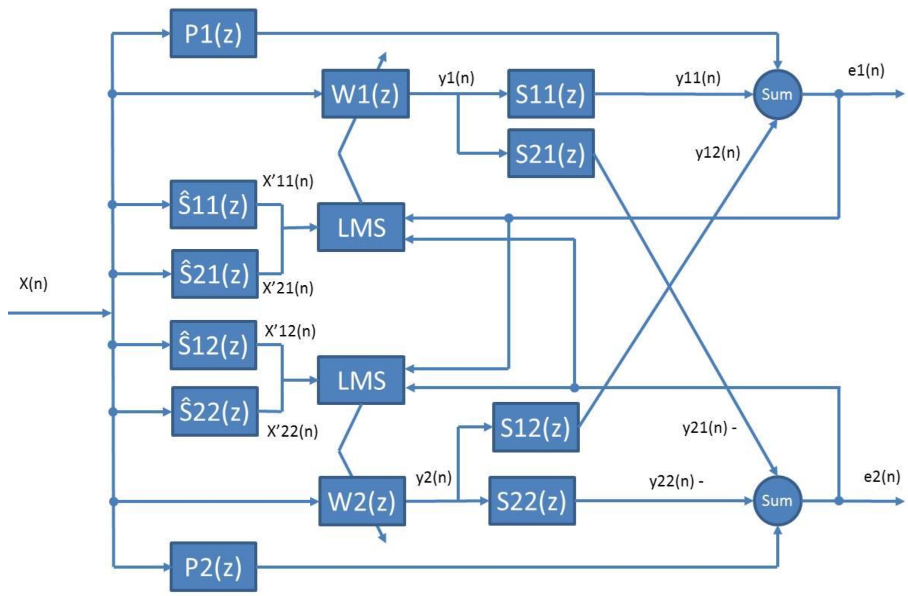

Figure 3-1 The 1×2×2 ANC system FXLMS algorithm block diagram

The 1×2×2 ANC system is similar to the single-channel broad-band feed-forward system using FXLMS algorithm; the algorithm requires the reference input signal to go through the estimated secondary-path to compensate the secondary-path effect. Normally, the white noise is used for off-line secondary-path modeling. In Chapter 4, different training signals are used for secondary-path training, such as stream sound, classic music or transformer noise, etc.

Figure 3-2 shows the secondary-path modeling result. According the result, estimated results fit well with the real secondary path. It proved that the adaptive filter $S_{ij}(z)$ coefficients are close to transform function of secondary path.

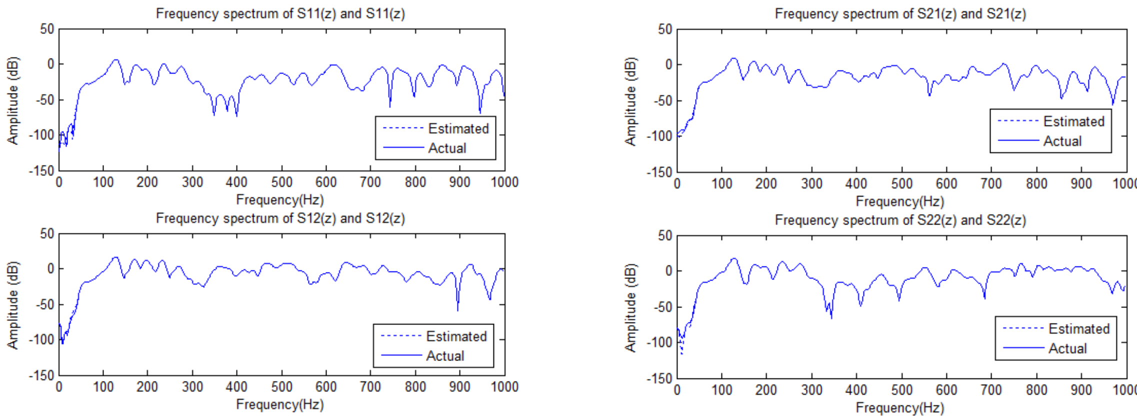

Figure 3-2 Frequency response of secondary path

The following step is $1\times2\times2$ feed-forward ANC system on-line training simulation whose data used is collected by TASCAM recorder. First, the single-frequency noise will be used to reference signal $x(n)$ for $1\times2\times2$ ANC system. The following Figure 3-3 shows the performance of simulation results for 200 Hz sin-wave cancellation. From the time domain of simulation result, the left-side and right-side sin-wav can be cancelled by the system. According to the performance of Figure 3-4, the average cancellation of left sides is around 31.28 dB and the right-side average cancellation is around 29.6dB. \

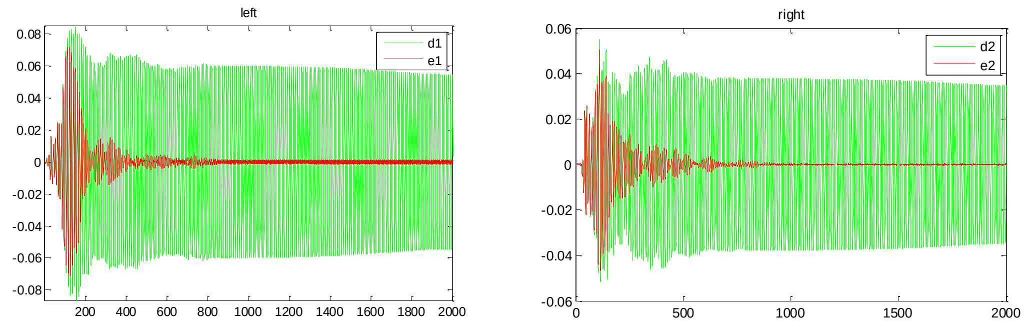

Figure 3-3 The time domain of reference signal and error signal

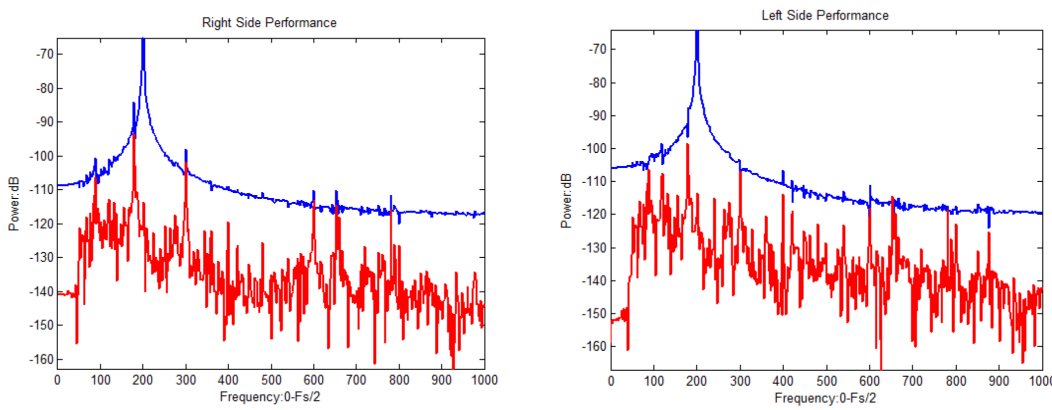

Figure 3-4 Frequency spectrum of 200 Hz sin-wav cancellation for $1 \times 2 \times 2$ ANC system

The neonatal intensive care unit (NICU) is a special intensive care unit for ill, low-birth-weight newborns or premature newborns [8]. NICU noise comes from the equipment noise and nurse activities noise. A large number of equipment has been implemented to detect the infant's heart rate, blood pressure and oxygen saturation. The frequency alarm alerts the nurse to take care of the dangerous baby. Table 3.1 shows the NICU noise sources come from equipment when they are working or alarming and human activities. Figure 3-5 shows the magnitude spectrum of the recorded NICU noise. Through analysis of the NICU noise we found that the main frequency components distributes below 4000Hz as shown in Figure 3-6.

Table 3.1 Common NICU Noise Sources [10]

Noise Sources	Noise Levels (dB)
Pump alarms	60-78
Finger tapping on the incubator	70-95
Pulse oximeters	86
Bubbling water in respiratory tubing	62-87
Loud voices	>100

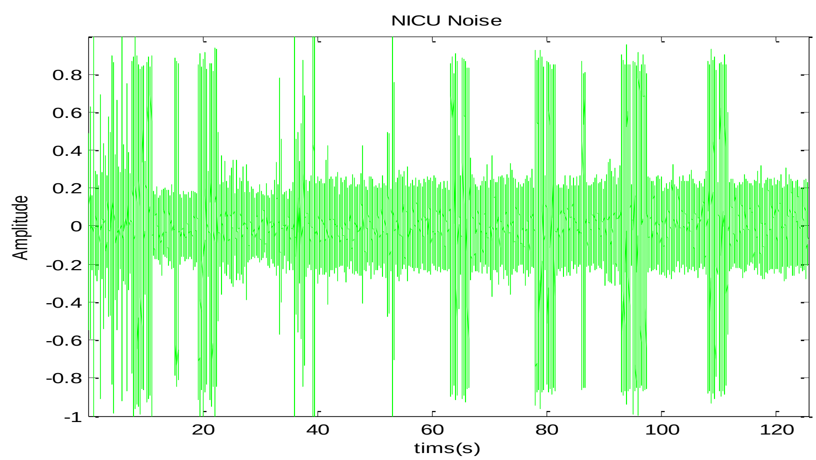

Figure 3-5 Time domain waveform of NICU noise

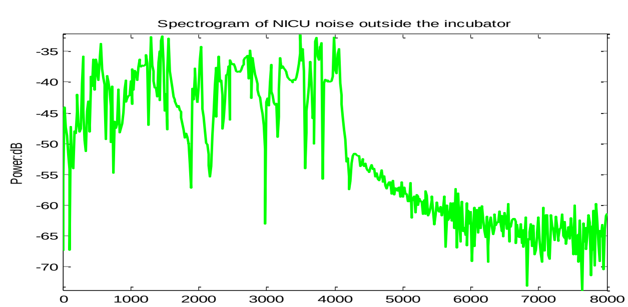

Figure 3-6 Frequency domain of NICU noise

The Frequency domain diagram represents the frequency components distribution range from 0 to 3800 Hz. If we cancel this part of noise components, the NICU noise will be canceled by 90%. Based on this idea, the NICU noise may be resampled from 16KHZ to 4000Hz and the signal put into multi-channel $1\times 2\times 2$ ANC system. Figure 3-7 shows the time domain waveform of NICU noise before and after ANC working.

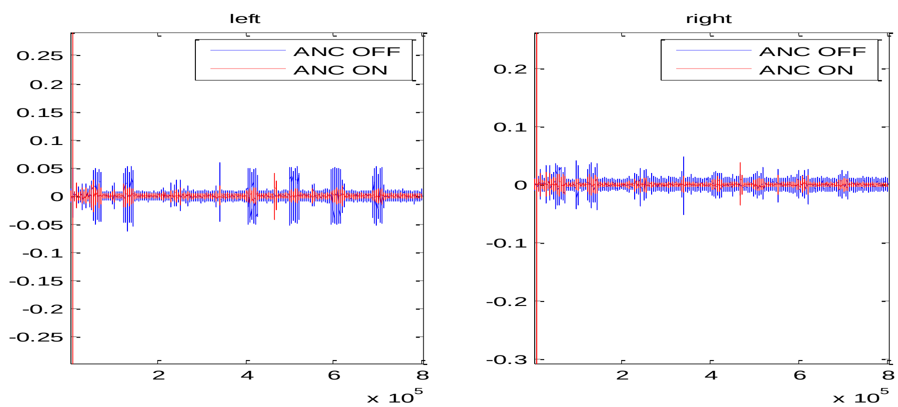

Figure 3-7 Time domain waveform of NICU noise for $1\times 2\times 2$ ANC system

The following Figure 3-8 shows the performance of simulation results for NICU noise cancellation. From the time domain of simulation result, the left-side and right-side sin-wav can be cancelled by the system. According to the performance of Figure 3-8, the average cancellation of left sides is around 6.4 dB and the right-side average cancellation is around 6.8 dB.

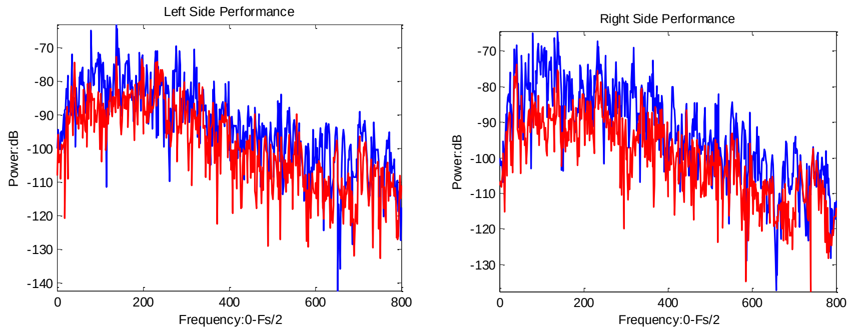

Figure 3-8 Frequency spectrum of NICU noise cancellation for $1\times2\times2$ ANC system

3.2 The Combined Reference $2\times2\times2$ ANC System (CR $2\times2\times2$ ANC)

This section presents the CR $2\times2\times2$ ANC system. The objective of this section is to investigate the performance of the multi-channel ANC with combined multiple-reference signal noise. The CR $2\times2\times2$ ANC system uses two microphones to pick up the noise source and generates two different reference signals (Figure 3-9). Through combination of reference signals, the system is intended to generate one proper reference signal, which includes most of the information of noise signal. The two secondary loudspeakers can be used to generate the anti-noise, which provided a quiet zone around baby's ears. The CR $2\times2\times2$ ANC system diagram is developed by basic $1\times2\times2$ ANC system diagram which was showed in Chapter 2. Assume the noise source is at position 1. The left microphone 1 is close to the noise source, and the right microphone 2 is far away from noise source, as shown in Figure 3-10.

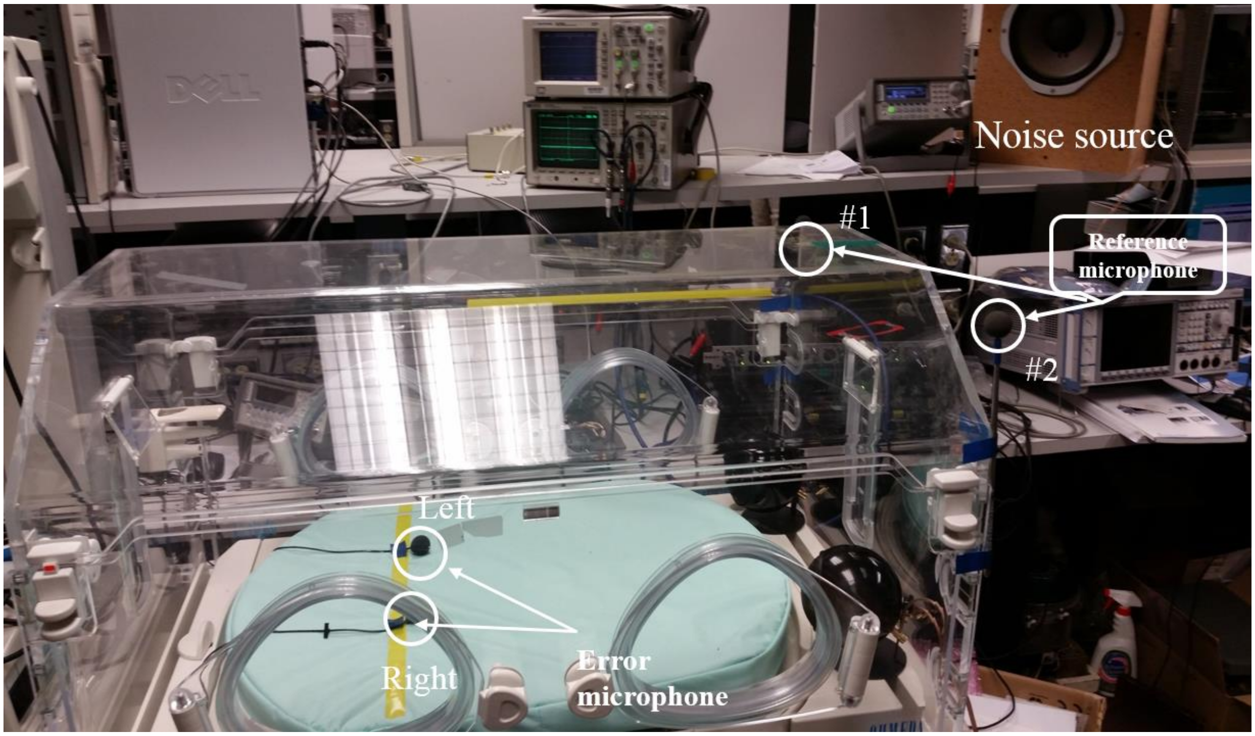

Figure 3-9 Two reference microphones for generating reference signals

Two reference signals are picked up as $x_1(n)$ and $x_2(n)$ by reference microphones #1 and #2. A new reference signal $x(n)$ is generated by the pre-processing $x_1(n)$ and $x_2(n)$. Assume the distances between noise source and reference microphones are d1 and d2. According to Figure 3-10, the noise source passes the acoustic paths T1 and T2. The distances from the noise source to reference microphones 1, 2 are d1 and d2, respectively. Therefore, the difference of distance $d = d2 - d1$ can be calculated. $\Delta t = d/c$.

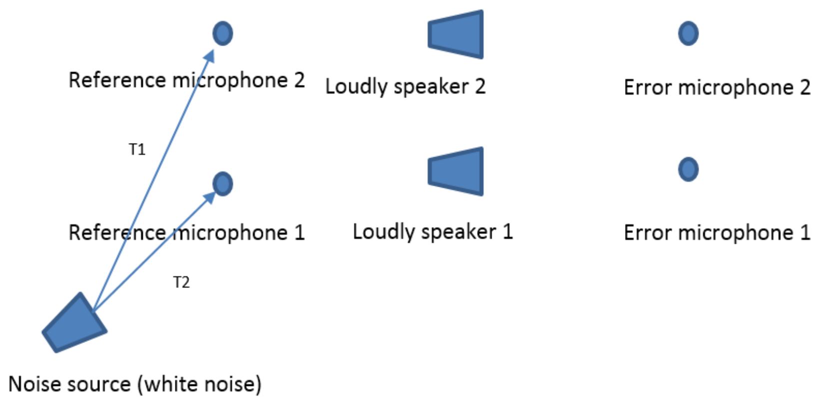

Figure 3-10 CR 2×2×2 ANC system position diagram

In the experiment, *d* is 1.71m. The delay time equals 0.0045s between the reference signal 1 and reference signal 2. The sampling frequency is 2000 Hz. If d1 and d2 are the same, the noise signal arrives at two microphones with the same phase and similar amplitude. However, when d1 and d2 are different, $x_1(n)$ and x_2(n) have different phases and amplitudes, and simply combining $x_1(n)$ and x_2(n) by adding them together can cause two signals to cancel each other. Therefore, if the two microphones pick reference signals with different phase and amplitude, a delay should be added to the one reference signal before the combination of two reference signals. $\Delta = Fs * \Delta t$. The Δ will be *9* in this case. Therefore, add the *9* delays to *X1* for right reference signal. And then, combine the two reference signals to generate the input signal of the system. The following Figure 3-11 shows the attenuation after added delay. The input signal of the system is

$$x(n) = x_1(n + \Delta) + x_2(n) \qquad (3.1)$$

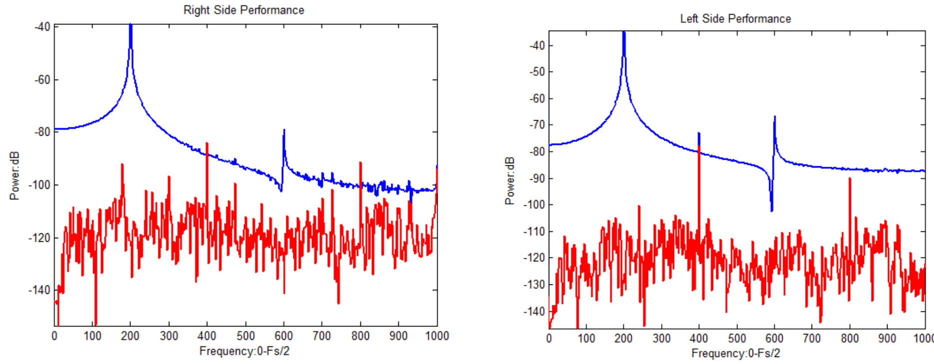

Figure 3-11 Frequency spectrum of 200 Hz sin-wav cancellation for combined reference $2\times2\times2$ ANC system

According to the performances of the proposed system, the left-side noise source has 41.58dB cancellation. The right-side noise has 32.29dB cancellation. Compared with the $1\times2\times2$ ANC system simulation result, the attenuation of two signals is increased by 9.7 dB and 3dB, respectively. When $W(z)$ goes to the optimal value, $W°(z) = \frac{P(z)}{S(z)}$. So, if there is a null in the spectrum of the secondary path, $\hat{S}°(z) = 0$, we may find a peak in the output of our ANC system. There is a null in the real-secondary-path at 400 Hz; therefore, we can find a high peak at 400 Hz in our result.

Then we use the NICU noise as input signal for CR $2\times2\times2$ ANC system and put the signal into combined multi-channel $2\times2\times2$ ANC system. Figure 3-12 shows the comparison of Frequency domain waveform before and after ANC working. According to the performances of the proposed system, the right-side noise source has average 8.9 dB cancellation. The left-side noise has average 7.2 dB cancellation.

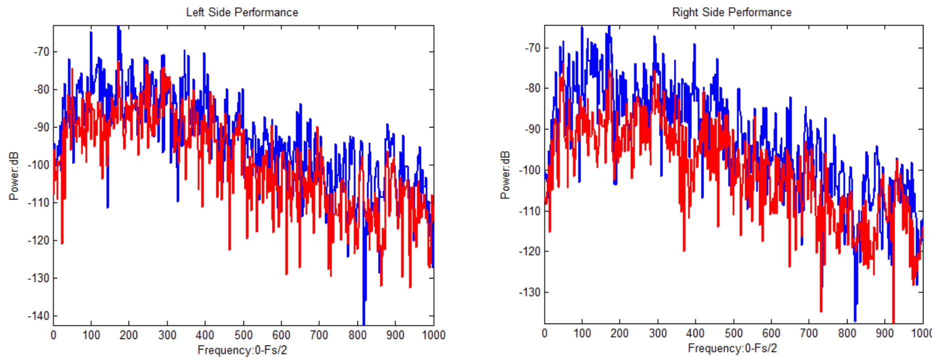

Figure 3-12 Frequency spectrum of NICU noise cancellation for combined reference $2\times2\times2$ ANC system

3.3. $2\times2\times2$ ANC system

Another solution to improve cancellation of the $1\times2\times2$ ANC system is to use $2\times2\times2$ ANC system as shown in Figure 3-13. The two reference signals are two input signals, respectively. According to the diagram, the $2\times2\times2$ ANC system consists of four adaptive filters.

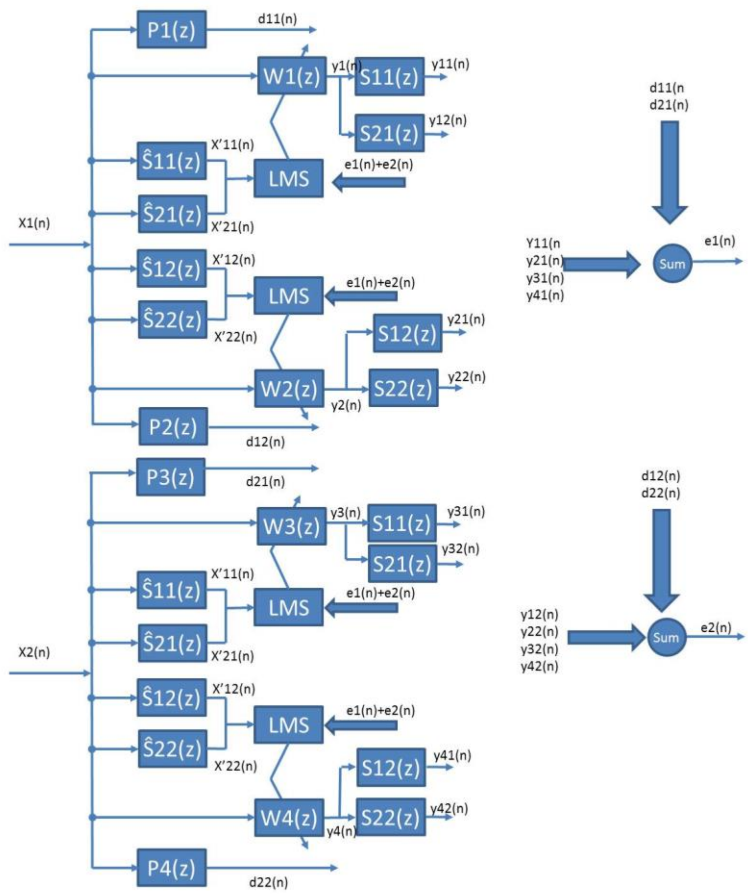

Figure 3-13 The 2×2×2 ANC system FXLMS algorithm block diagram

Compared with the 1×2×2 ANC system, the 2×2×2 FXLMS algorithm needs four adaptive filters instead of two and still needs four secondary path models. In addition, the anti-noise signal y_1(n) and y_2(n) are generated by each input channel signal passing two adaptive filters. And the 2×2×2 FXLMS algorithm block needs two error signals which are generated by two output signals. The two reference microphones have two primary path models. Therefore, the 2×2×2 ANC system uses two reference microphones, two secondary loudspeakers, and two error microphones, which is the same with CR 2×2×2 ANC system. However, it uses the double-channel FXLMS algorithm with two input channels for sensor error signals and two output channels for generating cancelling signal.

In order to investigate the performance of the $2\times2\times2$ ANC system, we used the same 200 Hz sinusoidal as the noise source, and the amplifier gains and experiment steps are the same as the $1\times2\times2$ ANC system presented for collecting the real-time data. Here are the results of the FXLMS algorithm. Figure 3-14 shows the time domain of the error signals and references signals. The sin-wav signals is cancelled by the $2\times2\times2$ ANC system. Since the left reference microphone is close to the noise source, it has better performance.

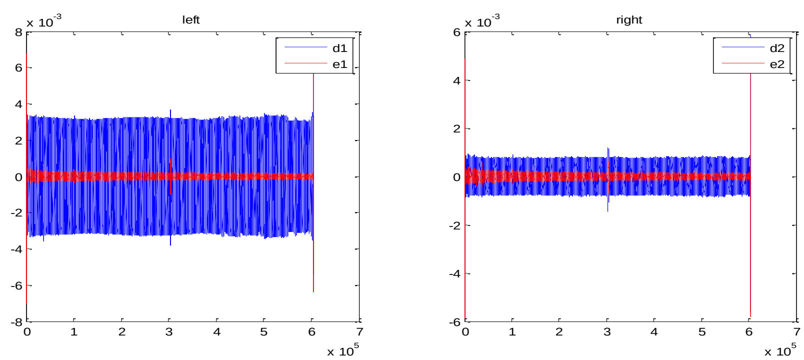

Figure 3-14 The time domain of reference signal and error signal for $2\times2\times2$ ANC system

In order to compare with the attenuation between two multi-channel ANC systems, the spectrum of the sin-wav signal shows that the $2\times2\times2$ FXLMS algorithm can effectively cancel single sinusoidal signal. For the left side, the algorithm achieves above 27 decibels average attenuation at the single frequency, and it achieves above 17 decibels average attenuation for right-side. It has one main harmonics that shows at 200 Hz with above 40 decibels attenuation, as Figure 3-15 shows. We change sin-wav to the NICU noise as the input signal for $2\times2\times2$ ANC system. Figure 3-16 shows the compared Frequency domain waveform of NICU before and after ANC working. According to the performances of the proposed system, the right-side noise source has 4.3 dB average cancellation. The left-side noise has 4.6 dB average cancellation.

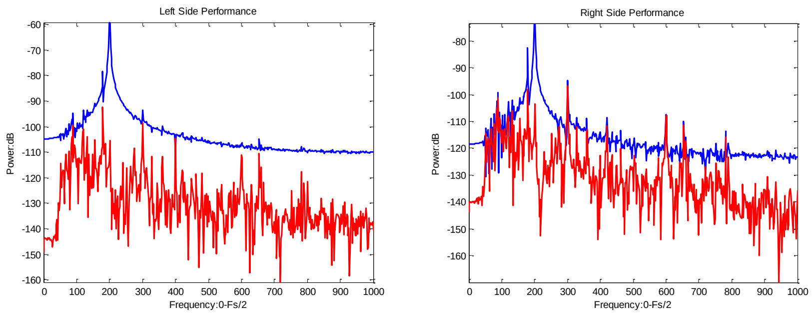

Figure 3-15 the frequency spectrum cancellation result for 2×2×2 ANC system

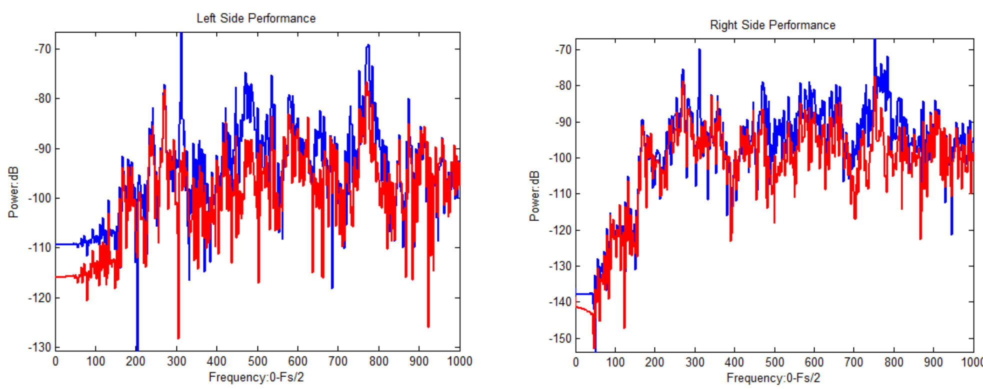

Figure 3-16 Frequency spectrum of NICU noise cancellation for 2×2×2 ANC system

3.4 Summary

The 2×2×2 ANC system has more computational complexity than the 1×2×2 ANC system(Table 3.2). According to the simulation results, the performance of CR 2×2×2 ANC system with the combined reference signals is better than the 2×2×2 ANC system and 1×2×2 ANC system. In 2×2×2 ANC system, each reference signal is input to system directly. Although the correlation of the two reference signals captured by the two reference microphones are relatively large, the two error signals will produce additional noise signal components while they are superimposed in the process. Therefore, the combined reference $2\times2\times2$ ANC system has best performance in our experiments. In the future, we will try spatial signal processing algorithm to pre-process the reference signal.

Table 3.2 Computational Complexity of Three ANC Systems

ANC system	$1\times2\times2$ ANC	CR $2\times2\times2$ ANC	$2\times2\times2$ ANC
Adaptive Filter quantity	2	2	4
"×"	(2W+8S+4L)*L	(2W+8S+4L+2)*L	(2W+8S+4L)*2L
"+"	(2W+8S+4L-10)*L	(2W+8S+4L-8)*L	(2W+8S+4L-10)*2L

Comparing the simulation result with three types of ANC system gives the data in Table 3.3.

Table 3.3 Cancellation Result from Three ANC Systems

ANC system Error Microphone	$1\times2\times2$ ANC	CR $2\times2\times2$ ANC	$2\times2\times2$ ANC
Sin-Wav Left	31.28dB	41.58dB	27dB
Sin-Wav Right	29.6dB	32.29dB	17dB
NICU Left	6.4dB	7.2dB	4.3dB
NICU Right	6.8dB	8.9dB	4.6dB

CHAPTER 4

ANC SYSTEM REAL-TIME IMPLEMENTATION

4.1 Single-Channel Hybrid ANC System Real-time Implementation

Infant incubators are used in neonatal intensive care units (NICU) for the care of pre-term babies [9]. Hybrid ANC systems use the filtered-X least mean square (FXLMS) algorithm. A hybrid ANC system includes the feed-forward and the feedback ANC system to remove undesired voice from inside and outside the incubator. This chapter is trying to implement the hybrid system for the infant incubator and evaluate the system performance through real-time experiment. The purpose of the experiment is to verify the simulation result and compare the proposed hybird ANC algorithms under the real noise environments.

According to the U.S Environmental Protection Agency, the continuous noise levels inside an incubator are higher than no-injured level from 50-75 dBA, depending on the instruments being used inside an NICU, and peak noise level can be in excess of 120 dBA [1]. The noise has been found to damage the NICU residents' health, including hearing loss and sleeping disturbances, affect baby's mental development. Therefore, high levels of noise inside the incubators should be reduced from baby's environments.

The single-channel hybrid ANC system is a combination of feed-forward and feedback ANC structures, as shown in Figure 4-1. The input signal $x(n)$ is generated by reference microphone. Combining both the outputs $y_1(n)$ and $y_2(n)$ from the feed-forward ANC filter $A(z)$ and the feedback ANC filter $C(z)$ generates the secondary signal $y(n)$ which is played by one loudspeaker. From the Figure 4-1, the estimated primary signal is generated by the error signal and the output signal and used as the input for the feedback system.

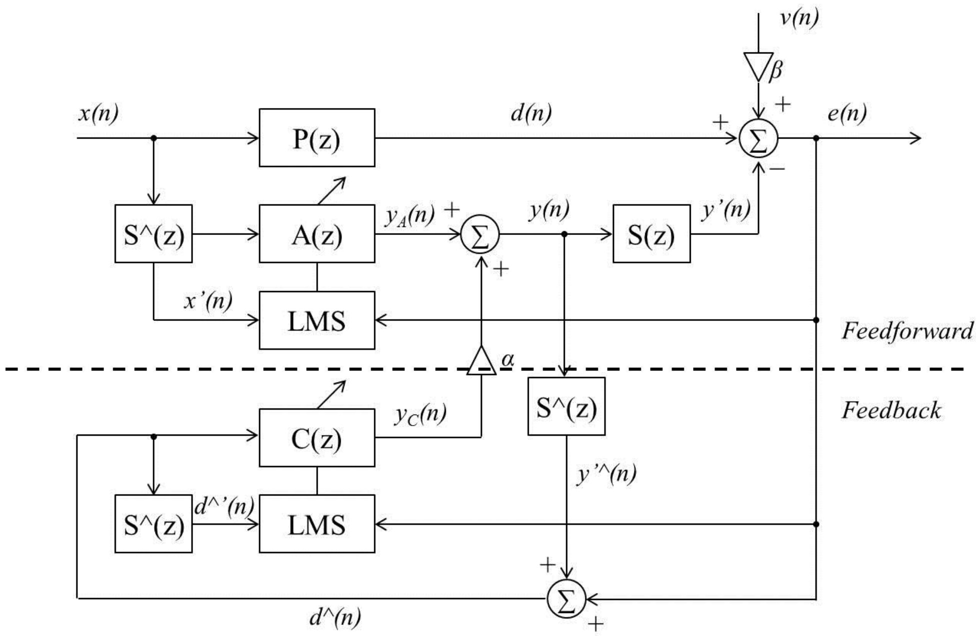

Figure 4-1 A single-channel hybrid ANC system combing feed-forward and feedback ANC system [10]

According to Figure 4-1, the developed multi-channel hybrid ANC algorithm combines feed-forward and feedback ANC algorithms. It can cancel noises from both inside and outside incubators. The algorithm of the hybrid ANC algorithm is given by:

$$y_A(n) = A^T(n)x'(n) \qquad (4.1)$$

$$y_c(n) = C^T(n)d'(n) \qquad (4.2)$$

where

$$\mathrm{A}(n) = [A_0(n)A_1(n) \dots A_{L-1}(n)]^T \qquad (4.3)$$

$$\mathrm{C}(n) = [C_0(n)C_1(n) \dots C_{L-1}(n)]^T \qquad (4.4)$$

$$\mathrm{y(n)} = A^T(n)x'(n) + \ C^T(n)d'(n) \qquad (4.5)$$

The coefficient matrix $\mathbf{A}(n)$ represents the feed-forward ANC filters and **C**(n) represents the feedback ANC filter. The input reference signal is x(n); d(n) is the reference signals from the estimated primary signals and α is the scaling factors for power to the feedback loops. The following equations show the signal vectors:

$$x(n) = [x(n)x(n-1) \dots x(n-L+1)]^T \qquad (4.6)$$

$$d(n) = [d(n)d(n-1) \dots d(\mathrm{n}-L+1)]^T \qquad (4.7)$$

The error signal *e*(*n*) is mixed with additive noise *v*(*n*) and measured by error microphone. *v*(*n*) is a predictable noise which is not correlated with the reference *x*(*n*). The additive noise can be expressed in a vector.

$$V(n) = [V(n)V(n-1) \dots V(n-L+1)]^T \qquad (4.8)$$

The secondary-path impulse response function ***S***(n) is given by

$$S(n) = [S(n)S(n-1) \dots S(L-1)]^T \qquad (4.9)$$

The single-channel error signals are

$$e(n)=d(n)+V(n)-y(n)$$

$$e(n) = d(n) + V(n) - [A(n) + \ \alpha C(n)]x(n) \qquad (4.10)$$

The filter coefficients are updated based on the minimum energy at the error sensors criterion which is often used in active noise control. A quiet zone is created around an error sensor. The sum of mean square errors (MSE) is

$$\varepsilon(n) \approx e^T(n)e(n) \qquad (4.11)$$

Based on LMS adaptive algorithm approach, one can adjust the coefficients of the feed-forward and feedback adaptive filter.

$$A(n+1) = A(n) + \mu x'(n)e(n) \qquad (4.12)$$

$$C(n+1) = C(n) + \mu d'(n)\mathrm{e}(n) \qquad (4.13)$$

The TMS320C6713 Development Starter Kit (DSK) will be used for the implementation of the single-channel hybrid ANC system based on the previous diagram and equation. The C6713 DSP is based on the high-performance, advanced long instruction word (VLIW) architecture developed by Texas Instruments (TI), making this DSP an excellent choice for multi-channel audio/speech applications [12]. The C6713 DSP chip is a floating-point processor which contains a central processing unit (CPU), internal memory, and enhanced direct memory access (EDMA) controller. The interface includes a 32-bit external memory interface (EMIF), four multi-channel buffered serial ports (McASP and McBSP), two 32-bit timers, a host port interface (HPI) for high-speed communication between chips in multi-DSP system [12].

The TMS320C6713 DSP chip is very powerful by itself, but for development of programs, a supporting architecture is required to store programs and data and bring signals on and out of the board. In order to use this DSP chip in a lab environment, a circuit board containing appropriate components, designed and manufactured by Spectrum Digital, Inc, is provided. Together, Code Composer Studio, the DSP chip, and supporting hardware make up the DSP Development Starter Kit (DSK). The following Figure 4-2 (copied from Spectrum Digital, Inc.) is the standard package of TMS320C6713 DSK distributed by Spectrum Digital, Inc [13]. It contains the power adapter, USB emulator cable, C6713 chip board, and Code Composer Studio software, and reference manuals. The DSP Starter Kit uses USB communications for true plug-and-play functionality. Both experienced and novice designers can get started immediately with innovative product designs with the DSK's full-featured Code Composer Studio IDE.

Figure 4-2 The standard package of the TMS320C6713 DSK.

Code Composer StudioTM (CCStudio) is an integrated development environment (IDE) for Texas Instruments (TI) embedded processor families [11]. CCStudio comprises a suite of tools used to develop and debug embedded applications. It includes complier for each of TI's device families, source code editor, project build environment, debugger, profiler, simulators, real-time operating system and many other features. The IDE provides user interface through each step of faster than before [11].

The interface of Code Composer Studio is shown in Figure 4-3. It contains a few key parts. The very top of the interface is the toolbar of the IDE software, the left part is the Project View of the current project, the debug toolbar is close to the project view part, the middle is the source file editor, and the below is the build and compile information of current project. The below right has the variables watch window there if it is opened. The details can be found from the Code Compose Studio Getting Started Guide by Texas Instruments [12].

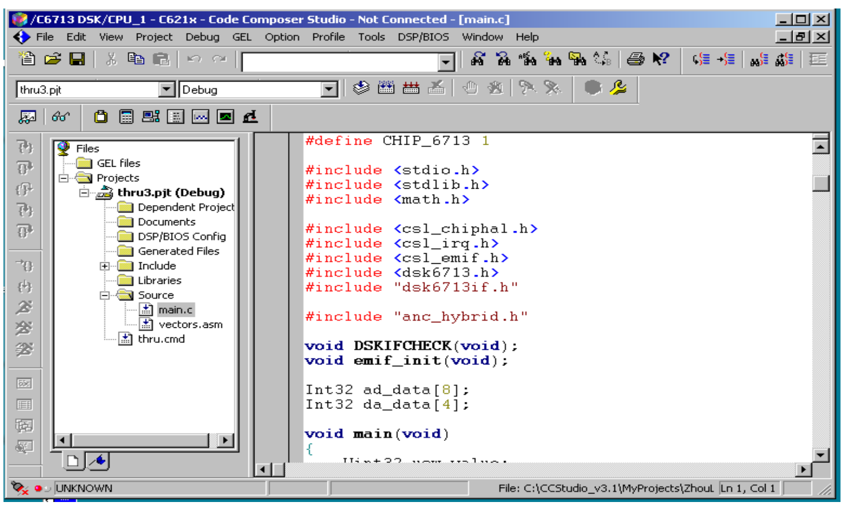

Figure 4-3 The Code Composer Studio interface.

From Figure 4-4 the implementation of hybrid ANC System with TMSC6713 is achieved to reduce noise in real-time active noise control application. The system setup includes two loudspeakers and two microphones: one reference microphone is which located at the upper left corner, one error microphone which is inside of incubator near baby's left ear, one loudspeaker plays anti-noise which located at the left corner in the incubator, one louder speaker plays interference noise located at the right corner in the incubator. It imitates the inside noise.

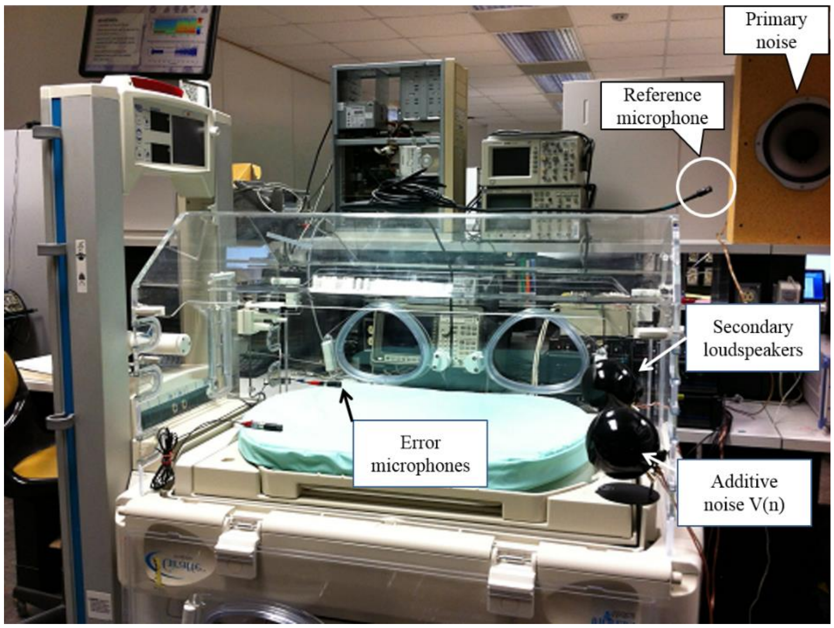

Figure 4-4 Single-channel hybrid ANC system implementation [8]

The hybrid ANC program consists of two parts: off-line secondary-path modeling part and on-line training part. Figure 4-5 shows the flowchart of the hybrid system program. At first, the variables of system are initialized. We define the parameters, such as filter length, step size value and so on. The program contains some functions in the C file *main.c*. The main algorithm processing function is used to interrupt the program. Secondary, the program starts sampling data and counting. If the counting number is less than defined training number, the system runs the off-line modeling program. It models an array of secondary-path $\hat{S}(z)$, which its length depends on the secondary-path length. After the LMS algorithm estimates the impulse response of secondary path, the system runs the hybrid algorithm function. The input signal passes the adaptive filter of feed-forward system to generate an output signal $y\_a[n]$. The feed-forward output signal $y\_a[n]$ combines with the feedback output signal $y\_c[n]$ to generate the anti-noise which becomes the hybrid system output $y[n]$. We use the hybrid system output and error signal input to reconstruct the feedback input signal. Then, the error signal $e[n]$ and hybrid system output $y[n]$ import to LMS algorithm for updating the feed-forward and feedback adaptive filter coefficients. At last, the system runs continuously in the hybrid system program until it receives a stop command. The program runs the initialization part to clear history data when it starts again.

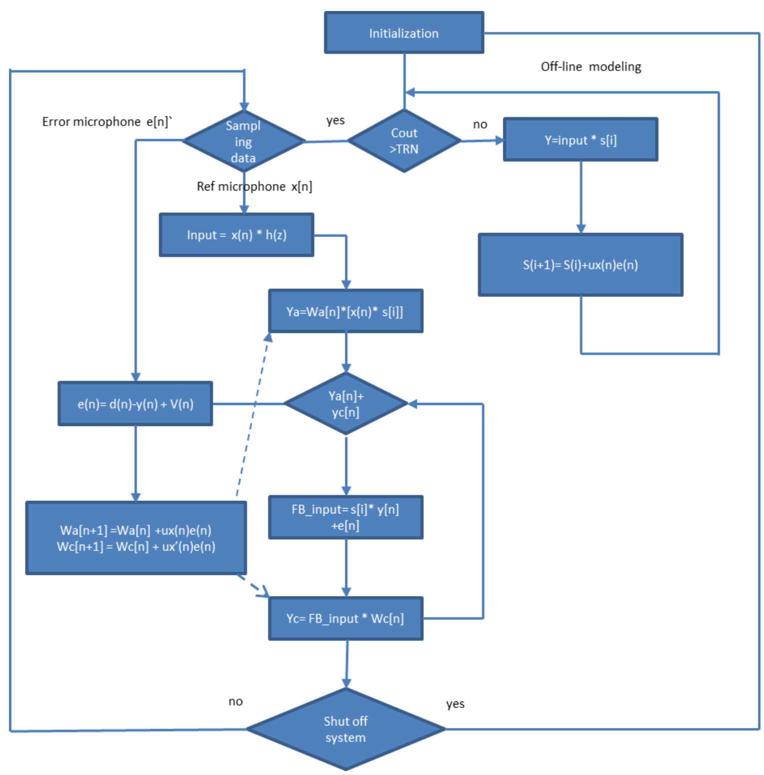

Figure 4-5 Hybrid ANC program flowchart

From the Figure 4-5 simulation results, single-channel feed-forward ANC system gets 30 dB cancellations by using 200HZ sin-wav. Second, single-channel feedback ANC system is implemented. It gets 15dB cancellation by using 200 Hz sin-wav. Finally, combined feed-forward and feedback ANC system expected to get 30 dB cancellations around error microphone, as Figure 4-6 shows. Then, the NICU noise instead of sin-wav to becomes input signal $x(n)$. Figure 4-7 shows the average reduction achieved of 15.2 dB in frequency range low 1000 Hz.

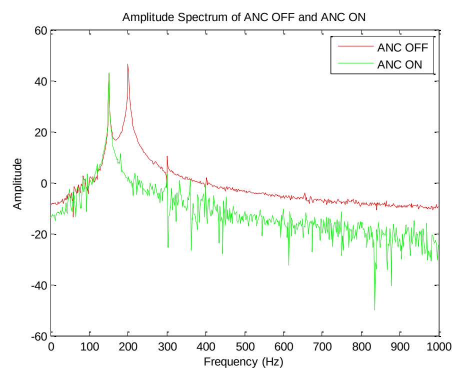

Figure 4-6 Amplitude spectrum of hybrid ANC system for sin-wav

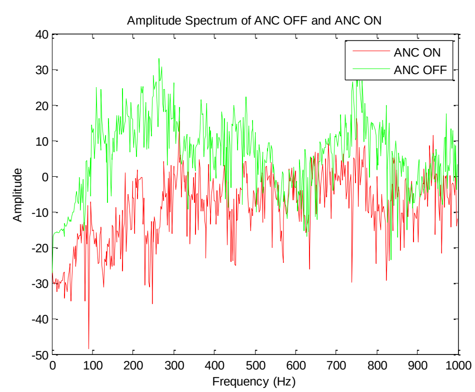

Figure 4-7 Amplitude spectrum of hybrid ANC system for NICU noise

4.2 On-Line Secondary-path Modeling MATLAB Simulation

This section studies ANC system with on-line secondary path modeling. Since the secondary-path is time varying in many application areas, we need to update the secondary-path without turning off the whole ANC system. The baby's head always moves, so the secondary-path channel always changes. And also, some background noises could not be stopped, such as medical equipment power motors. Therefore, if we want to improve the performance of ANC systems, the on-line secondary-path modeling is necessary. Based on the feed-forward ANC system, it implements the three block diagrams: Figure 4-8, Figure 4-10 and Figure 4-11. Then, we use audio source instead of the random noise to train the second path. The interference of audio source affects the cancellation of the unwanted noise. We use audio with different volumes to train ANC system and compare result. Then, find out the solution for elimination interference. The on-line secondary path modeling ANC system tries to model second-path coefficients without turning off the primary noise source. It finds out two solutions to reduce the interference from audio source and compares the results of simulation.

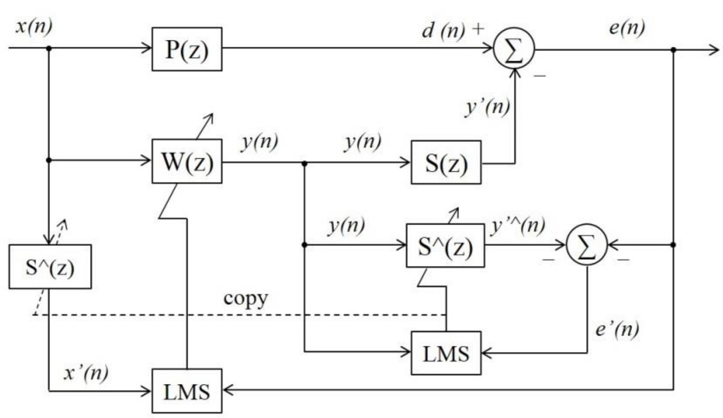

Figure 4-8 Block diagram of ANC with audio masking

The optimum of the secondary-path modeling results can be expressed as [1]:

$$\hat{S}^{\circ}(z) = S(z) - \frac{P(z)}{W(z)} \qquad (4.14)$$

where $W(z)$ goes to optimum values, $W^{\circ}(z) = \frac{P(z)}{S(z)}$, then $\hat{S}^{\circ}(z)$ =0; this is such a conflict. The noise reduction performance of this ANC with this on-line modeling technique is shown in Figure 4-8. The sampling frequency is 2 kHz, length of adaptive filter $\hat{S}(z)$ is 128 with updating step size 0.004. The adaptive filter $W(z)$ length is 128 with updating step size 0.00005. The NICU noise can barely be canceled and the secondary-path modeling is failed, as Figure 4-9 shows.

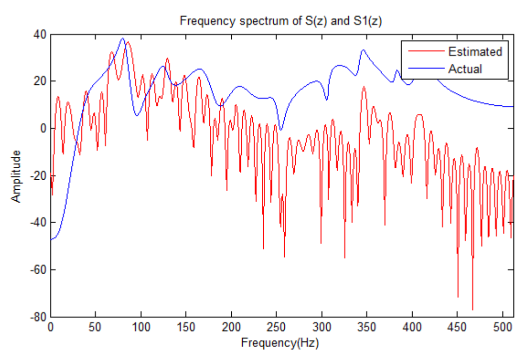

Figure 4-9 Frequency spectrum of $S(z)$ and $\hat{S}(z)$ for on-line training

The second solution is to use random noise as excitation signal to model the secondary-path as shown in Figure 4-10. We add white noise to on-line training system. Sampling frequency of NICU noise is 2000 Hz. It is expected to reduce the interference of original noise. We adjust α value to control the energy of the white noise and observe the relationship between α and $\hat{S}(z)$.

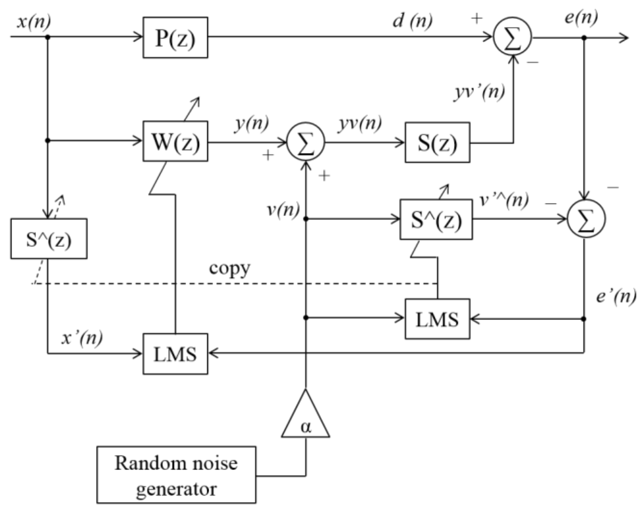

Figure 4-10 Block diagram of on-line secondary-path modeling using additive random noise

The input signal uses incubator noises. The sampling frequency is 2KHz, length of adaptive filter $\hat{S}(z)$ is 128 with updated step size 0.0002; length of adaptive filter $W(z)$ is 128 with updating step size 0.00005. Gain α is 0.8.

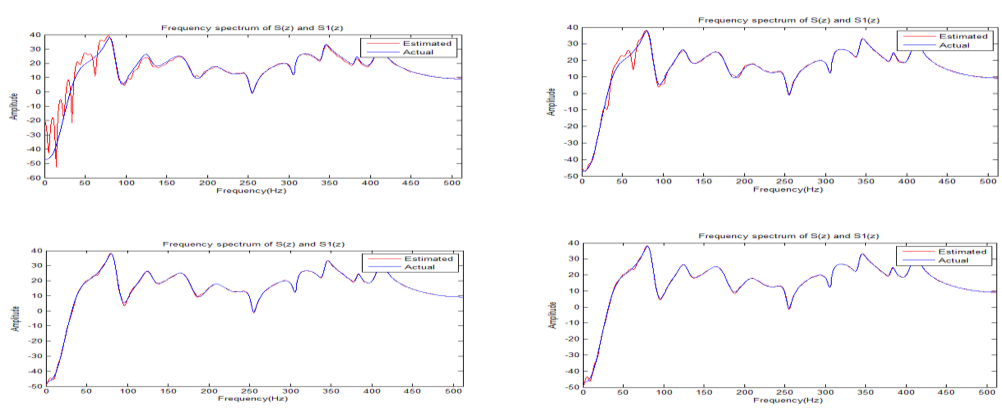

Fig.4-11. Frequency spectrum of $S(z)$ and $\hat{S}(z)$ a=0.2,0.5,0.8 and 1

Fig. 4-11 shows the frequency spectrum of $S(z)$ and $\hat{S}(z)$. The increased energy of white noise leads the coefficients to close to reality value. When α equals 1, ANC system models secondary-path coefficients very well. Therefore, the incubator noise can be canceled and the secondary-path modeling is a success.

Then we setup the complete structure on-line modeling technique using audio sound as excitation, as shown in Figure 4-12. This ANC system uses audio signal to model the second path on-line. The adaptive filter $H(z)$ is used to remove the interference noise component which has higher power than the excitation signal, thus the $f(n)$ will converge faster. The ANC performances and on-line secondary-path modeling results are shown in Figure 4-13 and Figure 4-14.

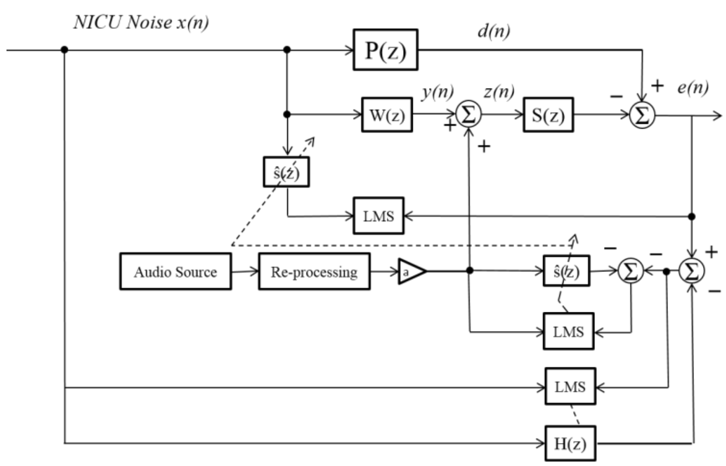

Figure 4-12 On-line secondary-path modeling using audio signal

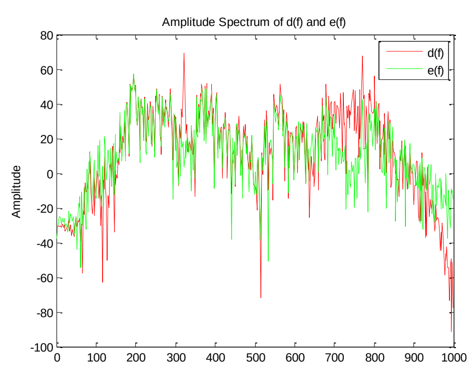

Fig.4-13 Frequency spectrum of $d(f)$ and $e(f)$ a=0.2

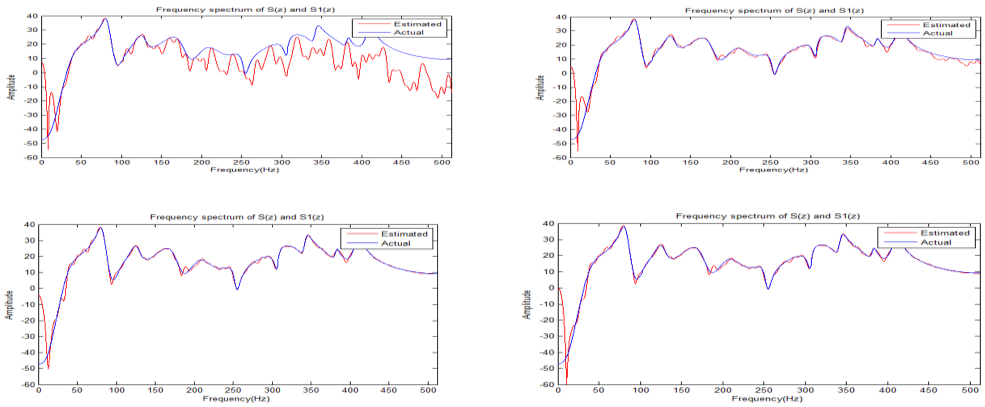

Figure 4-14 Frequency spectrum of $S(z)$ and $\hat{S}(z)$ a=0.2,0.5,0.8 and 1 for on-line training

The input signal uses incubator noise (NICU). The sampling frequency is 2Hz, length of adaptive filter $\hat{S}(z)$ is 128 with updating step size 0.0002; length of adaptive filter $W(z)$ is 128 with updating step size 0.00005. Gain α is 0.8. The secondary-path can be modeled better, but at as a cost; the NICU noise reduction level decreases as we add more noise to the environment.

4.3 On-Line Secondary Modeling Real-Time Experiment

The TMS320C6713 DSK will be used for the implementation of on-line secondary modeling system. The on-line modeling algorithm was written based on the previous diagram, in Figure 4-7. As it has shown in simulation, the real-time implementation algorithm is similar to simulation. In this section, we design the program diagram as Figure 4-15 shows. We make this DSP project achieve our purpose.

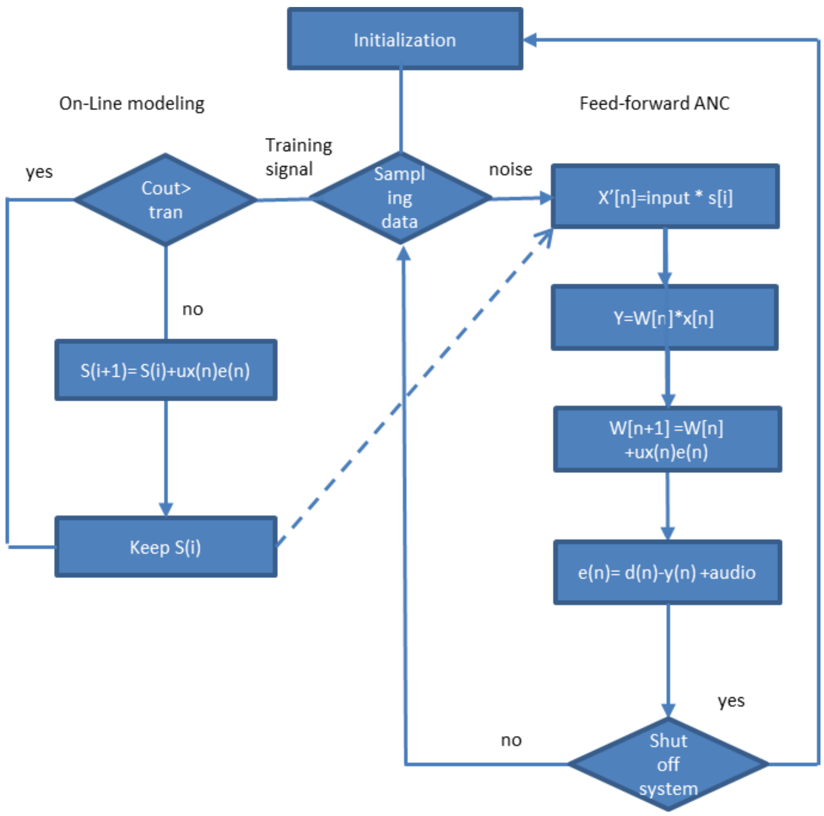

Figure 4-15 Program flowchart of on-line modeling secondary path

The on-line modeling program consists with two parts: on-line secondary-path modeling part and on-line noised cancellation part. As the previous program, the variables of the system are initialized. The filter length and step size value are defined by the initialization program. Next step, the program starts sampling data and counting. If the counting number is less than defined training number, the system plays training audio sound. The training signal models an array of secondary-path $\hat{S}(z)$ under the noise playing background. After the LMS algorithm calculated the impulse response of secondary path, the system does not update the secondary-path coefficients. And, the input signal passes the adaptive filter of feed-forward system to generate an output signal y[n]. The error signal $e[n]$ and output $y[n]$ import to LMS algorithm for updating the feed-forward adaptive filter coefficients. Finally, the system continuously cycles for on-line modeling system program until it receives a stop command. The program runs the initialization part to clear history data again when it is started by the end user.

On-line modeling simulation result is shown in Figure 4-16; the secondary-path modeling curve is close to the measured result. For this system, it achieved the 15 dB cancellation. We increase the audio signal volumes, and it hardly achieves high cancellation since it added more noise into the system. The secondary-path modeling result becomes worse. Then, the adaptive filter length is changed to 256 and the secondary-path filter length is adjusted to 128 points. After training, the performance of cancellation is better than previous one. The highest cancellation achieved 22 dB. However, the secondary-path modeling is worse than long filter length (see Figure 4-17).

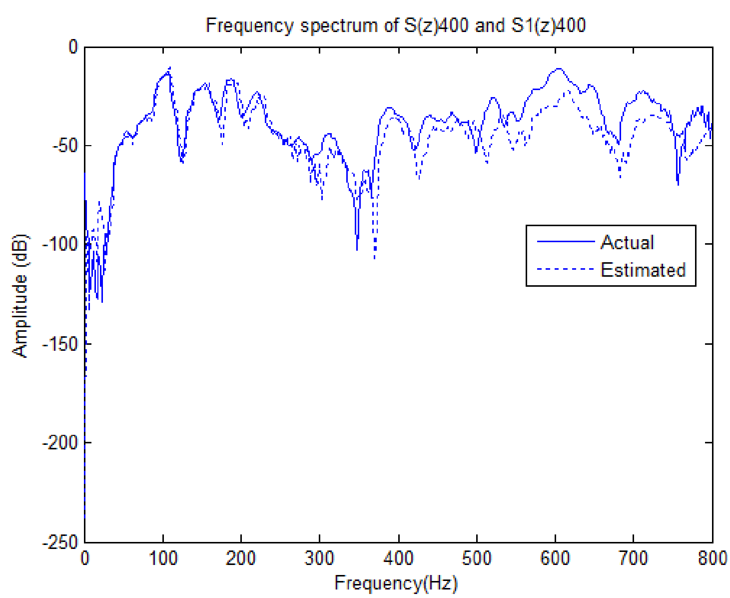

Figure 4-16 Frequency spectrum of $S(z)$ and $\hat{S}(z)$ filter length 400 for on-line training

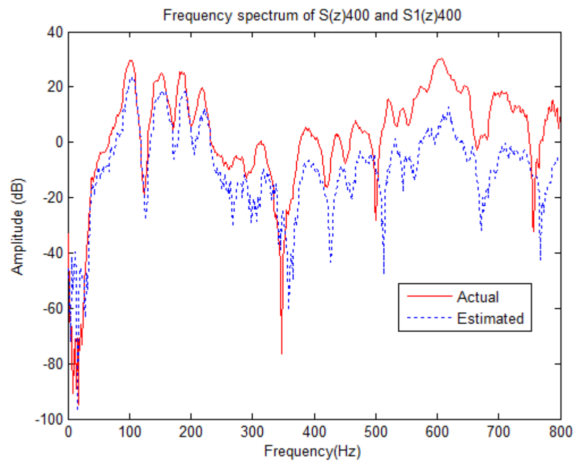

Figure 4-17 Frequency spectrum of $S(z)$ 256 and $\hat{S}(z)$ 128 filter length for on-line training

4.4 Summary

In this chapter, the real-time experiments for single-channel hybrid system and on-line modeling ANC system for incubator are conducted. For hybrid system, it does not only cancel the outside of incubator noise but also cancel the inside noise. Compared with $1\times1\times1$ feed-forward ANC system, the hybrid system has better performance. The on-line training system is studied to track the secondary-path coefficients. We also investigated the relationship between the convergence of the system and the energy of the training signal. When the energy of the training signal increases close to the original noise, the modeling performance becomes better. However, the audio training signal brings extra noise into the system. Therefore, the performance of cancellation becomes worse. Our future work will focus on finding the suitable parameters, such as filter length and audio gain.

CHAPTER 5

CONCLUSIONS

This thesis discusses the implementation of active noise control system in the infant incubator and improves the environment for infants in NICU. Based on the real Giraffe infant incubator, the hybrid active noise control system and ANC with secondary-path on-line modeling are implemented by using TMS320C6713 DSK in real-time experiments.

According to the popular structure of active noise control system, this thesis discusses the basic principles of the classic least mean square (LMS) algorithm for adaptive filters and the filtered-X least mean square (FXLMS) algorithm. Several secondary-path modeling algorithms and their simulation results are presented. The development of multiple-channel active noise control with combined reference signals is proposed. The implementation of ANC system with on-line secondary path modeling is investigated. From the simulation results, we find that the energy of additive signal needs to be adjusted to a reasonable value in the real-time experiment and real application.

The single-channel hybrid system and on-line modeling ANC system for incubator are conducted in this thesis. The hybrid system out-performed the $1\times1\times1$ feed-forward ANC system. The on-line secondary-path modeling algorithm can track the time-varying secondary-path effectively. In the future, noise source detection combined with reference signal processing can provide a better solution for the development of multi-channel ANC system in multiple unknown noise sources environment with reasonable cost. Specially, the future research will be pre-processing the reference signal by using array processing to improve ANC system’s performance.

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