Disturbing Sound Cancellation

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Abstract

When doing recording work in the studio, disturbing sound must be removed. In this thesis, the purpose of this thesis is to formulate a mathematical equation, by using MATLAB to identify a system, then using the system to do cancellation of disturbing sound. The method of doing cancellation is to subtract the simulated output by the actual output, and then the disturbing sound was cancelled. The main thesis work will focus on the system identification, which is the process of determining the characteristic of an unknown system. Three systems were identified with the same model structure, which is linear (ARX) model. After finding out the model, the model quality must be evaluated. If the model is valid, there is a discussion if it is possible to run the mathematical equation in the real application, and how is the market today.
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1. Introduction

Sound is always the combination of many sources, if different sound sources mix together, echo can be introduced, and cause the sound impact. In practice, when people doing recording work in the studios, all the instruments sound are recorded at the same time, but in some case, people do not need all the sound, some undesired sound have to be removed. The idea to remove the unwanted sound is to use the method of system identification, build a model, by subtracting the simulated output with actual output, cancel the disturbing sound. System identification requires the knowledge of digital signal processing. This sound cancellation could be carried out by adaptive filter theory, and the sound cancellation technique could be implemented in an integration circuit or means. Fortunately, the sound cancellation could also be processed and simulated in the MATLAB software. In this thesis, MATLAB is used to do system identification. System identification is extremely helpful to identify systems, which you are difficult to model from principles or specifications. The system identification requires data, it uses the input and output signals you measured from the system or given to estimate the values of parameter in a certain model structure [1:1-2]. In this case, there is a need to build the model using the time-domain input and output signals.

Since system identification requires data, for example, input data and output data, and they will be used to capture the important parameters of the system, so before do the experimental design, one thing needs to ensure that is measure the right variables with sufficient accuracy and duration to capture the parameters, which is wanted to the model. So the important is:

• Use an input signal, which could capture the system properly and adequately.
• Use a data that long enough to capture the important parameters of system.
• Chose an suitable sampling intervals, or do the cross-correlation [1:2-9].

In generally, analyze the data quality before building the model is the first to do. For example, analyze the input signal to determine peak frequency, do input and output cross-correlation, and determine the output delays, important time
constants, and determine the model structure. In this thesis, the unknown system is a linear time invariant system, but linear ARX model is chosen for the estimation model, no other complex models will be estimated.

In the system identification area, ARX model is always used for describing dynamic data for both linear and non-linear systems. The ARX model structure contains a number of time intervals and a set of parameters. There are a lot of algorithms can be applied to identify the parameters of the linear ARX model. In all these algorithms, the least squares algorithm is the most often used. So the parameters of the system in the discrete time domain are identified by least squares algorithms, and the numbers of time intervals must be decided beforehand. However, in practical, an unknown system is hard to estimate a correct model order, and a correct model structure should have exactly enough output predictions from given inputs [2].

The purpose of this thesis is to formulate a mathematical equation, by using MATLAB to identify a system, then using the system to do cancellation of disturbing sound. The linear ARX model is chosen as the model structure. The method of doing cancellation is to subtract the simulated output by the actual output, and then the disturbing sound was cancelled without affecting the left sound.
2. Theory

In this chapter, theories are presented, which are relative to the thesis, or used in the thesis.

2.1 Digital signal processing

Digital signal processing is a processing of input signal by using the digital means. A digital signal could transmit by a wire or by a telephone line or by a radio wave, means numbers or digits, are performing a numerical calculation. When we pass a signal through a system in a device, we say that we have processed the signal, and such operations are usually referred to as signal processing [3:2-4].

2.2 System identification

The process of determining the characteristics of the unknown system, by a set of measurements performed on the system. For example, when we doing something in digital communication problem just described, if the frequency response of the channel is unknown, the measurement of the channel frequency response could be accomplished by transmitting a set of equal amplitude sinusoids, at different frequencies with a specified set of phases, within the frequency domain of the channel,. The channel will attenuate and phase shift of each of the sinusoids. By comparing the received signal with the input signal, the receiver obtains a measurement of the channel frequency response which can be used to design the inverse system [3:326-327].
2.3 Static model

In the static system, the values of the output signals depend on only the instantaneous values of its input signals, but do not depend on the past behavior of the system, this kinds of system is called Static System or Static Model. The model is a mathematical relationship between a system’s input and output signal. Normally, there are two kinds of forms to represent static models both in continuous-time and discrete-time form [3:58-59].

2.4 Discrete-time dynamic model

Discrete-time Dynamic Model: assume the input variables $u(t)$ and output variables $y(t)$ identify a system at a discrete time instants $t = nT_s$, where $T_s$ is a fixed time interval, and $n = 0, 1, 2, 3, 4, …$. The variables are said to be sampling interval $T_s$. then, now we could represent a relationship between the sampled input variables and output variables as a difference equation, such as:

$$y(t) + a_1y(t-T_s) + a_2 y(t-2T_s) = b_1u (t-T_s)$$  \hfill (2.4.1)

The $a_1$ and $a_2$, and $b_1$ are the model parameters. The $y(t)$ could be represented as a weighted sum of the past input and output values

$$y(t) = b_1u(t-T_s)- a_1y(t-T_s) - a_2 y(t-2T_s)$$  \hfill (2.4.2)

Normalized we use the simplest $T_s = 1$. So

$$y(t) = b_1u(n-1)- a_1y(n-1) - a_2 y(n-2)$$  \hfill (2.4.3)

Equation (2.4.3) is the normally used [1:2-5].

2.5 ARX model

A model structure is a mathematical relationship between input signal and output signals variable that contains unknown parameters.

For a single input and single output system (SISO), the ARX model structure is:

$$y(t) + a_1y(t-1) + \ldots +a_ny(t-n_a) = b_1u(t-n_k) + \ldots + b_nbu(t-n_k-n_b+1) + e(t)$$  \hfill (2.5.1)
y(t) is the output signal at time t, u(t) is the input signal at time t, n_a is the number of poles, n_b is the number of zeros plus 1, n_k is the input signal delay, and e(t) is the white-noise disturbance. Assumed e(t)=0 [1:4-31].

\[ y(n) = \frac{B(q-1)}{A(q-1)} u(n) \]  \hspace{1cm} (2.5.2)

### 2.6 Principle of estimating model parameters

The method to estimate the model parameters is to minimizing the error between the model output and the measured output response, assuming G is the model transfer function, u(t) is the input function, the output y_{model} of the linear model is given by:

\[ y_{model}(t) = G(u(t)) \]  \hspace{1cm} (2.6.1)

To determine the transfer function G, we just minimize the difference between the model output y_{model}(t) and the measured output y_{meas}(t). Assuming the error e(t):

\[ e(t) = y_{meas}(t) - y_{model}(t) = y_{meas}(t) - G(u(t)) \]  \hspace{1cm} (2.6.2)

y_{model}(t) is the simulated response of the model from the input u(t), the error e(t) is called model error, it should be as small as possible [1:2-11].

### 2.7 Linear time invariant system

Linear system is based on the superposition principle, the response of the system to a weighted sum of the signals be equal to the individual input signals. Time invariant system is if its input and output characteristics do not change over time [3:302-303].
2.8 Cross-correlation

Suppose that we have two real signal sequences \( x(n) \) and \( y(n) \), and each of them have finite energy. The cross-correlation of \( x(n) \) and \( y(n) \) is a sequence:

\[
r_{xy}(l) = \sum_{n=-\infty}^{\infty} x(n) y(n-l), \quad l=0, \pm 1, \pm 2, \ldots
\]  

(2.8.1)

The index \( l \) is the time shift parameter, and the subscript \( xy \) on the cross-correlation sequence \( r_{xy}(l) \) indicate the sequences being correlated. To right shift \( l \) positive, to left shift \( l \) negative [3:115-117].

In the case, input signal is a directing recording, the output signal is a microphone recording of the input signal. The delay is produced during the recording. Assume \( u \) is input signal, \( y \) is output signal, \( r_{uy}(l) \) indicate the sequences \( u \) and \( y \) being correlated. The high correlation value is measured.
3. Process

There are three sets of data can be uses to identify three different systems. These three sets of data are completely different. Two sets of data are given by Niklas Rothpfeffer, and the third set of data is provided by Per Landin. It is very important to know the original sets before doing the thesis work. This is done by listening the data, and analyzes the frequency response in MATLAB.

The process of doing system identification requires that:

- Analyze the input signal and output signal for preparing the system identification in time or frequency domain.
- Selecting a (ARX) model structure. There are many models structures, linear model or nonlinear model.
- Applying an estimation method in order to estimate the values for the adjustable parameters in the linear ARX model structure. The least square algorithm is most used.
- Evaluate the estimated linear ARX model to check whether the model is adequate for my application needs[1:2-2].

Before doing system identification, one thing is needed be done beforehand, find the time delay between input and output signals. Apply the theory cross-correlation to find the delay.

MATLAB code:

```matlab
N=length(u);
N is the length of the input data. u is input, y is output.
[val ind]=max(abs(xcorr(u,y,N))); Return maximum value and index where the values are found. xcorr estimates the cross-correlation between y and u.
Delay=ind-N;
plot(-N:N, abs(xcorr(u,y,N)))
```
The MATLAB code for ARX model:

```matlab
data=Iddata(y,u)
```

Create an iddata object for time-series data, containing a time-domain output signal y and an input signal u.

```matlab
orders(na nb nk)
```

Vector of integers, na, nb are the order of the ARX model, nk is the delay, they are the corresponding integer values.

```matlab
arx(data,orders)
```

Return a model as an idpoly object, and use the least-squares method to specify orders

The idea to estimate the model parameters is to minimizing the error between the model output and the measured output response. Apply the equation (2.6.2):

\[ e(t) = y_{\text{meas}}(t) - y_{\text{model}}(t) \]

The MATLAB code:

```matlab
mod=arx(data,[na nb nk]);
```

Formulate a Linear ARX model.

```matlab
Ysim=sim(mod,Iddata([],input));
```

The `sim` command causes the specified simulink model to be executed. Ysim.y is the simulate output, input is used to valid the model, use the ARX model and comes to simulink software.

```matlab
error=Ysim.y-outputdata;
```

Error is the difference between the model output and measured output. The small error means the estimated model is approaching the true system.

The method to estimate the model is to compare the model output with the measured output, or listen to the wanted music. The wanted music is the measured output subtracts the model output. Use the command `wavwrite` to export the wanted sound.
MATLAB code for cancellation:

Mod=arx(data,order);
B=Mod.b;
Get the B coefficient.
A=Mod.a;
Get the A coefficient.

ymod=Filter(B,A,input)
Filter the input data.
sound =y-ymod
wavwrite(sound)
Get the wanted sound and export the sound

3.1 The first set of data

The first set of data is plotted in time domain. The input signal is a direct recording piano sound and chirp sound, and the output signal is microphone recording piano sound and chirp sound with some music, the ARX model will be built using time domain input-output signals. Fig.1 is input signal, Fig.2 is output signal. The input signal contains two different spectral contents, so two different order models will be identified by using first half input and output signal, one model for piano sound and one model for chirp sound. Then compare two different order models, chose the better one as the final model to cancel the piano and chirp sound in the second half part of the output signal. Theoretically, only music is left.
3.1.1 Analyze the input and output signal

Before building the model of the system, the input signal needs to be analyzed. As can be observed in Fig.1, there are two kinds of data spectrum, piano sound is in the domain (0.16625e6:2.319e6), and chirp sound is in the domain
(2.55e6:5.25e6), chirp sound has high frequency. Then there is a need to analyze the output signal as well. In Fig.2, the output piano signal in the domain is about (0.1e6:2.4e6), and the output chirp signal in the domain is about (2.5e6:5.3e6). There is also a music spectrum in the domain (5.5e6:11.3e6) mixed with piano sound output and chirp sound output.

Find the time delays between input signal and output signal. Apply the theory in section 2.8. Fig.3 shows the cross-correlation of input and output signals for the first data set. Y-axis shows the number of lags between input and output signals. When x is -109, it has the biggest lags.

![Figure 3 cross-correlation of input and output signal from the first data set](image)

In Fig.3, x=-109, so the output time delay is 109. The system is left shift.

### 3.1.2 Synchronization

The synchronization is presented, for different spectral content of the first data set. In all the figures, direct recording as input signal is in blue, and microphone recording as output signal is in red. As can be observed in Fig.4, it shows the
synchronization for the input and output piano sound. Fig. 5 shows the synchronization for the input and output chirp sound.

The MATLAB code for the synchronization is:

```
circshift();
```

Circularly shifts the values in the array, by a shiftsize element.

Figure 4 piano input and output synchronization from the first data set

Figure 5 chirp input and output synchronization form the first data set
3.1.3 Identify the models

Since the synchronization is done, so $n_k$ is zero, which is the corresponding integer value. Then the system order becomes $[n_a \, n_b \, 0]$. Apply the method in 2.6 to find the system order. The model errors are presented in 3D-graph figures below. Theoretically, the model error decrease as the model order increase. The smaller the model error, the better the estimated model is. The estimated model more approach to the true system. A model with positive model error is a destructive model. Positive model error destroys the system, more damage than good.

The model error will be represented by db. 0 db is a limitation, since error=$y-G(u)$, 0 db means error=$y$. In the case, the model error must be less than 0 db.

Fig.6 represents model error of piano sound from the first data set. The lowest model error is about -18 db, which is good enough for the system. The model error decrease as $n_a$ and $n_b$ increase. Small model error means good system, so $n_a = 13$, and $n_b = 16$ are chosen for the ARX model of piano sound.
Figure 6 model of piano sound from the first data set

Fig. 7 represents model error of chirp sound from the first data set. The lowest model error is about -4 db, which is also good enough for the system. The model error decrease as $n_a$ and $n_b$ increase. Small model error means good system, so $n_a = 17$, and $n_b = 16$ are chosen for the ARX model of chirp sound.

Figure 7 model of chirp sound from the first data set
Once the order of the model is chosen, model estimation is needed to be done. For the first set of data, two different models are identified. By comparing the two different order models, the better model will be chosen to do disturbing sound cancellation. The results will be presented in the next chapter.

3.2 The second set of data

The second set of data is plotted in time domain. The input data is a directing recording piano sound with chirp sound and noise, output data is the input data recorded by microphone. The ARX model will be built using time domain input-output signals. Fig.8 shows the direct recording chirp sound. Fig.9 shows the microphone recording chirp sound.

Figure 8 input chirp sound from the second data set
3.2.1 Analyze the input and output signal

The input signal is mix sound, which contains piano sound, chirp sound, and noise. Since the chirp sound has a lot of frequencies, so it could excite the system properly and adequately, the chirp sound is chosen to identify the system.

Find the time delays between input signal and output signal. Apply the theory in section 2.8. Fig.10 shows the cross-correlation of input and output signals from the second data set. Y-axis shows the number of lags between input and output signals. When x is -467, it has the biggest lags.

Figure 9 output chirp sound from the second data set
Figure 10 cross-correlation of input and output signals from the second data set

In Fig.10, $x=-467$, so the output time delay is 467. The system is left shift.

### 3.2.2 Synchronization

The synchronization is presented. In Fig.11, direct recording as input signal is in blue, and microphone recording as output signal is in red. As can be observed in Fig.11, it shows the synchronization for the input and output chirp sound.

Figure 11 chirp input and output synchronization from the second data set
3.2.3 Identify the model

Since the synchronization is done, so \( n_k \) is zero, which is the corresponding integer value. Then the system order becomes \([n_a \ n_b \ 0]\). Apply the method in 2.6 to find the system order. The model errors are presented in 3D-graph figures below. Theoretically, the model error decrease as the model order increase. The smaller the model error, the better the estimated model is. The estimated model more approach to the true system. A model with positive model error is a destructive model. Positive model error destroys the system, more damage than good.

Fig.12 represents model error of chirp sound from the second data set. The lowest model error is about 24 db, which is very bad for the system. The model error in db is positive; this model is a destructive model, could destroy the system, more damage than good.

Since the linear ARX model is bad for the second set of data. Other model structure could be chosen. There is an easy way to do system identification for complex model structure, using the System Identification Tool Box [1]. Synchronization also needs to be done beforehand.
Figure 12 chirp sound model from the second data set

### 3.3 The third set of data

The third set of data is plotted in time domain. Input data is a simulated disturbing sound, output data is microphone recording input data mixed with music. Fig.13 shows the simulated input signal. Fig.14 shows the output signal. The first half of input and output signal are used to identify an ARX model, then use the model to cancel the disturbing sound in second half part of output signal. The wanted sound is the music. Theoretically, only music is left after the disturbing sound cancellation.
Figure 13 simulated input signal from the third data set

Figure 14 output signal from the third data set

3.3.1 Analyze the input and output signal

Do the cross-correlation of the input and output signals, find out the important time interval, or time delay. In Fig.15, the important time interval is -842.
MATLAB code:
plot(-1000:1000, abs(xcorr(v(1:period),y(period+(1:period)),1000)))
xcorr is doing cross-correlation of v and y.

Figure 15 cross-correlation of input and output signal from the third data set

3.3.2 Synchronization

The synchronization is presented. In Fig.16, direct recording as simulated input signal is in blue, and microphone recording as output signal is in red. As can be observed in Fig.16, it shows the synchronization for the input and output sound. Few samples still have certain delay.
3.3.3 Identify the model

Since the synchronization is done, so $n_k$ is zero, which is the corresponding integer value. Then the system order becomes $[n_a \ n_b \ 0]$. Apply the method in 2.6 to find the system order. Use the first half of input and output signals to do system identification.

The model errors are presented in 3D-graph figures below. Theoretically, the model error decrease as the model order increase. The smaller the model error, the better the estimated model is. The estimated model more approach to the true system. A model with positive model error is a destructive model. Positive model error destroys the system, more damage than good.

The model error will be represented by db. 0 db is a limitation, since $\text{error} = y - G(u)$, 0 db means $\text{error} = y$. In the case, the model error must be less than 0 db.

As shown in Fig.17, it is the identified ARX model. It represents the model error in both $n_a$ and $n_b$ dimensions. The lowest model error is about -8 db, which is good enough for the system. The model error decrease as $n_a$ and $n_b$ increase. Small
model error means good system, so \( n_a = 28 \), and \( n_b = 30 \) are chosen for the ARX model of the third set of data.

![Figure 17 model of estimation from the third data set](image)

Once the order of the model is chosen, model estimation is needed to be done. The method is to compare the model output with the measured output, or listen to the wanted music. The results will be presented in the next chapter.
4 Results

System identification is a method to find the values of parameters in a particular model structure. In this thesis, system identification is used to find the parameters in linear ARX model structure. In this chapter, the mathematical equations will be presented respectively according the linear ARX model order \( n_a \ n_b \) by the frequency response, and the comparison of model output with actual output will be shown as well.

4.1 First set of data

For the first set of data, two different spectral content of input signal are used to identify two different models. One model is identified by using the low frequency spectral content, which is piano input. Another model is identified by using the high frequency spectral content, which is chirp input. Since the synchronization is done before, so \( n_k \) is zero, which is the corresponding integer value. Then the model order becomes \([n_a \ n_b \ 0]\). The linear ARX model order of piano input is \([13 \ 16 \ 0]\), the linear ARX model order of chirp input is \([17 \ 16 \ 0]\).

4.1.1 Model for piano sound

Fig.18 shows the frequency response of the piano model from the first set of data. As can be observed, the value of magnitude is less than 10 db.
Figure 18 the frequency response of piano model from the first data set

4.1.2 Model for chirp sound

Fig.19 shows the frequency response of the chirp model from the first set of data. As can be observed, the value of magnitude is less than 5 db.

Figure 19 the frequency response of chirp model from the first data set
4.1.3 Compare the piano model and chirp model

Because system identification uses an input signal which could excite the system properly and adequately, so choosing which spectral content of input signal is very important for system identification. Since either piano sound input or chirp sound input is used to identify two systems, piano sound is a low frequency spectral content, the linear ARX model for piano sound input has an order [13 16 0], chirp sound is a high frequency spectral content, the linear ARX model for chirp sound has an order [17 16 0]. These two models are compared, to see which one is approach to the true system.

Fig.20 shows the simulated piano output of the piano model. Fig.21 shows the simulated piano output of the chirp model. Fig.22 shows the actual piano output. By comparing both Fig.20 and Fig.21 to Fig.22, it is easy to see, the Fig.20 is very similar to Fig.22. Which means the output disturbing piano sound can be cancelled. This is also can be testified, when I listening to the sound after doing cancellation, only music can be heard. So the piano model with an order [13 16 0] can cancel the disturbing piano sound without affecting the music.

![Figure 20 piano output of piano model from the first data set](image-url)
Fig. 21 piano output of chirp model from the first data set

Fig. 22 actual piano output from the first data set

Fig. 23 shows the simulated chirp output of the piano model. Fig. 24 shows the simulated chirp output of the chirp model. Fig. 25 shows the actual chirp output. By comparing both Fig. 23 and Fig. 24 to Fig. 25, the Fig. 24 is more similar to Fig. 25. This is also can be testified, when I listening to the sound after doing cancellation, the chirp model with an order $[17 \ 16 \ 0]$ can cancel more chirp sound.
Figure 23 chirp output of piano model from the first data set

Figure 24 chirp output of chirp model from the first data set
4.2 Second set of data

By using the linear ARX model structure to identify the system for the second set of data, unfortunately, the model is destructive. The second set of data cannot apply the linear ARX model structure to identify a system, since it is a non-linear system. System Identification Tool Box [1] can be used to find out the system.

4.3 Third set of data

The third set of data could identify a good linear model, before doing the system identification, it is necessary to determine the time delay, this should always be done before identifying the system, and the reason to do this is to make sure the system you identified is more precise and valid.

Fig.26 shows the frequency response of the transfer function from the third set of data. As can be observed, the value of magnitude is less than 0 db.
Fig.26 the frequency response of transfer function from the third data set

Fig.27 shows the simulated output from the identified model. Fig.28 shows the actual output from the third data set. By comparing these two figures, shown in Fig.29, the simulated output is very similar to the actual output, which means the disturbing sound can be cancelled mostly. This is also can be testified, when I listening to the sound after doing cancellation, only music can be heard. Synchronization is done before. So the linear ARX model with an order [28 30 0] can cancel the disturbing sound without affecting the music.
Figure 27 model output from the third data set

Figure 28 actual output from the third data set

Figure 29 comparison of model output with actual output
5. Discussion

Two different models were identified from the first data set, by using different frequency spectral content of the input signal. Chirp model is with a higher order than piano model. Because of chirp sound contains wide spectrum, so more parameters need to be used for a good model.

There are challenges for this research topic, the first challenge is to determine the mathematical equation, since the system is unknown, so there is a need to have some given data and combine my experience to formulate the mathematical equation. Since the input and output signals are used to capture the parameters of the system, so the data must be very reliable, and the number of time lags between input and output signals need to be decided before identifying the system. In order to measure the right variables with sufficient accuracy, it must to use the right frequency spectral content of the input signal to find the model parameters. That could make sure excite the system properly and adequately, and also measure the input data long enough to capture the important parameters of the system.

The second challenge is if the model error between the output of identified model and the actual output could approach the minimum possible model error in some models within the given linear ARX model. The answer is “Yes”, theoretically, the high order of the system, the less model error the better system. The minimum possible model error could be \(-\infty\) db. In practice, model error of the model cannot approach \(-\infty\) db. If the model error is smaller than 0 db, the identified model is advisable. As can be observed in Fig.6, Fig.7, Fig.17, the model error decreases, when increase the model order. So the higher the model order, the better the system is [4].

The third challenge is if assuming that the minimum possible estimation error is achievable by one or more best possible models, and then there is a question, does the output identification model approach one of these models? The answer is
“yes”, no matter what kind of model you use, if the estimation error is small enough, the model is good [4]. The problem that could have are the noise and nature sound could be recorded into microphone, which could cause some echo or disturbing, this will affect the model of estimation. So there is a great need to remove this disturbing and undesired noise, but in practice, this noise cannot be completely removed.

How to implement the identified model in the real application? The point is that, the mathematical equation is possible to be implemented in the real application. An efficient technique for the implementation of the widely used orthogonal transforms when their input samples are originating from a tap delay line (TDL), the transformation need to be performed after each new sample enters the TDL, is addressed [5]. The model is a linear system of equation, when implementing the model on hardware, the recursive least-squares method can be used to reduce requirement memory and computational resources. Since filtering is a standard function for DSP: s, FPGA: s and microprocessors, which means that should not cause too much of issue. The data rates are fairly low, it is at a maximum 44100 samples/s, with devices running at speeds of at least 10’s MHz, so this should not be much of a problem.

The disturbing sound cancellation technique has a great marketability. For example, this noise cancellation technique can be used in microphone design, and headphones, and studio work. There is news [6]; Wolfon introduced a mass-market noise cancelling stereo headset using the ANC technology, on Wednesday, February 10, 2010. They can cancel up to 99% of noise in low frequency. There is another market for the flight plane, a noise cancelling headphone is necessary to use in the airplane.
6. Conclusions

The purpose of this thesis is to formulate a mathematical equation, by using MATLAB to indentify a system, then using the system to do disturbing sound cancellation. The noise in the data sets is not considered in this thesis. Fortunately, the piano model from the first data set could cancel the disturbing sound without affecting the wanted sound. The chirp model from the first data set could cancel a big part of disturbing sound. The model from the third data set could also cancel the most of disturbing sound without affecting the music. But for the second set of data, the identified model is destructive. The second set of data cannot use the linear ARX model structure to identify a system.

The principle behind the system identification is that, design a model equation based on the current model of the unknown system, which is sufficiently close to the true system. Since the model is identified, implementing the model on hardware could be a future work.
7. References


