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PAPER

Development of a software tool for eliminating nonlinear distortion

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Abstract: In this paper, we present the effectiveness of a software tool for eliminating nonlinear distortions of loudspeaker systems through some experiments. The nonlinear distortions affect the sound quality of loudspeaker systems. We have presented some identification methods of loudspeaker systems and some compensation methods of nonlinear distortions. However, the software tool for compensating the distortions has not been realized yet. We therefore develop the software tool. Experimental results show that the 2nd- and 3nd-order nonlinear distortions of a loudspeaker system can be reduced in the range of 10 [dB] to 20 [dB] by the developed tool when noise level is below 70 [dBA].

Keywords: Loudspeaker, Nonlinear distortions, Volterra filter, Eliminating distortions

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1. INTRODUCTION

The fundamental principle of loudspeaker systems has never been changed since the invention. The loudspeaker systems have so complex structure to transform electric signal into mechanical vibration and to radiate acoustic wave so that the loudspeaker system produces linear and nonlinear distortions. These distortions consequently deteriorate the sound quality. Recently, small-sized loudspeaker systems for desktop have come into wide use as personal computers (PC) advance. Since these size gets smaller than general speaker systems inevitably, the level of nonlinear distortions get bigger generally. Furthermore, it has been reported that the nonlinear distortions have influence for auditory sense in high-sampling audio [1,2]. Hence, the importance of eliminating the nonlinear distortion has been increasing in recent years.

Identifying the linear and nonlinear elements of loudspeaker systems and designing a nonlinear inverse system are essential to eliminate these distortions. We have demonstrated the effectiveness of an elimination method using Volterra filters [3]. In this method, the Volterra kernels of a target loudspeaker system are identified by adaptive Volterra filters [4] and the nonlinear distortion is eliminated by a nonlinear inverse system using the identified Volterra kernels [5]. By the way, since the tendency listening to music with PC has been increasing, user's demands of eliminating the nonlinear distortion with PC have been increasing. However, a software tool for eliminating the nonlinear distortion has not been realized yet. We have therefore developed the software tool, which can identify loudspeaker systems, design the corresponding nonlinear inverse system, and compensate the nonlinear distortions. Experimental results demonstrate the effectiveness of the software tool.

2. ELIMINATION METHODS OF NONLINEAR DISTORTION USING VOLTERRA FILTER

In this chapter, we explain how to eliminate the nonlinear distortion of a loudspeaker system using Volterra filters.

2.1. Discrete Volterra Series

Nonlinear systems such as loudspeaker systems can be modeled by using the Volterra series expansion. Since this paper treats the 3rd-order element and assumes that the Volterra kernels have a finite memory length N, the input-output relation of the systems is represented by

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$$y(n) = \sum_{k_1=0}^{N-1} h_1(k_1)x(n-k_1) + \sum_{k_1=0}^{N-1}\sum_{k_2=0}^{N-1} h_2(k_1,k_2)x(n-k_1)x(n-k_2) + \sum_{k_1=0}^{N-1}\sum_{k_2=0}^{N-1}\sum_{k_3=0}^{N-1} h_3(k_1,k_2,k_3)x(n-k_1)x(n-k_2)x(n-k_3) \quad (1)$$

where $x(n)$ and $y(n)$ are the discrete input and output signals, respectively; $h_1(k_1)$, $h_2(k_1,k_2)$, and $h_3(k_1,k_2,k_3)$ are the 1st-, 2nd-, and 3rd-order discrete Volterra kernels, respectively. The Volterra kernels satisfy the symmetry property; therefore, these kernels are invariant regard less of the order of its terms, without loss of generality. For example, the 3rd-order Volterra kernel has the property as

$$\begin{aligned} h_3(k_1,k_2,k_3) &= h_3(k_1,k_3,k_2) = h_3(k_2,k_1,k_3) \\ &= h_3(k_2,k_3,k_1) = h_3(k_3,k_1,k_2) = h_3(k_3,k_2,k_1) \end{aligned} \quad (2)$$

2.2. Discrete Fourier Transform of Volterra Series

Discrete Fourier transform (DFT) of Eq. (1) is given by

$$\begin{aligned} Y(m) &= H_1(m)X(m) \\ &+ A_1[H_2(m_1,m_2)X(m_1)X(m_2)] \\ &+ A_2[H_3(m_1,m_2,m_3)X(m_1)X(m_2)X(m_3)] \end{aligned} \quad (3)$$

where $X(m)$ and $Y(m)$ are the N point DFTs of $x(n)$ and $y(n)$, respectively; $H_1(m)$, $H_2(m_1,m_2)$, and $H_3(m_1,m_2,m_3)$, which are called the 1st-, 2nd-, and 3rd-order Volterra Frequency Responses (VFR), are those of $h_1(k_1)$, $h_2(k_1,k_2)$, and $h_3(k_1,k_2,k_3)$, respectively. Furthermore, the VFR also satisfy the symmetry property due to the symmetry of the Volterra kernels. For example, the 3rd-order VFR has the property as

$$\begin{aligned} H_3(m_1,m_2,m_3) &= H_3(m_1,m_3,m_2) = H_3(m_2,m_1,m_3) \\ &= H_3(m_2,m_3,m_1) = H_3(m_3,m_1,m_2) = H_3(m_3,m_2,m_1) \end{aligned} \quad (4)$$

Moreover, the VFR satisfies the conjugate symmetry property. For example, the 3rd-order VFR has the property as

$$H_3(m_1,m_2,m_3) = H_3{}^*(N-m_1, N-m_2, N-m_3) \quad (5)$$

A_1 and A_2 are called the reduction operators. The reduction operator has the role of mapping a function with multi-dimensional dependent variables to a function with one-dimensional variables [6]. A_1, which maps a function with two-dimensional dependent variables to that with one-dimensional variables, is represented as

$$\begin{aligned} Y(m) &= A_1[Y_2{}'(m_1,m_2)] \\ &= \frac{1}{N}\sum_{m_1+m_2=m \text{ or } m+N} Y_2{}'(m_1,m_2) \end{aligned} \quad (6)$$

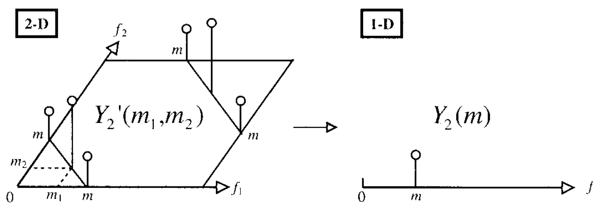


Fig. 1 Reduction operator.

where $Y(m)$ and $Y_2{}'(m_1,m_2)$ are the function with one- and two-dimensional dependent variables, respectively. Figure 1 shows the reduction operator. In Eq. (6), the spectra $Y_2{}'(m_1,m_2)$ at the frequencies satisfying $m_1+m_2=m$ or $m+N$ are summed, so that the spectrum $Y(m)$ at the frequency m can be obtained. Similarly, A_2, which maps a function with three-dimensional dependent variables to that with one-dimensional variables, is represented as

$$\begin{aligned} Y(m) &= A_2[Y_3{}'(m_1,m_2,m_3)] \\ &= \frac{1}{N^2}\sum_{m_1+m_2+m_3=m \text{ or } m+N \text{ or } m+2N} Y_3{}'(m_1,m_2,m_3) \end{aligned} \quad (7)$$

where $Y_3{}'(m_1,m_2,m_3)$ are the function with three-dimensional dependent variables. In Eq. (7), the spectra $Y_3{}'(m_1,m_2,m_3)$ at the frequencies satisfying $m_1+m_2+m_3=m$ or $m+N$ or $m+2N$ are summed, so that the spectrum $Y(m)$ at the frequency m can be obtained.

2.3. Identification of Loudspeaker System Using Frequency Response Method [7]

To obtain the 3rd-order VFR, three sinusoidal waves, whose frequencies are m_1, m_2, and m_3 ($m_1 < m_2 < m_3$), are input into a nonlinear system. Then, the VFR is determined by substituting spectra of input signal $X(m_1)$, $X(m_2)$ and $X(m_3)$ and output signal for

$$H_3(m_1,m_2,m_3) = \frac{Y(m_1+m_2+m_3)}{X(m_1)X(m_2)X(m_3)}\frac{N^2}{a} \quad (8)$$

where the coefficient a is the number of the symmetry of the VFR. Moreover, the 3rd-order VFR is determined by repeating this procedure at various frequencies. Since sinusoidal waves are input, this method has much superior identification accuracy.

2.4. Nonlinear Inverse System

Figure 2 shows the structure of a nonlinear inverse system to eliminate the 2nd- and 3rd-order distortions. In Fig. 2, the linear inverse filter h_1^{-1} satisfies $h_1 \cdot h_1^{-1} = z^{-\Delta}$. To design the nonlinear inverse system, the linear inverse filter h_1^{-1} and the 2nd- and 3rd-order Volterra kernels h_2, h_3 are needed. Furthermore, those accuracies affect the performance of the nonlinear inverse system.

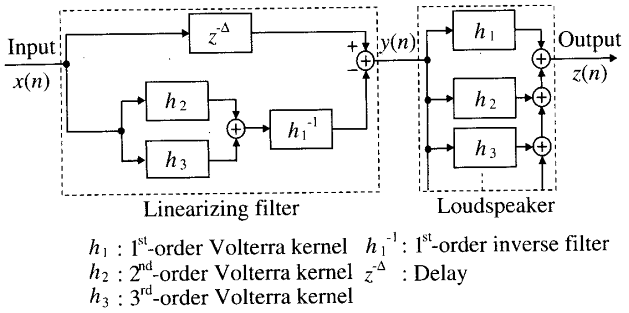
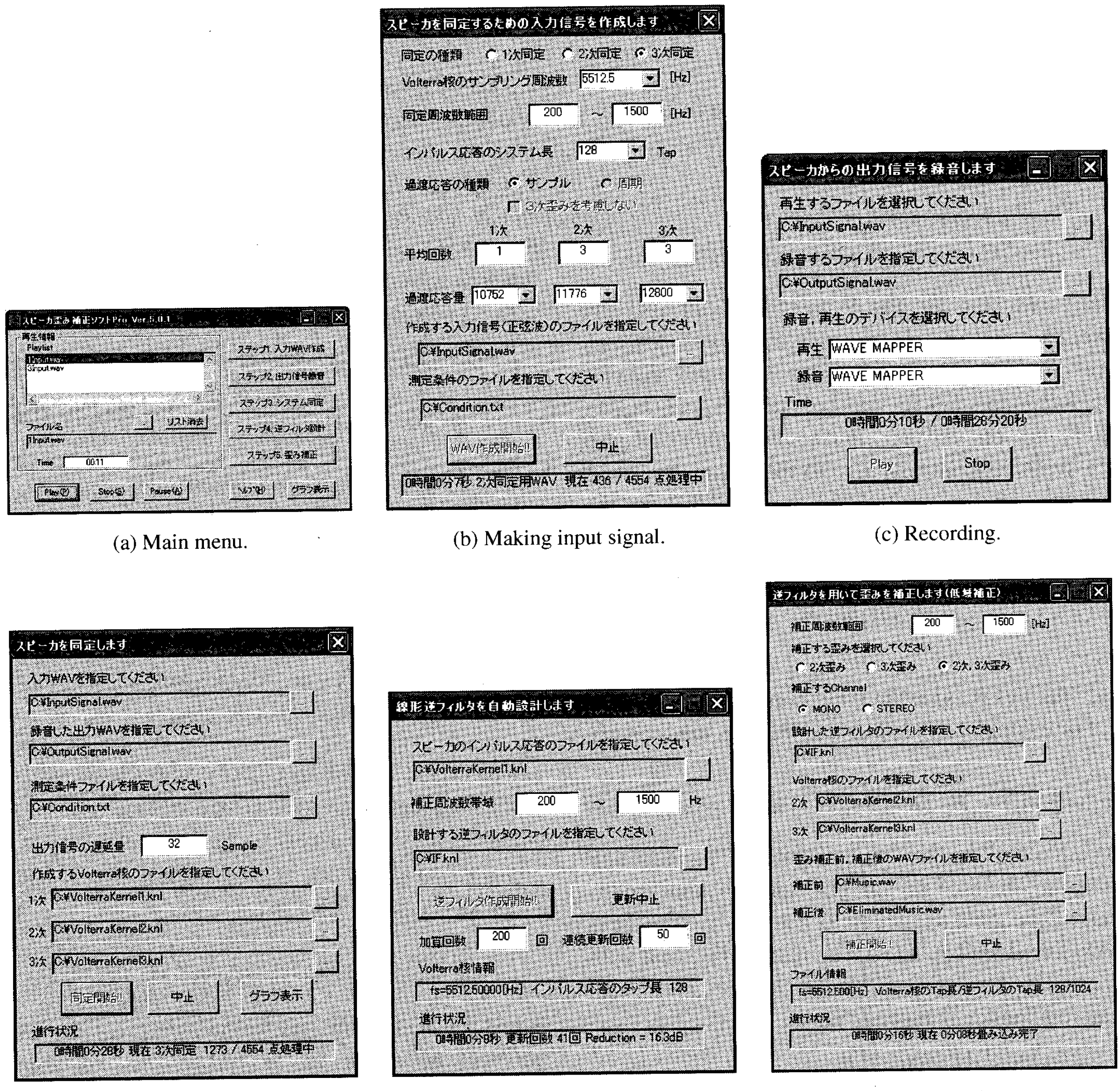


Fig. 2 The structure of a nonlinear inverse system.

3. SOFTWARE TOOL FOR ELIMINATING NONLINEAR DISTORTION

In this chapter, we explain how to use the developed software tool. We purpose eliminating the distortions of small-sized loudspeaker system for PC by software. The main menu is shown in Fig. 3(a). Functions of the software tool are as follows;

- Identification of loudspeaker system
- Design of linear inverse filter



(a) Main menu. (b) Making input signal. (c) Recording.

(d) Identification. (e) Making linear inverse filter. (f) Elimination.

Fig. 3 Functions of the tool.

- Eliminating distortions
- Playback WAV files, and so on

3.1. Making Input Signal (Fig. 3(b))

We can make an input signal for the identification as a WAV file. We need to set up the order of Volterra kernels, frequency range, sampling frequency, tap length, the number of average, and so on. Furthermore, the record conditions are made as text file. Moreover, the sampling frequency can be flexibly set in a frequency range lower than 44,100 [Hz] and the maximum frequency to be able to eliminate the nonlinear distortion becomes consequently below half of the sampling frequency.

3.2. Recording Output Signal (Fig. 3(c))

We can set up playback and record devices freely and record the output signal of a loudspeaker system.

3.3. Identification of Volterra Kernels (Fig. 3(d))

By setting up the WAV files of the input signal made in Sect. 3.1 and the output signal recorded in Sect. 3.2, we can obtain the Volterra kernels.

3.4. Design of Linear Inverse Filter (Fig. 3(e))

We can design the linear inverse filter by using the 1st-order Volterra kernel in Sect. 3.3. By using Frequency-Domain Summational Normalized Block LMS (FDSNBLMS) algorithm [8], only the filter coefficients within a target frequency range is updated. Futhermore, the stepsize parameter is set up automatically and the design of the linear inverse filter is finished when the reduction, which is defined as Eq. (9), becomes maximum.

$$Reduction = 10\log_{10}\left[\sum_{m=m_1}^{m_2} D(m)^2 \Big/ \sum_{m=m_1}^{m_2} \{D(m) - Y(m)\}^2\right] \tag{9}$$

$D(m)$: Spectrum of desired signal.
$Y(m)$: Spectrum of signal thorough linear inverse filter.
Eliminating frequency range: m_1–m_2 [Hz]

3.5. Eliminating Distortions (Fig. 3(f))

By setting up a playback WAV file, the linear inverse filter, and the Volterra kernels, we can make the corresponding WAV file to make the 2nd- and 3rd-order distortions eliminate.

4. EXPERIMENTS OF ELIMINATING NONLINEAR DISTORTIONS

Next, we experiment with eliminating the 2nd- and 3rd-order nonlinear distortions by the developed software. Tables 1 and 2 shows the measurement conditions and the identification conditions. First, we measure the Volterra kernels of a loudspeaker system and design the linear inverse filter. Next, we play back sinusoidal waves for a test and the corresponding output waves for the nonlinear inverse system, then record the output signals of the loudspeaker system. Finally, we calculate the frequency responses of harmonic and intermodulation elements by using Eqs. (10)–(14).

Table 1 Measurement conditions.

CPU	Pentium IV 2.0AGHz
Main memory	DDR-SDRAM 1GB
OS	Windows XP
Sound card	On board
Microphone	MM-MC5 (SANWA SUPPLY)
Loudspeaker	MM-SP80SV (SANWA SUPPLY)

Table 2 Identification conditions.

Sampling frequency	5,512.5 [Hz]
Frequency range	200–1,500 [Hz]
Tap length of 1st-order kernel	256
Tap length of 2nd-order kernel	128
Tap length of 3rd-order kernel	128

$$H_{2m}(m) = \frac{Y(2m)}{X(m)} \tag{10}$$

$$H_{m_2 \pm m_1}(m) = \frac{Y(m_2 \pm m_1)}{X(m_2)} \tag{11}$$

$$H_{3m}(m) = \frac{Y(3m)}{X(m)} \tag{12}$$

$$H_{m_2 \pm 2m_1}(m) = \frac{Y(m_2 \pm 2m_1)}{X(m_2)} \tag{13}$$

$$H_{m_3 \pm (m_1 + m_2)}(m) = \frac{Y(m_3 \pm (m_1 + m_2))}{X(m_3)} \tag{14}$$

In Eqs. (11) and (13), m_1 is kept at 215 [Hz] and m_2 is swept in the frequency range of 258 [Hz] to 1,464 [Hz]. Similarly, m_1 and m_2 are kept at 215 [Hz] and 258 [Hz] respectively, and m_3 is swept in the frequency range of 301 [Hz] to 1,464 [Hz] in Eq. (14). Here, the noise level is 50 [dBA] in both the identification and elimination and the playback sound pressure level is 92 [dB]. Noise level is defined as follows.

$$\text{Noise level [dBA]} = 10\log_{10}\frac{P_A{}^2}{P_0{}^2} \tag{15}$$

$P_A{}^2$: A-weighted sound pressure level.
$P_0{}^2$: Standard sound pressure level (20 [μPa]).

Figure 4(a)–(h) shows that the frequency responses of a loudspeaker system and the 2nd- and 3rd-order nonlinear distortions before and after elimination. We also calculate the frequency response of the 4th-order harmonic distortion as shown in Fig. 4(i) because the nonlinear inverse system

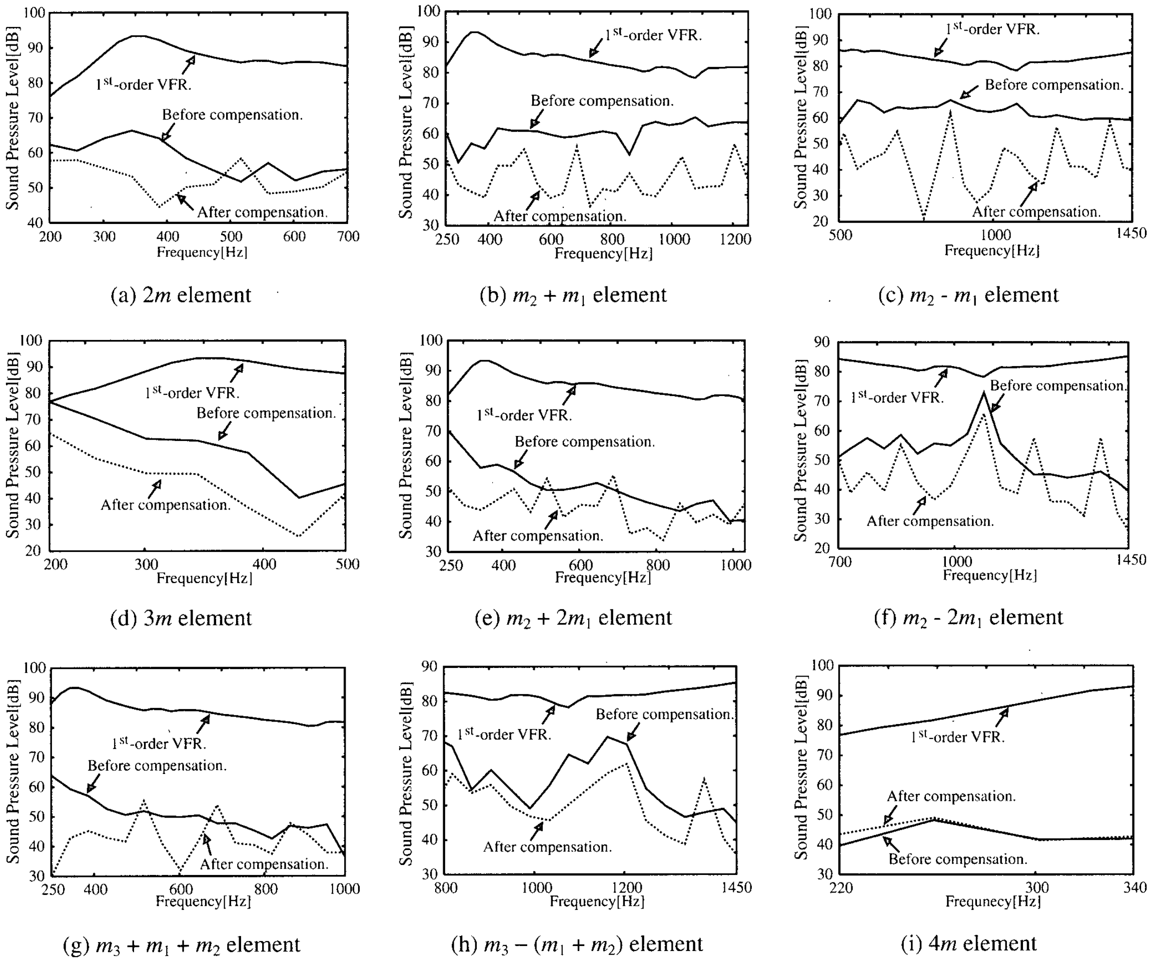


Fig. 4 Frequency responses of a loudspeaker system and the nonlinear distortions before and after elimination.

in Fig. 2 may make the higher-order distortions increase. In Fig. 4, x-axis and y-axis express the swept frequency and the sound pressure level respectively. It can be seen from Fig. 4 that the levels of the 2nd- and 3rd-order nonlinear distortions are reduced in the range of 10 [dB] to 20 [dB] compared with before elimination over the whole frequency band. Moreover, all nonlinear distortion levels after compensation are lower than the background noise level. The increase of the 4th-order harmonic distortion is also a little. These results demonstrate the effectiveness of the developed software tool.

Next, we experiment with eliminating the 2nd- and 3rd-order nonlinear distortions using the Volterra kernels identified under some noisy environments. Here, we use white Gaussian noise to make noisy environments. Table 3 shows the effects of eliminating nonlinear distortions. In

Table 3 Effectiveness of eliminating nonlinear distortions.

Noise level	50 [dBA]		60 [dBA]		70 [dBA]		80 [dBA]	
Elements	Points	Effect [dB]	Points	Effect [dB]	Points	Effect [dB]	Points	Effect [dB]
$2m_1$	1/12	5.9	3/12	5.3	2/12	7.4	3/12	2.6
$m_2 + m_1$	0/24	15.4	0/24	13.8	0/24	10.4	2/24	7.0
$m_2 - m_1$	0/24	19.6	0/24	17.2	1/24	12.3	6/24	9.0
$3m_1$	0/7	13.0	1/7	11.8	0/7	9.7	2/7	7.0
$m_2 + 2m_1$	4/19	6.7	3/19	5.9	7/19	1.4	13/19	−2.1
$m_2 - 2m_1$	2/19	8.2	5/19	5.3	8/19	0.7	14/19	−5.8
$m_3 + m_1 + m_2$	4/18	7.8	4/18	5.1	9/18	0.9	13/18	−4.4
$m_3 - (m_1 + m_2)$	1/18	6.9	1/18	6.7	8/18	4.1	9/18	−2.0
Average effect	10.4 [dB]		8.9 [dB]		5.9 [dB]		0.3 [dB]	

Table 3, 'Points' is the number of points increasing in the distortion per the number of evaluating points; 'Effect' is an average eliminated quantity to all evaluating points per element and 'Average effect' is an average eliminated quantity to all elements per 'Noise level'. The negative value for 'Effect' means the increase of the distortion. It can be seen from Table 3 that the effect is smaller as the increase of noise level. These results demonstrate nonlinear distortions of a loudspeaker system can be reduced fully when the noise level is below 70 [dBA]. Therefore, this software is available for general environment, such as home use of personal computers with general devices, because 70 [dBA] is the same as the sound pressure level of the telephone ring.

In this experiment, it took about 40 minutes to identify a loudspeaker system, 3 minutes to design the linear inverse filter, and about 5 minutes to compensate a 4-minutes WAV file, respectivelly.

5. CONCLUSIONS

In this paper, we have developed a software tool for eliminating nonlinear distortions. The experimental results have demonstrated the effecitiveness of the software tool. Effectiveness of eliminating nonlinear distortions of loudspeaker system in high-sampling audio are a subject to study in the future.

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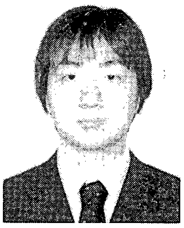

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