LINEARIZATION OF LOUDSPEAKER SYSTEMS USING MINT AND VOLTERRA FILTERS

Yasuo Nomura and Yoshinobu Kajikawa

Department of Electronics, Faculty of Engineering, Kansai University
3-3-35, Suita-shi, Osaka 564-8680, Japan

ABSTRACT

In this paper, we propose a linearization (compensation of nonlinear distortion) method for loudspeaker systems using the MINT (Multiple-input/output INverse-filtering Theorem) and Volterra filters. In the proposed method, linear inverse filtering of a target loudspeaker system is realized by using the MINT so that exact linear inverse filtering can be realized. The linearization performance becomes consequently very high. On the other hand, since the conventional linearization method cannot realize exact linear inverse filtering, the performance deteriorates remarkably. Experimental results demonstrate that the proposed method has about 20dB higher performance than the conventional one.

1. INTRODUCTION

Recently, digital audio systems have been spreading. In the digital audio systems, some distortions occurring in the transmission paths have been reduced significantly, and the sound quality has been improved considerably. However, loudspeaker systems, which are a human interface in the digital audio systems, have a lot of distortions, especially, nonlinear distortions. The performance of the whole digital audio systems consequently deteriorates. Hence, the compensation of the nonlinear distortions (linearization of loudspeakers) is a very important issue in the digital audio systems.

The compensation (linearization) can be achieved by using a Volterra filter [1, 2], which identifies the nonlinearity of a target loudspeaker system, and a linear inverse filter, which compensates the linear distortion [3, 4, 5, 6]. One of some factors influencing the compensation performance is the estimation accuracy of the Volterra filter. However, this estimation accuracy can be made high by using an identification method employing multi-sinusoidal waves [5, 6]. Another factor is the design accuracy of the linear inverse filter to compensate linear distortions. In other words, whether exact linear inverse filtering can be realized influences the compensation performance. However, the exact linear inverse filtering cannot be realized because loudspeaker systems have nonminimum phases. In this case, only an approximate inverse filtering is realized. If the approximate accuracy is low, the compensation performance of nonlinear distortions deteriorates remarkably. We therefore propose a novel linearization method. In the proposed method, we use the MINT (Multiple-input/output INverse-filtering Theorem) [7], which can realize an exact linear inverse of a target acoustic system. The compensation performance of nonlinear distortions is consequently very high.

2. CONVENTIONAL LINEARIZATION METHOD AND ITS PROBLEM

Figure 1 shows a block diagram of a conventional linearization system, which can compensate the nonlinear distortions of loudspeaker systems. In Fig. 1, $D_1(z)$ and $D_2(z_1, z_2)$ represent the transfer functions of the first- and second-order Volterra kernels of a loudspeaker system, respectively. $\hat{D}_2(z_1, z_2)$ is a Volterra filter to model the second-order Volterra kernel of the loudspeaker system, and $H_1(z)$, which is a linear inverse filter of $D_1(z)$, is designed so as to satisfy the following condition.

$$D_1(z)H_1(z) = z^{-\Delta} \tag{1}$$

The second-order nonlinear transfer function of the whole system is consequently represented by the following equation.

$$\begin{aligned} & D_2(z_1, z_2)z^{-\Delta} - D_1(z)H_1(z)\hat{D}_2(z_1, z_2) \\ &= D_2(z_1, z_2)z^{-\Delta} - z^{-\Delta}\hat{D}_2(z_1, z_2) \\ &= \{D_2(z_1, z_2) - \hat{D}_2(z_1, z_2)\}z^{-\Delta} \\ &= 0 \end{aligned} \tag{2}$$

If $\hat{D}_2(z_1, z_2)$ is equal to $D_2(z_1, z_2)$ of the loudspeaker system and $H_1(z)$ is designed so as to satisfy the condition shown in Eq.(1), the nonlinear inverse system can completely compensate the second-order nonlinear distortion. The high accuracy $\hat{D}_2(z_1, z_2)$ can be obtained if narrow band signals are used to model $D_2(z_1, z_2)$. On the contrary, $H_1(z)$ to satisfy the condition of Eq.(1) can exist if and only if $D_1(z)$ is a minimum phase function. However, the acoustical trans-

This research was financially supported by MEXT KAKENHI(14750320).

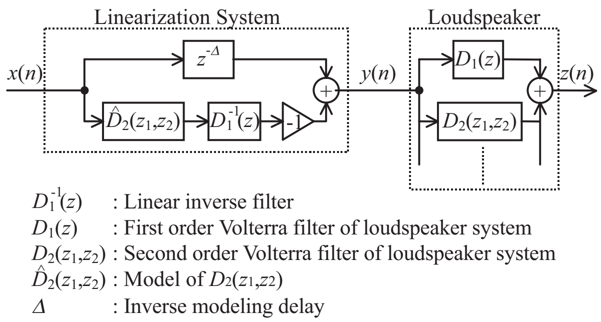


Fig. 1. Block diagram of the conventional linearization system.

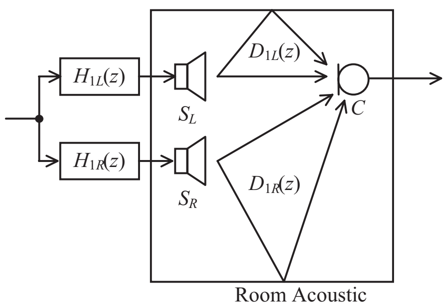


Fig. 2. Sound field inverse filtering using the MINT.

fer function $D_1(z)$ is generally considered to be a nonminimum phase function. Hence, only an approximate inverse filter is obtained. It is therefore very difficult from Fig. 1 to compensate (cancel) $D_2(z_1, z_2)$ completely because $H_1(z)$ does not satisfy Eq.(1). Accordingly, the performance of linearization system is greatly influenced by whether exact linear inverse filtering can be realized.

3. LINEARIZATION METHOD OF LOUDSPEAKER SYSTEMS BY USING MINT

3.1. MINT [7]

In this section, we explain the MINT(Multiple-input/output INverse-filtering Theorem), which can realize an exact linear inverse of a target acoustic system. Consider the acoustic system shown in Fig. 2. In Fig. 2, the transfer function $D_{11}(z)$ from loudspeaker S_1 to receiving point C is defined by

$$D_{11}(z) = z^{-u} d_{11}(z) \tag{3}$$

where z^{-u} is the time delay between S_1 and C, $d_{11}(z)$ the M'th order polynomial of z^{-1}, which represents reflection sound effects. The transfer function $D_{21}(z)$ from loudspeaker S_2 to receiving point C is also defined by

$$D_{21}(z) = z^{-(u+w)} d_{21}(z) \tag{4}$$

where $z^{-(u+w)}$ is the time delay between S_2 and C, $d_{21}(z)$ the N'th order polynomial of z^{-1}. To realize inverse filtering of the system, $H_{11}(z)$ and $H_{21}(z)$ must satisfy the expression

$$1 = d_{11}(z)H_{11}(z) + z^{-w} d_{21}(z)H_{21}(z) \tag{5}$$

This relationship is called Diophantine equation. The solutions for this equation exist if and only if $d_{11}(z)$ and $z^{-w}d_{21}(z)$ are relatively prime (in other words, do not have any common zero in the z-plane). The solutions is expressed by

$$\begin{aligned} H_{11}(z) &= H_{11,min}(z) + z^{-w} d_{21}(z)Q(z) \\ H_{21}(z) &= H_{21,min}(z) - d_{11}(z)Q(z) \end{aligned}$$

where $Q(z)$ is an arbitrary polynomial. $H_{11,min}(z)$ and $H_{21,min}(z)$ are the only pair of the minimum order solution that satisfies Eq.(5) and the orders have the following relation.

$$\begin{aligned} deg\ H_{11,min}(z) &< deg\ z^{-w} d_{21}(z) = N + w \\ deg\ H_{21,min}(z) &< deg\ d_{11}(z) = M \end{aligned}$$

The property of the Diophantine equation is not concerned with whether $d_{11}(z)$ and $z^{-w}d_{21}(z)$ are nonminimum phase functions. If some symmetrical positions of loudspeakers and a microphone are avoided, $d_{11}(z)$ and $z^{-w}d_{21}(z)$ does not have a common zero. Hence, exact inverse filtering is realized.

Next, we describe the computation of $H_{11,min}(z)$ and $H_{21,min}(z)$. Figure 3 shows a system arrangement to obtain $H_{11,min}(z)$ and $H_{21,min}(z)$ by using adaptive filters. First, the transfer functions of $D_{11}(z)$ and $D_{21}(z)$ are modeled beforehand. Next, as shown in Fig. 3, two adaptive filters $\hat{H}_{11,n}(z)$, $\hat{H}_{21,n}(z)$ are connected to the outputs of the

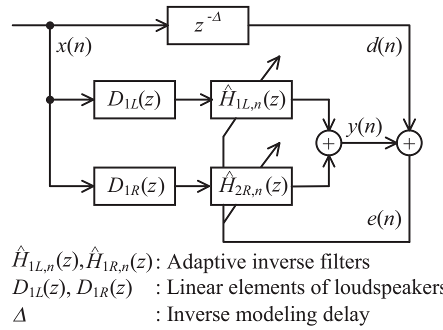
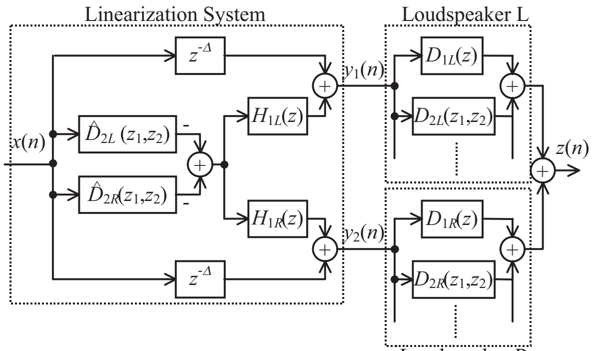


Fig. 3. Block diagram of identification method for $H_{11,min}(z)$ and $H_{21,min}(z)$ by using adaptive filters.



$H_{1L}(z)$, $H_{1R}(z)$: Linear inverse filters
$D_{1L}(z)$, $D_{1R}(z)$: First order Volterra filters of loudspeakers
$D_{2L}(z_1,z_2)$, $D_{2R}(z_1,z_2)$: Second order Volterra filters of loudspeakers
$\hat{D}_{2L}(z_1,z_2)$, $\hat{D}_{2R}(z_1,z_2)$: Models of $D_{2L}(z_1,z_2)$ and $D_{2R}(z_1,z_2)$
Δ : Inverse modeling delay

Fig. 4. Block diagram of the proposed linearization system using the MINT.

modeled transfer functions. Finally, the coefficients of the two adaptive filters are updated as minimizing the following error signal.

$$e(n) = x(n - \Delta) - y(n) \tag{6}$$

With the above procedure, you can obtain the filters to realize exact linear inverse filtering.

3.2. Linearization System Using MINT

In this section, we introduce a system arrangement to apply the MINT to the linearization system, which can linearize nonlinear systems.

Figure 4 shows the block diagram of the proposed system. In Fig. 4, $H_{11}(z)$ and $H_{21}(z)$ are FIR filters in the MINT as explained in the previous section. The relation of these filters is shown in the following equation again.

$$D_{11}(z)H_{11}(z) + D_{21}(z)H_{21}(z) = z^{-\Delta} \tag{7}$$

Hence, the second-order nonlinear property of the whole system in Fig. 4 is represented by

$$\begin{aligned} &\{D_{12}(z_1, z_2) + D_{22}(z_1, z_2)\}z^{-\Delta} \\ &\quad -\{D_{11}(z)H_{11}(z) + D_{21}(z)H_{21}(z)\} \\ &\quad \cdot\{\hat{D}_{12}(z_1, z_2) + \hat{D}_{22}(z_1, z_2)\} \\ = &\quad z^{-\Delta}\{D_{12}(z_1, z_2) + D_{22}(z_1, z_2) \\ &\quad -\hat{D}_{12}(z_1, z_2) - \hat{D}_{22}(z_1, z_2)\} \end{aligned} \tag{8}$$

If

$$D_{12}(z_1, z_2) = \hat{D}_{12}(z_1, z_2), D_{22}(z_1, z_2) = \hat{D}_{22}(z_1, z_2), \tag{9}$$

Table 1. Experiment conditions

Sampling frequency	44100[Hz]
Tap length of $D_{1L}(z)$	512
Tap length of $D_{1R}(z)$	512
Tap length of $H_{1L}(z)$	511
Tap length of $H_{1R}(z)$	511
Tap length of $D_1^{-1}(z)$	2048
Tap length of 2nd-order models	256
Delay of the proposed system	256
Delay of the conventional system	1024
Input Voltage	6.0V
Loudspeaker system	MM-SP102SV

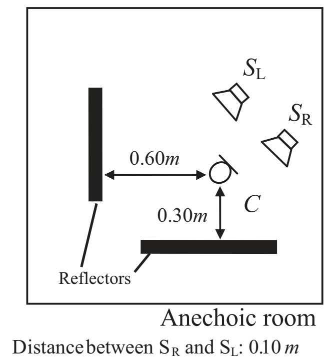


Fig. 5. Experimental environment

that is, the second-order Volterra kernels of two loudspeakers are identified accurately, the nonlinear distortion can be compensated completely.

4. EXPERIMENTAL RESULTS

To verify the applicability of the proposed method, some experiments were conducted. Table 1 shows experimental conditions. We identified linear and nonlinear characteristics of two loudspeakers in order to conduct compensation experiments. Figure 5 shows experimental environment. Figures 6∼8 show harmonic distortions and intermodulation distortions, respectively.

These figures show that the proposed compensation method can reduce more nonlinear distortions than the conventional one. This is because the proposed method uses the MINT and can consequently realize the exact linear inverse filtering. It can be also seen from Table 1 that the proposed method has low computational complexity and short delay compared with the conventional one. Hence, the proposed method also has an advantage on system realization.

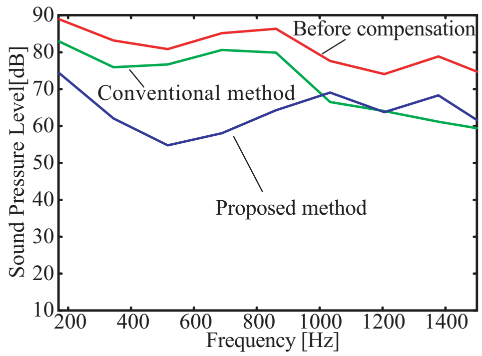


Fig. 6. Compensation results of harmonic distortions

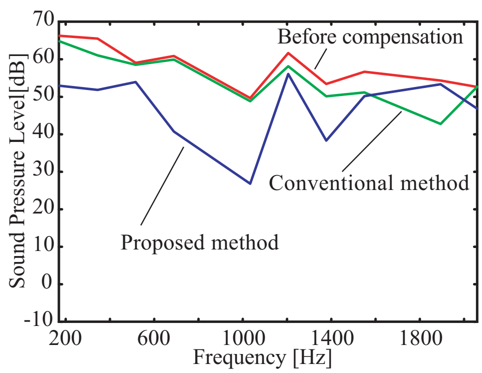


Fig. 7. Compensation results of intermodulation distortions (sum elements)

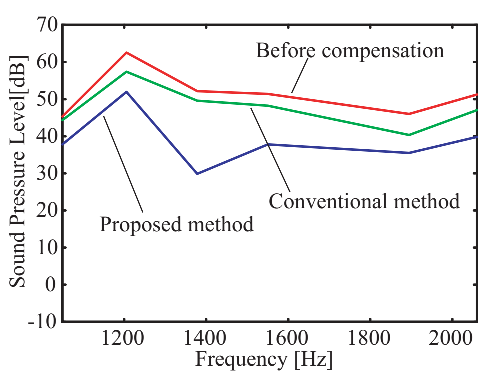


Fig. 8. Compensation results of intermodulation distortions (difference elements)

5. CONCLUSIONS

In this paper, we have proposed a novel linearization system using the MINT. Since exact inverse filtering can be realized by using the MINT, the proposed method has high compensation ability than the conventional one. Moreover, the whole computational complexity of the proposed method is half as large as that of the conventional one and the system delay is also small. Hence, the proposed method is efficient for system realization.

6. REFERENCES

[1] M. Schetzen, *Volterra and Wiener theories of nonlinear systems*, Krieger, Florida, 1989.

[2] V. J. Mathews, *Polynomial Signal Processing*, John Willy & Sons, New York, 2000.

[3] A. J. M. Kaizer, "Modeling of the nonlinear response of an electrodynamics loudspeaker by a volterra series expansion," *Journal of Audio Engineering Society.*, vol. 35, pp. 421–432, 1991.

[4] W. A. Frank, "An efficient approximation to the quadratic volterra filter and its application in real-time loudspeaker linearization," *Signal Processing*, vol. 45, no. 1, pp. 97–114, 1995.

[5] M. Tsujikawa, T. Shiozaki, Y. Kajikawa, and Y. Nomura, "Identification and elimination of second-order nonlinear distortion of loudspeaker systems using volterra filter," in *Proc. of the 2000 IEEE International Symposium on Circuits and Systems*, Geneva, Switzerland, 2000, pp. 249–252.

[6] M. Tsujikawa, Y. Kajikawa, and Y. Nomura, "dentification of third-order volterra kernels by multi-sinusoidal waves and its application to loudspeaker systems," in *Proc. of the International Workshop on Signal Processing Applications and Technology*, Tokyo, Japan, 2000, pp. 147–150.

[7] M. Miyoshi and Y. Kanada, "Inverse filtering of room acoustics," *IEEE Trans. on Acoustics, Speech, and Signal Processing*, vol. 36, no. 2, pp. 145–152, 1988.