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SUBBAND PARALLEL CASCADE VOLTERRA FILTER FOR LINEARIZATION OF LOUDSPEAKER SYSTEMS.

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ABSTRACT

In this paper, we propose a low complexity realization for linearization of loudspeaker systems. Linearization of loudspeaker system is generally realized by a Volterra filter. However, Volterra filter has a problem of requiring a huge computational complexity. The Simplified Volterra Filter (SVF), which removes the lines along the main diagonal of the 2nd-order Volterra kernel, has been previously proposed as a way to reduce the computational complexity while maintaining the compensation performance for the nonlinear distortion. However, this method cannot greatly reduce the computational complexity. Hence, we propose a subband linearization system which consists of a subband parallel cascade realization method for the 2nd-order Volterra kernel and subband linear inverse filter. Experimental results show that this proposed linearization system can produce the same compensation ability as the conventional method while reducing the computational complexity.

1. INTRODUCTION

The fundamental principle of loudspeaker systems has never changed since their invention. Loudspeaker systems have a very complex structure required to transform an electric signal into a mechanical vibration and to radiate acoustic waves. As a result, a loudspeaker system produces both linear and nonlinear distortions. These distortions consequently cause a deterioration in the sound quality. Recently, small-sized loudspeaker systems for desktop computers have come into wide use as personal computers (PC) advance. Since these sizes inevitably become smaller than general loudspeaker systems, the level of the nonlinear distortion generally has increased. Moreover, smaller loudspeakers such as those in cellular phones produce even more nonlinear distortion. Hence, the importance of compensating for the nonlinear distortion has been increasing in recent years.

Generally, identifying the linear and nonlinear elements of loudspeaker systems and designing a linearization system are essential in order to compensate for these distortions [1, 2]. However, there is the problem of having a huge computational complexity for the convolution required between the input signal and the 2nd-order Volterra kernel. A Simplified Volterra Filter (SVF), which removes the lines along the main diagonal of the 2nd-order Volterra kernel, has been previously proposed as a way to reduce the computational complexity while maintaining the compensation performance for the nonlinear distortion [3]. However, this method cannot greatly reduce the computational complexity. Hence, we propose a subband linearization system which consists of a subband parallel cascade realization method for the 2nd-order Volterra kernel and subband linear inverse filter [4]. This proposed linearization system can produce the same compensation ability as the conventional method while reducing the computational complexity.

2. LINEARIZATION SYSTEM

Figure 1 shows the structure of a linearization system to compensate for the 2nd-order distortion. In Fig. 1, $H_1(z)$ and $H_2(z_1,z_2)$ express the linear and 2nd-order nonlinear elements of the loudspeaker system. In the linearization system, $\hat{H}_2(z_1,z_2)$ is a model of the 2nd-order nonlinear element of the loudspeaker, and $H_1^{-1}(z)$ is designed so as to satisfy the following condition:

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

Hence, the 2nd-order nonlinear element of the whole system is expressed as follows:

$$\begin{aligned} & H_2(z_1,z_2)z^{-\Delta} - H_1(z)H_1^{-1}(z)\hat{H}_2(z_1,z_2) \\ = \; & H_2(z_1,z_2)z^{-\Delta} - z^{-\Delta}\hat{H}_2(z_1,z_2) \\ = \; & \left\{H_2(z_1,z_2) - \hat{H}_2(z_1,z_2)\right\} z^{-\Delta} \\ = \; & 0 \end{aligned} \tag{2}$$

If $\hat{H}_2(z_1,z_2)$ is equal to $H_2(z_1,z_2)$ of the loudspeaker and $H_1^{-1}(z)$ is designed so as to satisfy the condition of (1), then we can compensate for the 2nd-order nonlinear distortion.

However, there is the problem of having a huge computational complexity for the convolution required between the input signal and the 2nd-order Volterra kernel. Thus, it is clear that this linearization system cannot be implemented easily.

3. PARALLEL CASCADE REALIZATION

We consider the 2nd-order Volterra system where the Volterra kernel now has a finite memory length N. The input-output relation of this system is represented by

$$y_2(n) = \sum_{k_1=0}^{N-1}\sum_{k_2=0}^{N-1} h_2(k_1,k_2)x(n-k_1)x(n-k_2) \tag{3}$$

where $x(n)$ and $y(n)$ are sampled input and output signals at the n'th sample time, respectively, and the 2nd-order Volterra kernel $h_2(k_1,k_2)$ has the symmetric property as shown by

$$h_2(k_1,k_2) = h_2(k_2,k_1) \tag{4}$$

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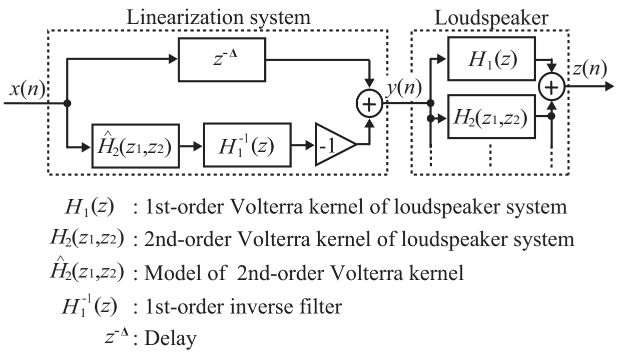


Figure 1: Structure of the conventional linearization system.

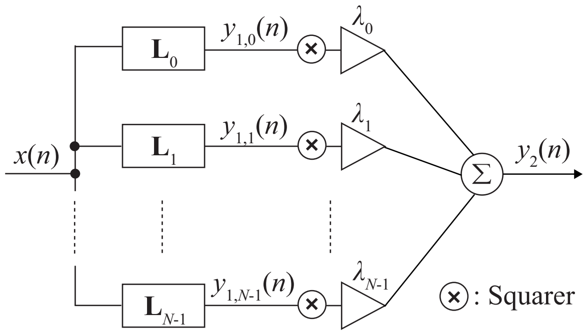


Figure 2: Structure of the 2nd-order parallel-cascade Volterra filter.

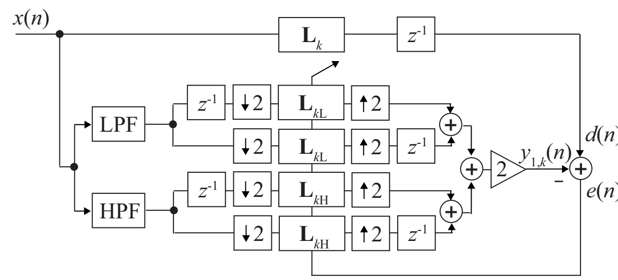


Figure 3: Structure of the subband adaptive filter.

The input-output relationship of the 2nd-order Volterra system in (3) can be given by using matrix-vector notation as indicated by

$$y_2(n) = \mathbf{X}(n)^T \mathbf{H}_2 \mathbf{X}(n) \tag{5}$$

where $\mathbf{X}(n)$ and $\mathbf{H}_2$ are an input vector and a $N \times N$ matrix of the 2nd-order Volterra kernel, respectively.

$$\mathbf{X}(n) = [\ x(n) \quad x(n-1) \quad \cdots \quad x(n-N+1)\ ]^T \tag{6}$$

$$\mathbf{H}_2 = \begin{bmatrix} h_2(0,0) & h_2(0,1) & \cdots & h_2(0,N-1) \\ h_2(1,0) & h_2(1,1) & \cdots & h_2(1,N-1) \\ \vdots & \vdots & \ddots & \vdots \\ h_2(N-1,0) & h_2(N-1,1) & \cdots & h_2(N-1,N-1) \end{bmatrix} \tag{7}$$

Now, the eigenvalue decomposition of a symmetric $N \times N$ matrix $\mathbf{H}_2$ is expressed as

$$\mathbf{H}_2 = \sum_{k=0}^{N-1} \lambda_k \mathbf{L}_k \mathbf{L}_k^T \tag{8}$$

where the terms λ_k are the eigenvalues of $\mathbf{H}_2$, and the terms $\mathbf{L}_k$ are the corresponding Nth-order eigenvectors and are expressed as

$$\mathbf{L}_k = [\ l_{0,k} \quad l_{1,k} \quad \cdots \quad l_{N-1,k}\ ]^T \tag{9}$$

Using this decomposition for the matrix $\mathbf{H}_2$ in (5) results in the following equation:

$$\begin{aligned} y_2(n) &= \sum_{k=0}^{N-1} \lambda_k [\mathbf{X}(n)^T \mathbf{L}_k][\mathbf{L}_k^T \mathbf{X}(n)] \\ &= \sum_{k=0}^{N-1} \lambda_k y_{1,k}^2(n) \end{aligned} \tag{10}$$

where

$$y_{1,k}(n) = \mathbf{X}(n)^T \mathbf{L}_k = \mathbf{L}_k^T \mathbf{X}(n) \tag{11}$$

The parallel cascade realization of the 2nd-order Volterra filter is defined as (10) and (11), and is shown in Fig. 2. It is understood from (10) that even if some parallel paths with relatively small eigenvalues are removed, the influence on the output signal $y_2(n)$ is slight. Hence, the computational complexity may be reduced while maintaining the accuracy of the output if small eigenvalues exist in the 2nd-order Volterra kernel. Moreover, if the linear FIR filter whose coefficients are the elements of eigenvector $\mathbf{L}_k$ is decomposed into a subband, the system realization can be distributed.

4. A SUBBAND PARALLEL CASCADE VOLTERRA FILTER

In this section, we explain the proposed subband parallel cascade Volterra filter. It is not easy to implement the conventional linearization system because of the huge serial arithmetic calculations. Therefore, the parallel cascade Volterra filter and linear inverse filter are made into a subband structure. Consequently, we can distribute the operation by parallel processing.

Figure 3 shows the configuration of a subband adaptive filter. Here, the LPF and HPF are determined from a Hadamard matrix. In case of dual partitioning, the coefficient values of the LPF are set to [0.5, 0.5], and those of the HPF are set to [0.5, -0.5]. Next, the subband parallel cascade structure, including the eigenvector and the subband linear inverse filter, are shown in Fig. 4 and Fig. 5, respectively. From these figures, we can understand that the operation can be distributed by parallel processing.

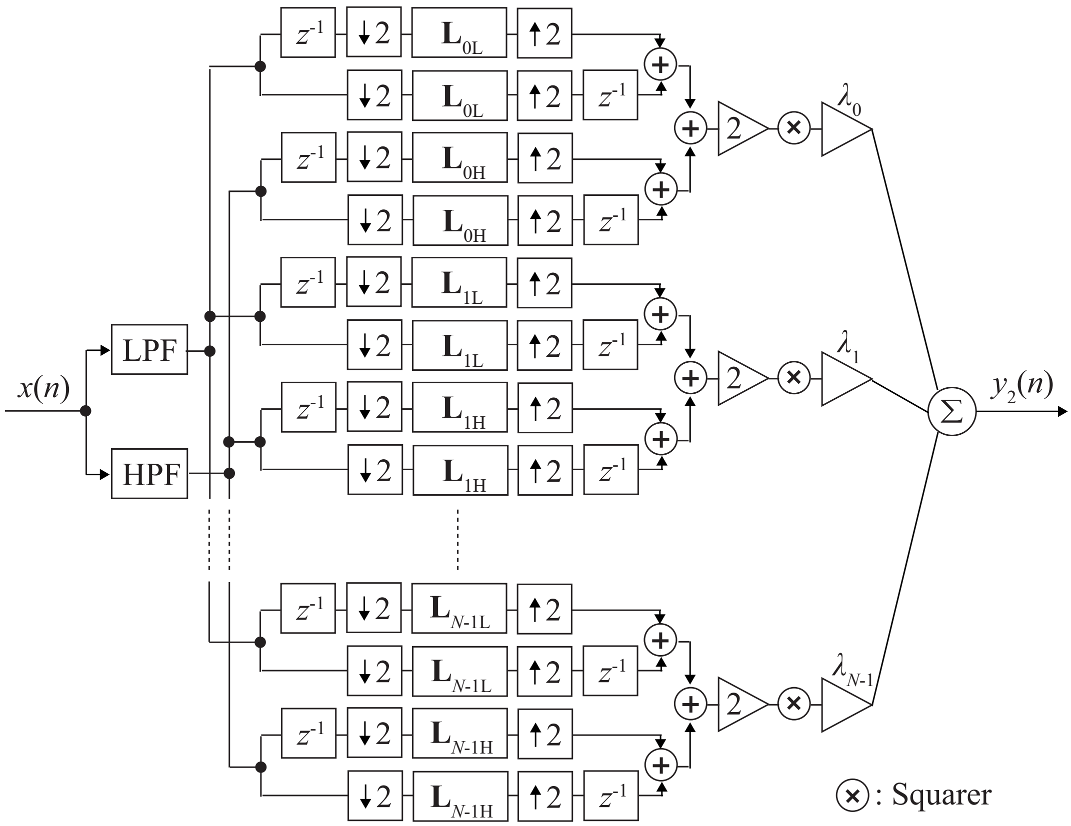


Figure 4: Structure of the 2nd-order subband parallel-cascade Volterra filter.

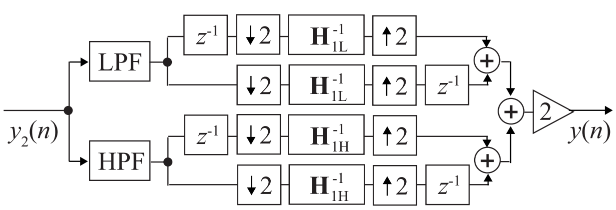


Figure 5: Structure of the subband linear inverse filter.

Table 1: Identification conditions.

Sampling frequency	44100 Hz
Frequency range	600-10000 Hz
Tap length of 1st-order kernel	256
Tap length of 2nd-order kernel	256
Input voltage	0.8 V

5. EXPERIMENTS REGARDING COMPENSATING FOR NONLINEAR DISTORTION

In this section, we conducted a nonlinear distortion compensation experiment for a loudspeaker system based on the conventional method and a proposed method which used the subband parallel cascade Volterra filter. In the experiment, we omitted the compensation for the linear distortion in order to examine the ability to compensate only for nonlinear distortion. The loudspeaker system was a cellular phone (kyocera), and the experiment was conducted within an anechoic box ($3.36m^3$).

5.1 Measurement of Volterra Kernel

We measured the 1st- and 2nd-order Volterra kernel of the loudspeaker system using the frequency response method[5]. Table 1 shows the identification condition. Figure 6 shows the eigenvalues of the 2nd-order Volterra kernel. The eigenvalues are arranged in a decreasing order of magnitude. We expect from Fig. 6 that a considerable number of eigenvalues could be reduced.

Figure 7 shows an approximated accuracy for the 2nd-order Volterra kernel versus the usage rate of eigenvalues (parallel paths). The approximated accuracy is defined as

$$\mathrm{AA} = 10\log_{10} \frac{\{\sum_{k=0}^{N-1}|\lambda_k| - \sum_{k=N-p}^{N-1}|\lambda_k|\}^2}{\{\sum_{k=0}^{N-1}|\lambda_k|\}^2} \tag{12}$$

where p is the number of reduced eigenvalues. In (12), 100% usage rate means that the approximated accuracy is 0dB(no approximation error). The approximated accuracy strongly relates to the performance compensation for the nonlinear distortion, and consequently the deterioration in the approximated accuracy is almost equal to the deterioration in the compensation in performance. It can be seen from Fig. 7 that

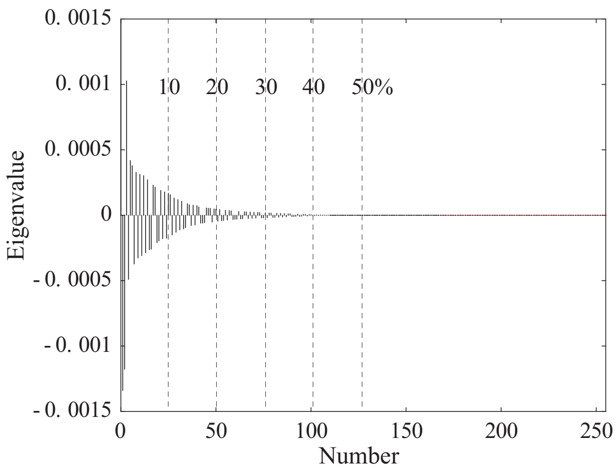


Figure 6: Eigenvalues of the 2nd-order kernel.

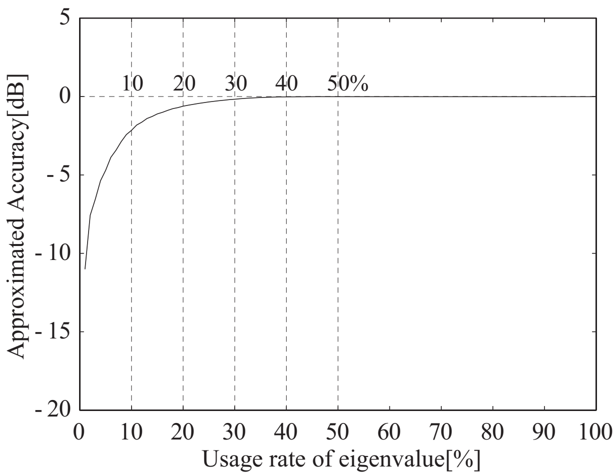


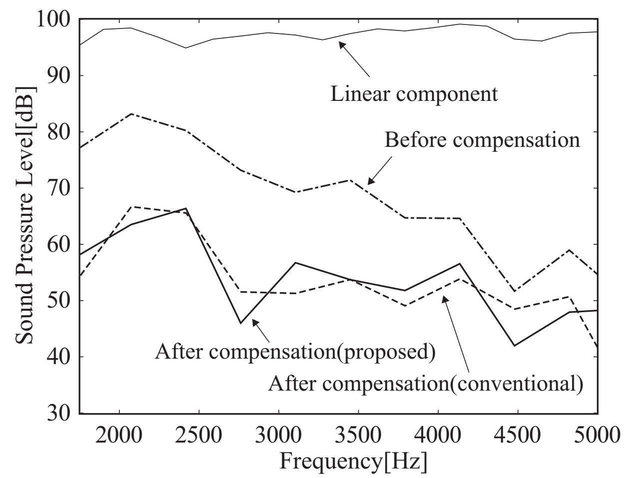
Figure 7: Approximated Accuracy.

significant errors arise only if more than 80% of the parallel paths are discarded.

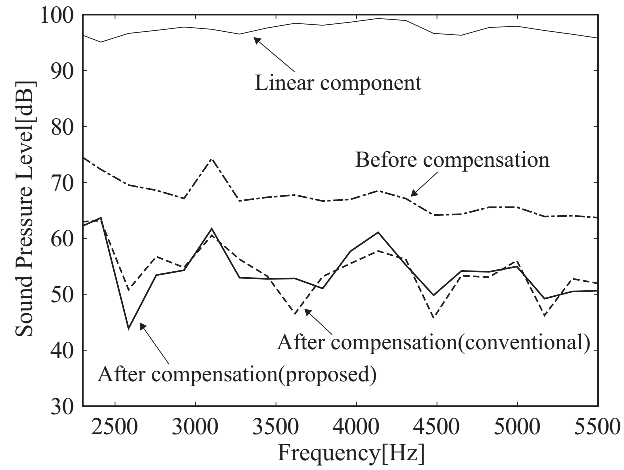
5.2 Experiment Results

We compared the conventional method and the proposed one. Here, the conventional method means the linearization system using the 2nd-order Volterra kernel and linear inverse filter, and the proposed method means the linearization system using the subband parallel cascade realization at the 20% usage rate of the eigenvalue.

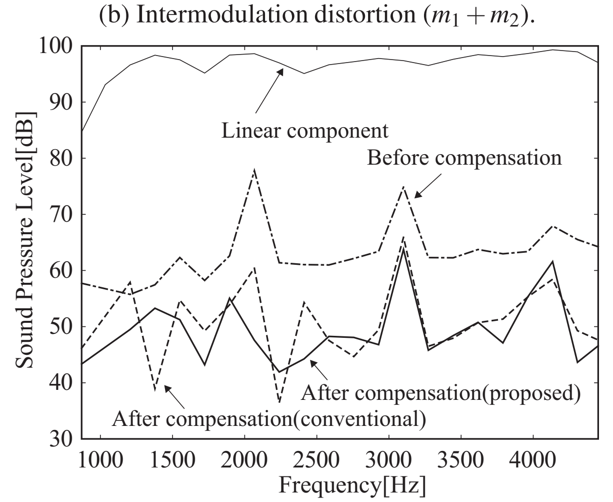
First, we generated the mixed sine waves with frequencies of m_1 and m_2 and produced a convolution of the linearization system and the corresponding sine waves. Next, we reproduced each signal from the loudspeaker and recorded the output signal. We then calculated the sound pressure levels of the 2nd-order nonlinear distortions for before and after the compensation. In the mixed sine waves, m_1 is fixed at 800Hz and m_2 is increased from 800Hz to 5500Hz. Figure 8 shows the sound pressure frequency responses of the loudspeaker system and the 2nd-order nonlinear distortions for before and after the compensation.



(a) Harmonic distrotion.



(b) Intermodulation distortion ($m_1 + m_2$).



(c) Intermodulation distortion ($m_1 - m_2$).

Figure 8: Sound pressure frequency responses of a loudspeaker system and the 2nd-order nonlinear distortions of $m_1 + m_2$ elements before and after compensation.

It can be seen from Fig. 8 that the level of the 2nd-order nonlinear distortions is reduced in the range of 10dB to 20dB over all frequency bands. Hence, the subband parallel cas-

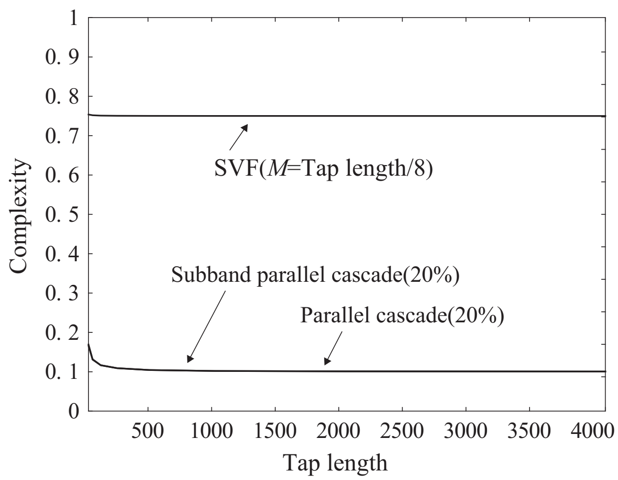


Figure 9: Computational complexity.

cade realization method can produce the same compensation ability as the conventional one while reducing the computational complexity.

6. COMPUTATIONAL COMPLEXITY

Finally, we compared the computational complexity for the convolution of each linearization system. The conventional method needs $2N^2+5N+1$ times of multiplication, the subband parallel cascade realization method $N^2-(p-6)N-2p+8$ times of multiplication, and the SVF method $2N^2+5N+1-2M(M+1)$ times of multiplication. The SVF method reduces the lines along the main diagonal of the 2nd-order Volterra kernel. M is the number of reduced lines. Figure 9 shows the computational complexity of each method. Here, the subband parallel cascade realization and the SVF methods are reduced to obtain the same compensation ability as the conventional method. In Fig. 9, the vertical axis shows the computational complexity normalized with respect to the conventional method.

It can be seen from Fig. 9 that the subband parallel cascade realization method has the lowest computational complexity of all methods. The computational complexity of the proposed method is about one-tenth that of the conventional one. Moreover, if the number of processors is more than N, the proposed method can distribute the operation by parallel processing.

7. CONCLUSIONS

In this paper, we have proposed a low computational complexity realization method of compensating for nonlinear distortion using a subband parallel cascade structure. The proposed method can produce the same compensation ability as the conventional one while reducing the computational complexity more effectively than the SVF method. Moreover, the operation can be distributed by parallel processing. In the future we will examine a method for reducing the computational complexity further.

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