LINEARIZATION ABILITY EVALUATION OF NONLINEAR FILTERS EMPLOYING DYNAMIC DISTORTION MEASUREMENT

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ABSTRACT
In this paper, the compensation ability of nonlinear distortions for loudspeaker systems is demonstrated using dynamic distortion measurement. Two linearization methods using a Volterra filter and a Mirror filter are compared. The conventional evaluation utilizes swept multi-sinusoidal waves. However, it is unsatisfactory because wideband signals such as those of music and voices are usually applied to loudspeaker systems. Hence, the authors use dynamic distortion measurement employing a white noise. Experimental results show that the two linearization methods can effectively reduce nonlinear distortions for wideband signals.

Index Terms— loudspeaker system, dynamic distortion method, Volterra filter, Mirror filter, linearization

1. INTRODUCTION
The fundamental principle of loudspeaker systems has not changed since their invention. Loudspeaker systems have a very complex structure that transforms an electric signal into a mechanical vibration and radiates acoustic waves. Loudspeaker systems consequently produce linear and nonlinear distortions which deteriorate the sound quality. It appears impossible to compensate these distortions completely by only structural improvements. Some researchers have therefore attempted to compensate the distortions using digital signal processing techniques.

So far, two linearization methods, using a Volterra filter[1, 2] and a Mirror filter[3, 4], have been proposed to compensate nonlinear distortions. The former is a method of approximating the nonlinear property of loudspeaker systems by a Volterra series expansion and the latter is a method of generating a compensation signal based on a nonlinear motion equation. A linearization method is usually assessed by the amount of reduction of nonlinear distortions for multi-sinusoidal waves when one frequency is fixed and another frequency is swept. However, wideband signals such as those of music and voices are generally applied to loudspeaker systems. Hence, the evaluation results for a pure tone with a particular frequency and level do not always correspond to those in actual driving situations. The authors therefore perform dynamic distortion measurement[5] using a white noise, which is a wideband signal.

Dynamic distortion measurement uses a wideband signal for evaluation. Narrowband components are removed from the wideband signal using a band-elimination filter and the resulting signal is applied to a loudspeaker system. Then, distortion components generated in the eliminated band are extracted using a band-pass filter whose bandwidth is the same as that of the band-elimination filter. In this letter, the compensation ability of the two linearization methods is examined by dynamic distortion measurement.

2. DYNAMIC DISTORTION MEASUREMENT USING WHITE NOISE
In this section, the principle of dynamic distortion measurement using a white noise is described. The procedure is shown in Fig. 1 and is as follows.

1. A white noise is generated as an evaluation signal.

2. From the generated white noise, narrowband components are removed using a band-elimination filter.

3. The band-eliminated white noise is applied to a loudspeaker system. Since loudspeaker systems generally have nonlinearity, nonlinear distortion components are produced over a wide band including the eliminated band.

4. The nonlinear distortion components in the eliminated band are extracted using a band-pass filter and their magnitudes are measured.

5. The above procedure is repeated while varying the center frequency of the band-elimination filter and the band-pass filter.
3. LINEARIZATION METHODS


Figure 2 shows the structure of the linearization method using the Volterra filter, which compensates the second- and third-order nonlinear distortions. In Fig. 2, $H_2(z_1, z_2)$ and $H_3(z_1, z_2, z_3)$ are models of the second-order and third-order nonlinear components of the loudspeaker system, respectively, and $H_1^{-1}(z)$ is designed so as to satisfy the condition

$$H_1(z)H_1^{-1}(z) = z^{-Δ}, \quad (1)$$

where $H_1(z)$ is the linear component of the loudspeaker system and $z^{-Δ}$ is the simple delay of $Δ$ samples. The input-output relationship of the structure shown in Fig 2 is represented by

$$u_L(n) = u(n - Δ)$$

$$+ \sum_{k=0}^{M-1} h_1^{-1}(k) y(n - k), \quad (2)$$

$$y(n) = \sum_{k_1=0}^{N-1} \sum_{k_2=0}^{N-1} \hat{h}_2(k_1, k_2) u(n - k_1) u(n - k_2)$$

$$+ \sum_{k_1=0}^{N-1} \sum_{k_2=0}^{N-1} \sum_{k_3=0}^{N-1} \hat{h}_3(k_1, k_2, k_3)$$

$$u(n - k_1) u(n - k_2) u(n - k_3), \quad (3)$$

where $h_1^{-1}(k)$, $\hat{h}_2(k_1, k_2)$ and $\hat{h}_3(k_1, k_2, k_3)$ are the time-domain representations of $H_1^{-1}(z)$, $H_2(z_1, z_2)$ and $H_3(z_1, z_2, z_3)$, which are called the first-order linear inverse filter and second- and third-order discrete Volterra kernels, respectively. $M$ and $N$ represent the memory lengths of each filter, respectively. The discrete signal $u_L(n)$ is conveyed to the loudspeaker system and compensates the nonlinear distortions (See reference [2] in detail).

3.2. Method Using Mirror Filter[3, 4]

The linearization method using a Mirror filter is based on the nonlinear motion equation of a loudspeaker system and utilizes the linear and nonlinear parameters of the loudspeaker system, which are determined and estimated from measured impedance curves and displacement characteristics. Figure 3 shows a block diagram of the linearization method using the second order IIR realization of the Mirror filter. In the figure, each parameter is defined as follows:

$$B_1 = \left(-2 + \frac{ω_0^2}{2f_s^2}\right) / α$$

$$B_2 = \left(1 - \frac{ω_0}{2Q_0f_s} + \frac{ω_0^2}{4f_s^2}\right) / α$$

$$h_{x0} = h_{x2} = \frac{h_{x1}}{2} = \frac{1}{4f_s^2} / α$$

$$C(x(n)) = 1 + \left[\frac{ω_0}{2Q_0f_s} \left(1 - \frac{Q_0}{Q_m}\right) (b(x(n))^2 - 1) + \frac{ω_0^2}{4f_s^2} (k(x(n)) - 1)\right] / α$$
4. EVALUATION OF COMPENSATION ABILITY

The compensation ability of the two linearization methods is examined using dynamic distortion measurement. A loudspeaker system whose specifications are shown in Table 1 was used in the following experiments. The microphone was located just 0.8 m from the loudspeaker system in an anechoic box (W1400, L1600, H1400).

\[
\begin{align*}
D(x(n)) &= B_1 + \frac{\omega_0^2}{2f_s^2} (k(x(n)) - 1) + \alpha \\
E(x(n)) &= B_2 + \left[ -\frac{\omega_0}{2Q_0f_s} \left( 1 - \frac{Q_0}{Q_m} \right) (b(x(n))^2 - 1) \right] + \frac{\omega_0^2}{4f_s^2} (k(x(n)) - 1) + \frac{\omega_0^2}{2f_s^2} (n(n)) - 1) + \alpha \\
G_0 &= \frac{B_0A_0}{R_e m_0},
\end{align*}
\]

where

\[
\begin{align*}
\alpha &= 1 + \frac{\omega_0}{2Q_0f_s} + \frac{\omega_0^2}{4f_s^2} \\
\omega_0 &= \sqrt{\frac{K_0}{m_0}} \\
Q_0 &= \frac{\sqrt{m_0K_0}}{R_m + B_0^2/R_e} \\
Q_m &= \frac{\sqrt{m_0K_0}}{R_m}
\end{align*}
\]

and \( B_0 \) is the force factor, \( R_e \) is the electrical resistance of the voice coil, \( m_0 \) is the mechanical mass, \( K_0 \) is the mechanical stiffness, \( R_m \) is the mechanical resistance and \( f_s \) is the sampling frequency. Also,

\[
\begin{align*}
Bl(x) &= Bl_0b(x) = Bl_0(1 + b_1x + b_2x^2) \quad (4) \\
K(x) &= K_0k(x) = K_0(1 + k_1x + k_2x^2), \quad (5)
\end{align*}
\]

where \( b(x) \) and \( k(x) \) express the nonlinearity of the force factor and the stiffness, respectively. The discrete signal \( u_L(n) \) is conveyed to the loudspeaker system and compensates the nonlinear distortions (See reference [4] in detail).

4.1. Identification of Volterra Kernels

The first-, second- and third-order Volterra kernels of the loudspeaker system were measured according to the method shown in reference [2]. Table 2 shows the identification conditions. The linearization system shown in Fig. 2 was constructed on the basis of the obtained Volterra kernels.

4.2. Estimation of Linear and Nonlinear Parameters for Mirror Filter

The linear parameters of the loudspeaker system were determined from the measured impedance curves based on the added mass method. Here, the added mass \( M \) was 288 mg to allow both resonance peaks to be distinguished. Next, the linear and nonlinear parameters were estimated using the method in reference [4]. In the estimation, the impedance curves and displacement characteristic of the diaphragm were used. These were measured using two sinusoidal waves whose frequencies were \( f_a = 220 \) Hz and \( f_b = 450 \) Hz, and the electric power was 4 W. Finally, the linearization system shown in Fig. 3 was constructed on the basis of the estimated linear and nonlinear parameters of the loudspeaker system.

4.3. Examination of Compensation Ability Using Dynamic Distortion Method

The compensation ability of the two linearization methods is examined based on dynamic distortion measurement. The experimental conditions are shown in Table 3. In the experiment, the average values of the distortion components produced in each observation band were measured using a band-pass filter whose center frequency was changed by increments...
Fig. 4. Comparison between the compensation ability of each linearization method for dynamic distortion.

Fig. 5. Comparison between the compensation ability of each linearization method for multi-sinusoidal waves.

of 1/3 octave from 250 to 2000 Hz.

The sound pressure levels of the dynamic distortion before and after compensation are shown in Fig. 4. Similarly, the sound pressure levels of nonlinear distortions for multi-sinusoidal waves are shown in Fig. 5. In the figure, the average values of nonlinear distortions for multi-sinusoidal waves are shown every 1/3 octave. We see by Fig. 4 that the level of dynamic distortions can be reduced by 15 to 20 dB by the linearization method using the third order Volterra filter, by 8 to 10 dB by linearization using the second-order Volterra filter, and by 3 to 8 dB by linearization using the Mirror filter. Hence, dynamic distortion measurement is effective for evaluating the compensation ability of linearization methods for loudspeaker systems. In the future, the authors will attempt to construct evaluation methods using music or voice signals.

5. CONCLUSIONS

In this letter, the compensation ability of two linearization methods employing the Volterra and Mirror filters was examined using dynamic distortion measurement. Experimental results demonstrated that the level of dynamic distortions can be reduced by 15 to 20 dB by the linearization method using the third order Volterra filter, by 8 to 10 dB by linearization using the second-order Volterra filter, and by 3 to 8 dB by linearization using the Mirror filter. Hence, dynamic distortion measurement is effective for evaluating the compensation ability of linearization methods for loudspeaker systems.

6. REFERENCES