ACTIVE REDUCTION OF NONLINEAR LOUDSPEAKER DISTORTION

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INTRODUCTION

Electrodynamic loudspeakers have inherent nonlinearities generating harmonic and intermodulation distortion in the reproduced sound. Some applications of active noise control require loudspeakers having low weight, small size, high efficiency to produce amplitudes at low frequencies. In these cases the signal distortions produced by the loudspeaker's nonlinearities impair the compensation of the primary sound. However, recent activities in loudspeaker research have developed physical models for the nonlinear mechanisms. They are the basis for digital controllers which compensate actively for loudspeaker distortion by preprocessing the electric input signal inversely. This paper gives a summary of this work and shows possible applications to active noise control.

LOUDSPEAKER MODELING

Lumped Parameter Model. The loudspeaker is modeled by a electromechanical equivalent circuit [1-8] presented in Fig. 1 where *u(t)* is the driving voltage. The dominant nonlinearities are represented by lumped parameters such as

- *b(x)* force factor of the electrodynamic motor,
- *k(x)* stiffness of the mechanical suspension and
- *L(x)* inductance of the voice coil

depending on the instantaneous excursion *x* of the voice-coil. Since the excursion is a low pass filtered signal, parameter variations are relatively fast generating additional distortion components in the audible band. Variations of the inductance *L(x)* also generate an electromagnetic reluctance force $F_m=L_x(x)i^2/2$ where $L_x(x)$ is the first-order derivative of *L(x)* and *i(t)* is the voice coil current. Another time-variant parameter is the voice coil resistance $R_e(T)$ which depends on the instantaneous voice coil temperature *T*. Due to the relatively high thermal capacity and resistance of the voice coil and magnet structure the temperature varies slowly and the variation of the electric resistance has only an influence on the linear transfer behavior (thermal power compression [9 – 10]) but does not produce new spectral components. The remaining lumped elements in the equivalent circuit in Fig. 1 are assumed as constants and correspond with the parameters used in linear modeling [11-13]:

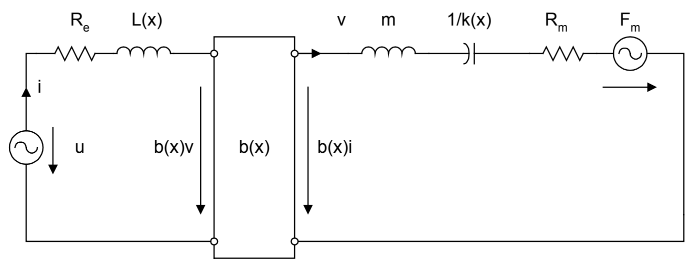

Fig. 1: Nonlinear electro-mechanical equivalent circuit

m mechanical mass of driver diaphragm assembly including voice-coil and air load,
R_m mechanical resistance of total-driver losses and real part of radiation impedance.

Nonlinear Differential Equation. The relationship in the equivalent circuit yields the set of nonlinear equations

$$u = R_e i + \frac{d\left(L(x)i\right)}{dt} + b(x)\frac{dx}{dt} \tag{1}$$
$$b(x)i = m\frac{d^2x}{dt^2} + R_m\frac{dx}{dt} + k(x)x + L_x(x)\frac{i^2}{2}.$$

written in the general state space form

$$\begin{aligned} \dot{\mathbf{x}} &= \mathbf{a}(\mathbf{x}) + \mathbf{b}(\mathbf{x})u \\ y &= h(\mathbf{x}) \end{aligned} \tag{2}$$

where $\mathbf{x}(t)= [x_1, x_2, x_3]^T = [x, dx/dt, i]^T$ is the state vector of the system comprising displacement x, velocity dx/dt of the voice-coil, the electrical input current i , y is the output corresponding to the voice-coil displacement x with

$$h(\mathbf{x}) = x_1 \tag{3}$$

and the components of $\boldsymbol{a}(\mathbf{x})$ and $\boldsymbol{b}(\mathbf{x})$ which are smooth nonlinear functions of the state vector $\mathbf{x}$ as defined by

$$\mathbf{a}(\mathbf{x}) = \begin{bmatrix} 0 & 1 & 0 \\ -\dfrac{k(x_1)}{m} & -\dfrac{R_m}{m} & \dfrac{b(x_1)}{m} + \dfrac{L_x(x_1)x_3}{2m} \\ 0 & -\dfrac{b(x_1)+L_x(x_1)x_3}{L(x_1)} & -\dfrac{R_e}{L(x_1)} \end{bmatrix} \tag{4}$$

$$\mathbf{b}(\mathbf{x}) = \begin{bmatrix} 0 & 0 & \dfrac{1}{L(x_1)} \end{bmatrix}^T . \tag{5}$$

State Space Model in Normal Form. Using a smooth, locally defined coordinate transformation T: x $\rightarrow$ z defined by

$$\begin{aligned} z_1 &= T_1(\mathbf{x}) = x_1 \\ z_2 &= T_2(\mathbf{x}) = x_2 \\ z_3 &= T_3(\mathbf{x}) = -\frac{k(x_1)x_1}{m} - \frac{R_m x_2}{m} + \left(\frac{b(x_1)}{m} + \frac{L_x(x_1)x_3}{2m}\right)x_3 \end{aligned} \tag{6}$$

which has also a smooth inverse $\boldsymbol{T}^{-1}$: $\mathbf{z} \rightarrow \mathbf{x}$ the general state space in Eq. (2) can be written in the normal state space form

$$\begin{aligned}
\dot{z}_1 &= z_2 \\
\dot{z}_2 &= z_3 \\
\dot{z}_3 &= f\left(\mathbf{T}^{-1}(\mathbf{z})\right) + g\left(\mathbf{T}^{-1}(\mathbf{z})\right)u = f(\mathbf{x}) + g(\mathbf{x})u \\
&\text{-----------------------------------} \\
y &= z_1
\end{aligned} \tag{7}$$

with

$$\begin{aligned}
f(\mathbf{x}) = &-\frac{k_x(x_1)x_1}{m} - \frac{k(x_1)x_2}{m} + \left[\frac{b_x(x_1)x_2}{m} + \frac{L_{xx}(x_1)x_2x_3}{2m}\right]x_3 \\
&-\frac{R_m}{m}\left[-\frac{k(x_1)x_1}{m} - \frac{R_m x_2}{m} + \left(\frac{b(x_1)}{m} + \frac{L_x(x_1)x_3}{2m}\right)x_3\right] \\
&-\left[\frac{b(x_1) + L_x(x_1)x_3}{mL(x_1)}\right]\left[\left(b(x_1) + L_x(x_1)x_3\right)x_2 + R_e x_3\right]
\end{aligned} \tag{8}$$

and

$$g(\mathbf{x}) = \frac{b(x_1) + L_x(x_1)x_3}{mL(x_1)}. \tag{9}$$

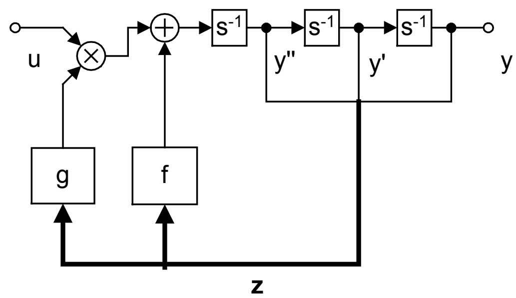

Fig. 2: Model of the Loudspeaker System

This representation shows the input/output dynamic as a direct relationship between the electric input voltage u and the third derivative of the displacement as discussed in detail in [14-16]. This model corresponds with the signal flow chart presented in Fig. 2. A chain of single order integrators generates the state vector **z** which is fed back via the static nonlinear functions $g(\mathbf{T}^{-1}(\mathbf{z}))$ and $f(\mathbf{T}^{-1}(\mathbf{z}))$ to the input where it is multiplied with and added to the voltage u, respectively.

NONLINEAR CONTROL STRUCTURE

Desired Dynamics. With a view of finding the nonlinear control law the dynamics of the overall system (controller + loudspeaker) are defined in normal state space form as

$$\begin{aligned}
\dot{z}_1 &= z_2 \\
\dot{z}_2 &= z_3 \\
\dot{z}_3 &= f_D\left(\mathbf{T}^{-1}(\mathbf{z})\right) + g_D\left(\mathbf{T}^{-1}(\mathbf{z})\right)w = f_D(\mathbf{x}) + g_D(\mathbf{x})w \\
&\text{-----------------------------------} \\
y &= z_1
\end{aligned} \tag{10}$$

where

$$f_D\left(\mathbf{T}^{-1}(\mathbf{z})\right) = -\frac{R_e k(0)}{mL(0)} z_1 - \left(\frac{k(0)}{m} + \frac{b(0)^2 + R_e R_m}{mL(0)}\right) z_2 - \left(\frac{R_e}{L(0)} + \frac{R_m}{m}\right) z_3 \tag{11}$$

$$g_D\left(\mathbf{T}^{-1}(\mathbf{z})\right) = \frac{b(0)}{mL(0)} \tag{12}$$

and w is the input signal of the controller.

Nonlinear Control Law. Considering the state space representation of the loudspeaker in normal form described by Eq. (7) the control law

$$u = \frac{g_D(T^{-1}(\mathbf{z}))}{g(T^{-1}(\mathbf{z}))}\left(w + \frac{f_D(T^{-1}(\mathbf{z})) - f(T^{-1}(\mathbf{z}))}{g_D(T^{-1}(\mathbf{z}))}\right) \tag{13}$$

yields the linearized dynamics in the overall system described by Eq. (10). Expressing the control law as a function of state vector $\boldsymbol{x}$ the control law in the direct form becomes

$$u = \alpha(\mathbf{x})\left[w + \beta(\mathbf{x})\right] \tag{14}$$

where the control gain is

$$\alpha(\mathbf{x}) = \frac{L(x_1)b(0)}{L(0)\left(b(x_1) + L_x(x_1)x_3\right)} \tag{15}$$

and the control additive is

$$\begin{aligned}\beta(\mathbf{x}) = {} & \frac{R_e}{b(0)}\left(k(x_1) - k(0)x_1\right)x_1 - \frac{L(0)}{b(0)}\left(b_x(x_1) + \frac{L_{xx}(x_1)}{2}x_3\right)x_2 x_3 \\ & + \frac{L(0)}{b(0)}\left(k_x(x_1)x_1 + k(x_1) - k(0) + \frac{\left(b(x_1) + L_x(x_1)x_3\right)^2}{L(x_1)} - \frac{b(0)^2}{L(0)}\right)x_2 \\ & + \frac{R_e}{b(0)}\left(\frac{L(0)\left(b(x_1) + L_x(x_1)x_3\right)}{L(x_1)} - b(x_1) - \frac{L_x(x_1)x_3}{2}\right)x_3 .\end{aligned} \tag{16}$$

The control law represented by Eq. (14) is static in the state variables of vector x. Alternatively, the control law can also be written in a dynamic form (mirror filter [17]) by using differentiators resulting in a structure with reduced complexity where each term has a physical meaning showing the compensation for nonlinear stiffness, nonlinear damping, parametric excitation, electromagnetic reluctance force and nonlinear self-inductance. Both control laws are based on the same physical model and provide exact linearization of the modeled nonlinearities in theory.

The control law causes minimal changes in the transferred signal because the control gain becomes constant $\alpha(\boldsymbol{x})=1$ and the control additive vanishes $\beta(\boldsymbol{x})=0$ if the nonlinear loudspeaker parameters equal the linear parameters of the desired system.

It is also an interesting feature that the control law described by Eqs. (14)-(16) is not restricted to a driver without enclosure as modeled here but gives exact input-output linearization for loudspeaker systems with radiation aids (closed box, vented-box, enclosure with bandpass characteristic, folded horns) as long as the strong relative degree remains constant ($\gamma=3$) and the elements of the radiation aid are linear.

State Measurement and Feedback. The control law requires permanently information on the instantaneous loudspeaker state (displacement, velocity, current). Direct measurement of the state signals and feeding back to the control law is a most common technique known as *Static State Feedback Linearization* and illustrated by Fig. 3. However, the practical application to loudspeaker faces some major problems:

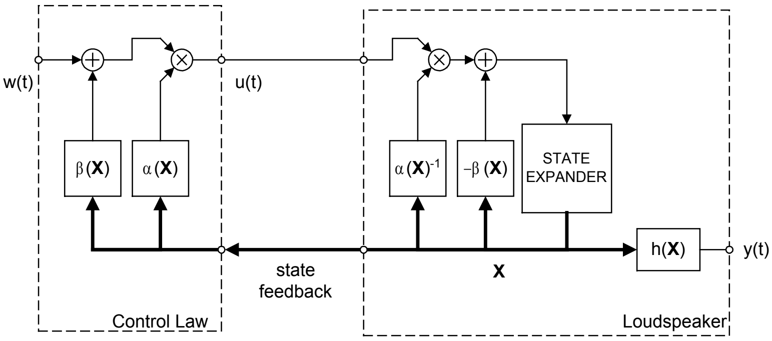

Fig. 3: Nonlinear Control with Static State Feedback

Whereas the input current at the loudspeaker terminals may be measured by a simple shunt at low cost, the measurement of the displacement requires a expensive sensor such as a laser displacement meter. The displacement signal can not be calculated by integrating the velocity or acceleration measured by less expensive sensors or the sound pressure available in some applications of active noise control because a precise measurement of the dc part of the displacement is required. The acoustic signal measured by a microphone also depends on the acoustical environment and is corrupted by measurement noise.

A second issue of *State Feedback* is associated with DA- and AD-conversion of the digital control signal *u(t)* and the analogue state vector **x**, respectively, where any time delay deteriorates the performance the controller.

State Estimation. To avoid the problems associated with permanent state measurement, the state information required in the control law can also be synthesized by a model system realized in the digital domain. There are two alternatives which are of practical interest. The model can be connected in parallel to the loudspeaker as shown in Fig. 4 and is a digital realization of the nonlinear differential equation given by Eq. (1). There is a need for precise knowledge of the linear and nonlinear loudspeaker parameters to guarantee stability within the state estimator which is itself a nonlinear feedback system. However, state estimation from the pre-distorted control signal *u* seems complicated in view of the fact that the linearized overall system (Controller + Loudspeaker) already provides a linear relationship between the states x_1, x_2 and the control input *w*.

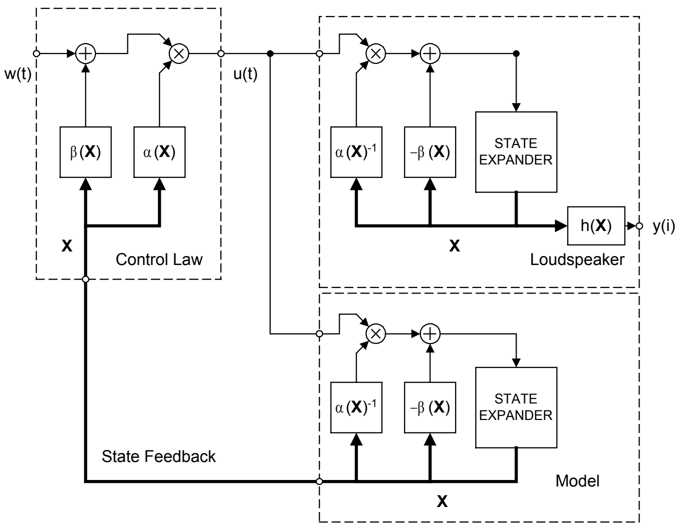

Fig. 4: Nonlinear Control with State Observer

Therefore, synthesizing the states from the control input u as shown in Fig. 5 is an interesting alternative to parallel modeling and leads to the mirror filter technique [17]. The state synthesis in the mirror filter yields an almost complete feed-forward structure which is the counterpart of the feedback system representing the loudspeaker. This is illustrated in Fig. 5 where the undesired distortion source is separated from the desired part within the nonlinear loudspeaker. The distortion source is just the inverse of the control law in the direct form. Clearly, both systems cancel each other out and the state estimation in the mirror filter becomes identical with the state generation in the real loudspeaker.

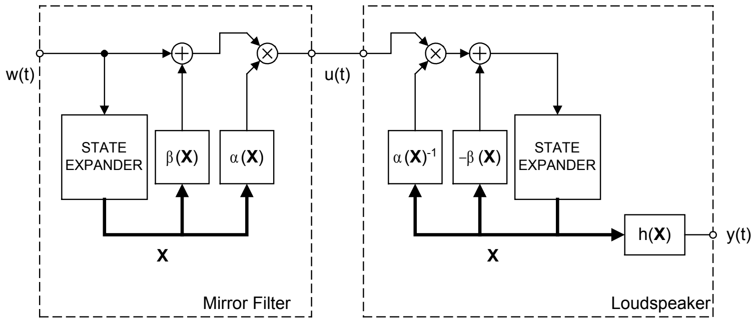

Fig. 5: Nonlinear Control with Mirror Filter

Approximative and Generic Control Structures. Whereas the control law represented by Eq. (14) allows perfect compensation of the nonlinearities considered in the loudspeaker model there are other approaches providing an approximative solution.

Replacing the nonlinear parameters in the mirror filter by a power series expansion and assigning the multitude of nonlinear term to homogeneous subsystems leads to a special polynomial filter dedicated to loudspeakers. Such a filter can also be derived by modeling the loudspeaker by the Volterra series and calculating the inverse system function [7]. Unfortunately, the computational complexity grows rapidly with the order of the system and higher-order subsystem can not be implemented in available DSP-systems at low cost.

Control architectures developed by other activities in nonlinear signal processing (robotics, image processing, and signal conditioning) can also be applied to loudspeakers [18 - 20]. These approaches use a more generic control structure (for example neural networks) and ignore the a priori knowledge available from physical modeling resulting in lower computational efficiency than controllers dedicated to loudspeakers. The number of free controller parameters is usually increased and a physical interpretation of the parameters is not possible.

ADAPTIVE PARAMETER ADJUSTMENT

The free parameters of the control structure must carefully be adjusted to the particular loudspeaker to cancel the distortion in the output signal successfully. In order to cope with parameter uncertainties and to ensure optimal performance of the aging loudspeaker, the controller should perform an automatic parameter adjustment on-line while reproducing an audio- or an ANC-signal. A system with such self-learning features is commonly called adaptive.

There are two ways for adjusting the controller to the loudspeaker [21]. *Direct updating* of the parameters in the controller has to cope with the nonlinearities in the loudspeaker which is part of the update process. The *indirect updating* requires an additional adaptive system to model the plant and to transfer identified parameters into the controller afterwards.

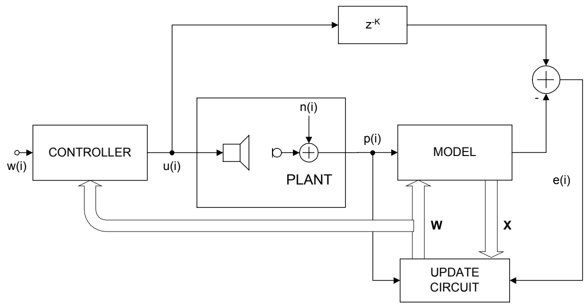

Fig. 6: Adaptive control based on inverse modeling

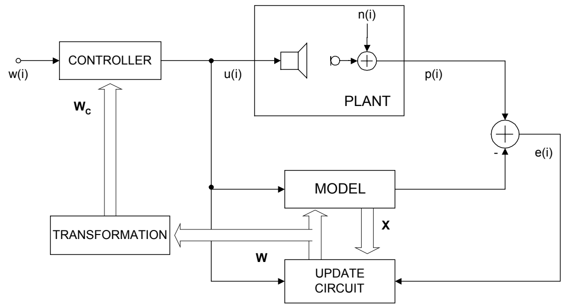

Fig. 7: Adaptive Control based on parallel modeling

Indirect Updating. The identification of the plant can be accomplished by an adaptive model connected in a serial or parallel arrangement to the loudspeaker. Fig. 6 shows the serial way where the adaptive model compensates for the nonlinearities of the preceding plant by estimating the inverse transfer function. An error signal *e(i)* is generated by comparing the model output with the delayed plant input *u(i-K)*. The identified parameters **W** can be copied into the controller where they are directly used as control parameters. Inverse modeling has a major disadvantage. When the measured signal p(i) is corrupted by noise n(i) as shown in Fig. 6 the estimate of the parameter vector W is biased and leads to a suboptimal performance of the controller [21].

Parallel modeling as shown in Fig. 7 is an interesting alternative to inverse modeling because the parameter estimation is immune against additional noise at the plant output. Here the error signal is defined as the difference between the plant output and the model output. After convergence of the parallel model the estimated model parameters **W** are transformed into control parameters $\mathbf{W}_C$ and supplied to the controller. The system used for parallel modeling is derived from the results of physical loudspeaker modeling. According to the nonlinear differential equation a feedback structure is required to describe the loudspeaker behavior at high amplitudes precisely. However, the stability of the model system can not ensured in the case of parameter uncertainties.

Direct Updating. Parallel and inverse plant modeling requires an additional nonlinear system which has almost the same complexity as the controller itself. Direct updating of the controller as shown in Fig. 8 is an interesting alternative because it dispenses not only with additional system identification and cumbersome parameter transformation but evaluates the final performance of the controller - the reduction of distortion in the reproduced sound. The adaptive circuit searches for minimal distortion in the loudspeaker output by correlating the gradient signals from the controller with the error signal. However, the update algorithm has to “look” through the nonlinear loudspeaker and the calculation of the precise gradient can not be performed by a DSP in real time. An intermittent update algorithm [21] solves this problem by using a simplified calculation of an approximative gradient and interrupts the learning when the estimation becomes invalid. This technique proves to be robust and convenient. It is not limited to the gradient-based algorithms (LMS) but can also be applied to the recursive least-square algorithm.

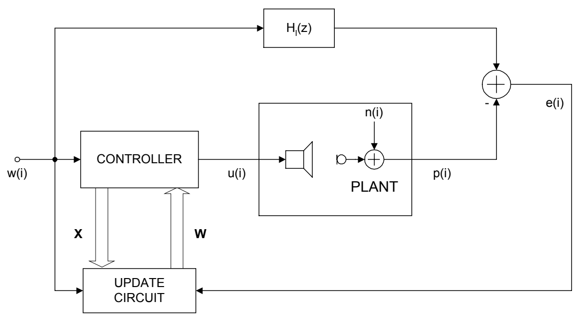

Fig. 8: Direct adaptive inverse control.

Nonlinear Detector System. The adaptive control schemes require an acoustic or mechanic output signal measured at the speaker. An additional sensor increases the costs and is impractical under harsh environment. However, electrodynamic loudspeakers can be used as sensor itself while reproducing a signal at the same time. The back electromotive force (EMF) produces a direct relationship between the electric input signals at the loudspeaker's terminals and the velocity of the voice-coil. A detector system as shown in Fig. 9 requires only the measurement of the current and voltage by using a simple shunt or a inexpensive current sensor. The velocity is estimated by an nonlinear algorithm derived from the voltage equation on the electric side or from the force equation on the mechanical side of the loudspeaker model. That results in highest accuracy possible while introducing a minimal number of unknown parameters which have to be identified. The detector is made adaptive to make the identification of the unknown parameters feasible while reproducing an audio signal. Adaptive detector algorithm have been developed which behaves stable and robust [22].

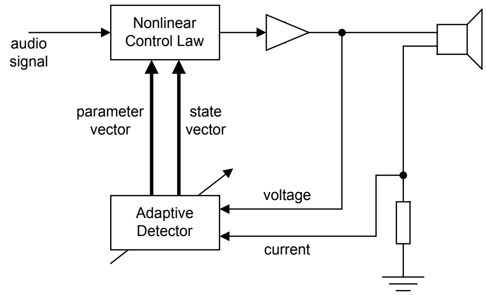

Fig. 9: Controller with EMF-detector.

Loudspeaker Protection. After convergence of the parameter adjustment, the adaptive controller knows the state signals and the linear and nonlinear parameters of the loudspeaker system. This information is valuable for realizing a protection against thermal and mechanical overload of the transducer. The temperature of the voice coil can easily estimated by monitoring the electric resistance $R_e(T)$. The maximal displacement is predicted by using the instantaneous displacement and velocity. If the predicted peak value exceeds an allowed threshold, the input signal will be attenuated in time to protect the mechanical system. The threshold describes the allowed working range of the particular loudspeaker and is automatically detected from the identified nonlinear parameters shown in Fig. 10 – 12.

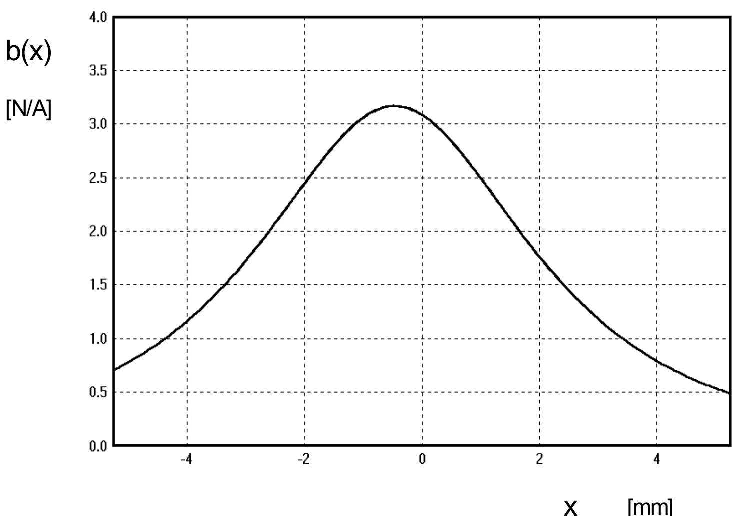

Fig. 10: Force factor as a function of voice-coil displacement

Variations of the force factor down to 20% of the value at the rest position indicates that the voice coil is almost out of the gap and the active compensation of the parametric excitation and nonlinear damping is not very efficient in respect with power consumption.

If the nonlinear stiffness exceed a critical value (usually an increase by factor 2 – 6), the moving capability of the mechanical suspension is exhausted and a further increase of the excursion could destroy the speaker.

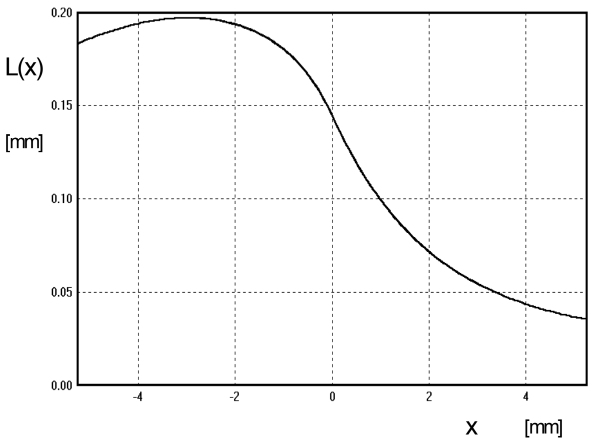

Fig. 11: Inductance versus displacement

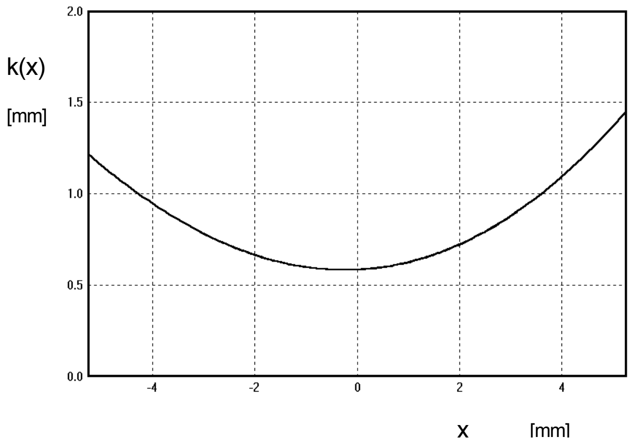

Fig. 12: Stiffness versus displacement

APPLICATION IN ACTIVE NOISE CONTROL

Digital Implementation. The multiplication of two time signals which is a key operation in nonlinear processing and adaptive parameter adjustment can easily performed by a digital signal processor (DSP) connected with AD- and DA-converters. The requirements on the DSP are relatively low. A fixed-point processor with 18 bits resolution of the operands and 20 MIPS processing power is sufficient to host the adaptive mirror filter with back-EMF detection. The required processing power can further reduced by performing a sequential updating of the parameters or by processing the control law with frozen parameters only and to activate the parameter updating when the DSP has free capacity.

The mirror filter does not generate any time delay in the processed signal which is an important feature for ANC-applications. On the other side the adaptive detector is tolerant of any time delay caused by an inexpensive sigma-delta ADC used for the conversion of the electric signals (voltage and current measured at the loudspeaker terminals) into the digital domain.

The nonlinear loudspeaker control can easily be integrated as a software module in applications of active noise and vibration control where a digital platform is already in use.

Operation. The self-learning capability of the adaptive control system simplifies the handling in practical application. Any electrodynamical loudspeaker driver mounted in a sealed or vented enclosure can be connected to the active control system. The controller activates a special learning routine to measure the parameters of the particular loudspeaker within a few minutes. A fast learning rate can be accomplished by using an internally generated broad-band noise signal which provides persistent excitation of the speaker. The controller behaves as an autonomous system requiring only general protection parameters such as maximal increase of voice-coil temperature as well as maximal variations allowed for force factor and stiffness parameters. After convergence of the control parameters to the optimal values the controller enters the normal operation mode where the external input signal is used and the parameter updating runs at lower speed in the background of the processor to compensate for aging and heating effects.

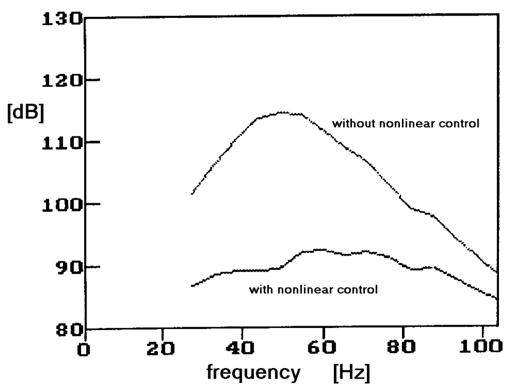

Fig. 14: Reduction of harmonic distortion

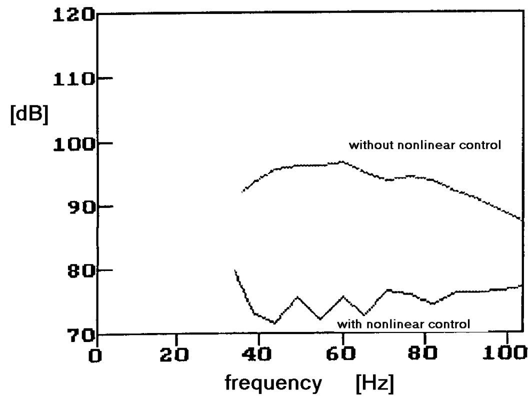

Fig. 15: Reduction of intermodulation distortion

Results. The performance of the controller can be evaluated on-line by monitoring the magnitude of the error signal generated in the update algorithm. Without nonlinear control the relative error exceeds 20 % and more at high amplitudes. After convergence of the parameter estimation the relative error decreases to values about 1-3 %. Exciting the loudspeaker with a two-tone signal and measuring the sound pressure signal, the harmonic and intermodulation components can easily separated from the fundamental components by spectral analysis. Fig. 14 and Fig. 15 show the reduction of the nonlinear distortions measured on a 5" driver mounted in a closed box system by activating and deactivating the nonlinear preprocessing.

Benefits for Active Noise Control. Loudspeaker nonlinearities may impair the performance of active noise control because the distortion components generated by the transducer are not canceled by the primary sound. Of course most of the loudspeakers behave sufficiently linear if the amplitude of the displacement is low. Operating a loudspeaker at small excursion demands for a larger area of the diaphragm to produce the required volume velocity at low frequencies. There are also attempts to improve the linearity of the passive driver. The progress in loudspeaker modeling, the new tools for measuring the large signal parameters and for identifying the constructional causes of the distortion (Distortion Analyzer [23]) allow to reduce asymmetric nonlinearities which are responsible for second- and higher-order distortion. However, the symmetric nonlinearities which are the dominant sources of distortion in well-made transducers are directly related with the ratio of gap depth and voice coil height and dimensions of the mechanical suspension. Improving the linearity increases the weight, size, and cost or reduces the maximal acoustic output and the efficiency of the transducer.

A nonlinear control system coupled with a special driver design gives new degrees of freedom. The passive driver can be optimized to give maximal output power at high efficiency, low weight, small size and with minimal cost. The controller compensates actively for the nonlinearities inherent in the driver and realizes the desired linear transfer characteristics (modification of the cut-off frequency and total loss factor) in the overall system. Parameter variations in the running-in period and caused by aging of fabric, foam and rubber materials or thermal effects are compensated permanently. Such an active loudspeaker system can be used as a module with well defined characteristics. The back-EMF detection dispenses with an additional sensor which is important for ANC-applications in a dusty, hot or in other way hostile environment where a high quality sensor can not mounted near to the loudspeaker. Due to the full thermal and mechanical protection the driver can be operated with reduced headroom just below the maximal load.

With the progress in sensor and DSP technology the passive loudspeaker has become the most critical component in active noise control. Nonlinear control techniques applied to loudspeaker system makes it possible to reduce the size, weight and cost of the ANC-systems which is important for new applications in cars and air crafts.

SUMMARY

Nonlinear loudspeaker models are the basis for predicting the nonlinear transfer behavior and for designing nonlinear control systems to compensate for loudspeaker distortion actively. Special control architectures dedicated to loudspeakers (mirror filter, state feedback control) are superior over generic structures (polynomial filter, neural network) in respect with performance, robustness and digital processing capacity required. The state information and the parameters have a physical meaning and allow to realize a reliable protection of the loudspeaker. Adaptive algorithms have been developed which detect the back-EMF from the electric signals at the loudspeaker terminals and use the loudspeaker itself as sensor system. The self-learning controller combined with a special driver design results in active loudspeaker systems reproducing sound with higher quality by using less material and energy.

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