# Adaptive Stabilization of Electro-dynamical Transducers

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## Introduction

Electro-dynamical transducers generating the required acoustical output at high efficiency, low cost, small size and minimum weight are strongly nonlinear systems causing not only harmonic and intermodulation distortion but also generating a DC-displacement which drives the coil away from the rest position.

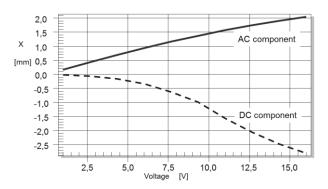


Fig. 1: Amplitude of the AC and DC components of the voice coil displacement for a sinusoidal stimulus above resonance

instability inherent in the electro-dynamical transduction principle is a major concern in the development of moving coil loudspeaker and can generate a DCcomponent which exceeds the RMS-value of the ACcomponent as shown in Fig. 1. The transducer engineer can cope with the instability by using a relatively stiff mechanical suspension (spider, surround) with progressive nonlinearity of stiffness characteristic  $K_{ms}(x)$  which increases the resonance frequency and reduces the maximal peak displacement X<sub>max</sub>.

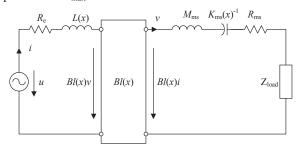


Fig. 2. Equivalent circuit of the electro-dynamical transducer

#### **Nonlinear Modeling of the Transducer**

The new control technique is based on a lumped parameter model depicted in Fig. 2 corresponding to the integrodifferential equations

$$u = R_e i + \frac{d(L(x)i)}{dt} + Bl(x)\frac{dx}{dt}$$
 (1)

 $Bl(x)i = (K_{ms}(x) - K_{ms}(0))x + L^{-1}\{sZ_m(s)\} * x$ 

with the force factor

$$Bl(x) = \sum_{i=0}^{N} b_i (x + x_{off}(t))^i,$$
 (2)

the stiffness of the mechanical suspension

$$K_{ms}(x) - K_{ms}(0) = \sum_{i=1}^{N} k_i \left( x + x_{off}(t) \right)^{i}$$
 and the voice coil inductance

$$L(x) = \sum_{i=0}^{N} l_i (x + x_{off}(t))^{i},$$
 (4)

which are nonlinear functions of the voice coil displacement x(t). The offset  $x_{\text{off}}(t)$  may be considered as a time-variant parameter depending on the interactions between transducer nonlinearities and stimulus, viscoelastic behavior of the suspension, gravity and other external influences. The offset  $x_{\text{off}}(t)$  may be also interpreted as a state variable which comprises only low frequency components far below the audio band.

Eq. (1) uses the convolution denoted by \* between displacement x and the inverse Laplace transform  $L^{-1}$  of the total mechanical impedance

$$Z_{m}(s) = \frac{\sum_{i=0}^{M} a_{i} s^{i}}{\sum_{i=0}^{M} c_{i} s^{i}}$$
 (5)

describing the effect of the mechanical stiffness  $K_{ms}(x=0)$  at the rest position, the mechanical resistance  $R_{ms}$ , the moving mass  $M_{ms}$  and the load impedance  $Z_{load}(s)$  of the coupled acoustical and mechanical system. The order M describes the number of poles and zeroes in the rational transfer function  $Z_m(s)$ . A transducer mounted in a sealed enclosure can be modeled by a second-order function  $Z_m(s)$  while a vented box system, panel or in a horn requires a higher-order system, which makes the identification of the linear parameters  $a_i$  and  $c_i$  more difficult.

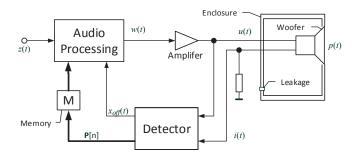


Fig. 3. Control System with stabilization of voice coil position

#### **Parameter Identification**

The free parameters of the lumped parameter model summarized in the parameter vector

$$\mathbf{P} = \begin{bmatrix} P_1 \dots & P_i & \dots & P_J \end{bmatrix}^T \tag{6}$$

and the instantaneous offset x<sub>off</sub>(t) can be identified from the voltage and current monitored at the transducer terminals while reproducing an audio signal (e.g. music). The system identification is based on minimizing an error signal such as the difference

$$e(t) = u'(t) - u(t) \tag{7}$$

between voltage u'(t) predicted by the model in (1) and the measured voltage u(t). The optimal parameter vector  $\mathbf{P}$  can be determined by searching for the minimum of the mean squared error

$$\mathbf{P} = \underset{\mathbf{P}, x_{off}}{\min} \left( E\{e(t)^2\} \right)$$
 (8)

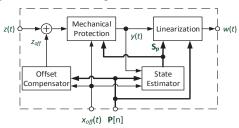
corresponding to the requirements

$$\frac{\partial E\left\{e(t)^{2}\right\}}{\partial P_{j}} = 2e(t)\frac{\partial u'(t)}{\partial P_{j}} \stackrel{!}{=} 0 \qquad j = 1,...,J$$
(9)

and

$$\frac{\partial E\left\{e(t)^{2}\right\}}{\partial x_{off}(t)} = 2e(t)\frac{\partial u'(t)}{\partial x_{off}(t)} \stackrel{!}{=} 0$$
 (10)

The detector has to identify the coil offset  $x_{\text{off}}(t)$  in the back-EMF at the transducer terminals by exploiting the nonlinear distortion generated in the audio band. The instantaneous value of  $x_{\text{off}}(t)$  is permanently supplied to the audio processing as illustrated in Fig. 3.



**Fig. 4.** Processing of the audio signal with mechanical protection, linearization and stabilization.

# Audio processing with stabilization

The state estimator as shown in Fig. 4 and the control law

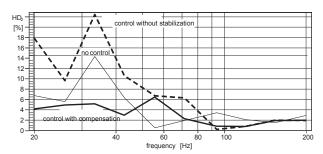
$$w(t) = \alpha(\mathbf{P}, \mathbf{S}_P, x_{off}) [y(t) + \beta(\mathbf{P}, \mathbf{S}_P, x_{off})]$$
(11)

consider the instantaneous voice coil offset  $x_{\rm off}(t)$ . The mechanical protection system compares the total displacement  $|x(t)+x_{\rm off}(t)|$  with the permissible limit value  $x_{\rm max}$  to detect a mechanical overload of the transducer and to attenuate the input signal in time. The instantaneous offset  $x_{\rm off}(t)$  may reduce the permissible peak or bottom value of the AC signal x(t) to avoid voice coil bottoming or irregular operation generating impulsive distortion commonly called "rub and buzz".

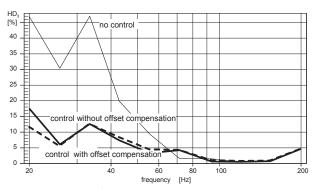
If a DC-coupled power amplifier is used in the application, an offset compensator as shown in Fig. 4 detects the symmetry point  $x_{\text{sym}}$  in the nonlinear force factor characteristic Bl(x) and synthesizes a DC signal  $z_{\text{off}}$  which moves the coil's rest position to the symmetry point  $x_{\text{sym}}$ , ensuring highest efficiency and minimum distortion.

# **Practical Evaluation**

The new control technique has been implemented in a microcontroller based on ARM 4 architecture and evaluation with artificial test signals, music and other audio signals. Fig. 5 and 6 show the reduction of harmonic distortion by using a nonlinear controller with and without compensation of the voice coil offset  $x_{\rm off}$  permanently measured by applying the sinusoidal stimulus.



**Fig. 5:** Relative 2<sup>nd</sup>-order harmonic distortion of the loudspeaker output without control (thin line), with nonlinear control without stabilization (dashed line) and nonlinear control with offset compensation (solid line).



**Fig. 6:** Relative 3<sup>rd</sup>-order harmonic distortion of the loudspeaker output without control (thin line), with nonlinear control without offset compensation (dashed line) and nonlinear control with offset compensation (solid line).

The compensation of the voice coil offset improves the active reduction of the 2<sup>nd</sup>-order harmonic distortion because any DC displacement or voice coil offset affects the asymmetry of the nonlinear curve shape. The 3<sup>rd</sup>-order harmonic distortion are less sensitive for a voice coil offset because they are more related with the symmetry of the nonlinear characteristics limited by geometrical constraints in the design of the motor and suspension.

# Conclusion

A new control technique for electro-dynamical transducer is presented which stabilizes the voice coil position, compensates for nonlinear distortion and generates a desired transfer response by preprocessing the electrical input signal. The control law is derived from transducer modeling using lumped elements and identifies all free parameters of the model by monitoring the electrical signals at the transducer terminals. The control system stays operative for any stimulus including music and other audio signals. The active stabilization is important for small loudspeakers generating the acoustical output at maximum efficiency.

## References

- [1] Klippel, Tutorial: "Loudspeaker Nonlinearities Causes, Parameters, Symptoms" *J. Audio Eng. Society* 54, No. 10 pp. 907-939 (Oct. 2006).
- [2] Klippel, "Adaptive stabilization of electro-dynamical transducers," Proceedings of the 22nd European Signal Processing Conference (EUSIPCO), Lisbon, Portugal, 2014, pp. 111-117.