Active Transducer Protection Part 1: Mechanical Overload

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The generation of sufficient acoustical output by smaller audio systems requires maximum exploitation of the usable working range. Digital preprocessing of audio input signals can be used to prevent a mechanical or thermal overload generating excessive distortion and eventually damaging the transducer. The first part of two related papers focuses on the mechanical protection defining useful technical terms and the theoretical framework to compare existing algorithms and to develop meaningful specifications required for the adjustment of the protection system to the particular transducer. The new concept is illustrated with a micro-speaker and the data exchange and communication between transducer manufacturer, software provider and system integrator are discussed.

1. INTRODUCTION

In many audio applications, the required acoustical output has to be reproduced at sufficient quality by using smaller, lighter and efficient transducers. The maximum voltage provided by available power amplifiers is usually large enough to move the voice coil to the maximum mechanical boundaries generating a mechanical overload and impulsive distortion, which have a high impact on the perceived sound quality. The heating of the coil is a second mechanism, which may damage the transducer. Micro-speakers as used in personal audio devices are a critical example because the suspension system provides no natural protection of the coil as found in normal woofers with a progressive spider. Amplifiers used in mobile phones can provide significantly more power than the micro-speaker can handle safely. This discrepancy causes the need for active protection, which is the most important feature of smart amplifiers.

The basic requirements and most important goals of protection systems are:

- Reliable protection of the transducer against mechanical and thermal overload for any audio input.
- Maximum of acoustical output
- Minimum of additional signal distortion and other artifacts degrading sound quality
- Minimum delay increasing the latency in the audio path
- Cost-efficient implementation

A variety of ideas and solutions for this important and challenging issue has been developed so far and is accessible as patent literature [1-16]. The paper presented by Vignon and Scarlett [17] is the first technical paper, which addresses the specification of transducer characteristics to simplify the communication between transducer manufacturer, software provider and system integrator. Test procedures and rating methods have been developed in this paper to specify the maximal voice coil excursion X_{max} required for tuning the protection systems in smart amplifiers. Two related papers presented here follow this approach and investigate this interesting topic in greater detail to improve the definition of the technical terms and to develop a common framework for evaluating different protection schemes.

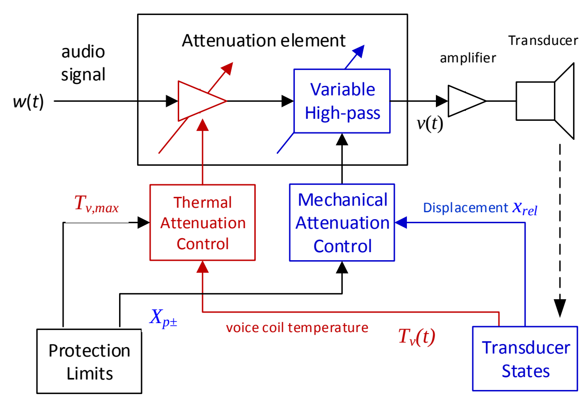

Figure 1 Input information required by an active protection system

The first paper presented here addresses the mechanical protection while the following paper is dedicated to the thermal protection. The mechanical overload can be prevented by activating a high-pass filter as depicted in Figure 1 and attenuating only low frequencies which contribute to the voice coil displacement $x_{rel}(t)$. The thermal protection requires the attenuation of low and high frequency components which increase the voice coil temperature T_v. Mechanical and thermal control systems activating the corresponding attenuation elements require the absolute voice coil position $x_{abs}(t)$ or relative voice coil displacement $x_{rel}(t)$ and temperature $T_v(t)$, which are the critical state variables indicating the two overload conditions. The information can be provided by direct measurement of the transducer state or by transducer modeling.

The attenuation control also shown in Figure 1 compares the measured or modeled state variables with protection limits, which describe the limits of the permissible working range.

2. DEFINITIONS

The instantaneous position of the voice coil

$$x_{abs}(t) = X_0(t, DUT) + x_{rel}(t) \quad (1)$$

depends on the voice coil rest position $X_0(t,\mathrm{DUT})$ and the voice coil displacement $x_{rel}(t)$ generated by the audio signal w(t). The voice coil rest position $X_0(t,\mathrm{DUT})$ is a (slowly) varying transducer parameter depending on the particular device under test (DUT) due to production spread. It also changes over time t due to aging, viscoelasticity, gravity, load and climate.

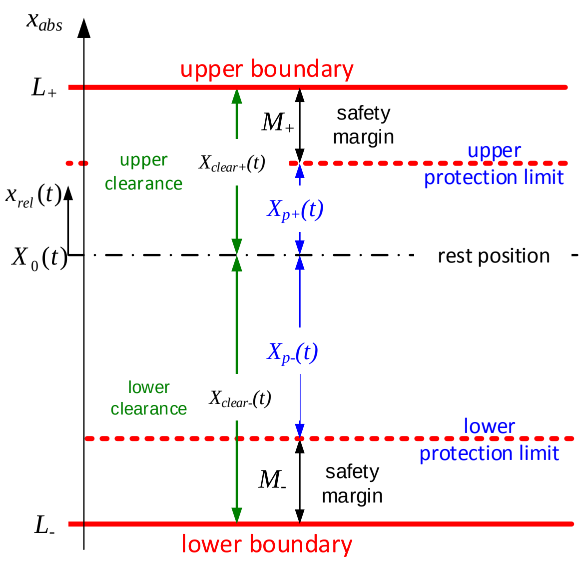

Figure 2 Limits and other characteristics describing the permissible displacement of the voice coil.

The mechanical protection shall ensure that the instantaneous voice coil position stays in the permissible working range:

$$L_- < x_{abs}(t) < L_+ \quad (2)$$

The boundaries L_+ and L_- depend on the particular design, but are considered as time invariant and independent of the particular device under test (DUT). The positive and negative clearances

$$\begin{aligned} X_{clear+}(t) &= L_+ - X_0(t) \\ X_{clear-}(t) &= L_- - X_0(t) \end{aligned} \quad (3)$$

represent the distance between the instantaneous rest position $X_0(t)$ and the upper and lower boundaries $L_\pm$, and give the theoretical limits of the voice coil displacement $x_{rel}(t)$. However, the attenuation control system uses the maximum positive and negative protection limits

$$\begin{gathered} -X_{p-}(t) \le x_{rel}(t) \le X_{p+}(t) \\ -X_{clear-}(t) + M_- \le x_{rel}(t) \le X_{clear+}(t) - M_+ \end{gathered} \quad (4)$$

which are the clearance values reduced by safety margins M_+ and M_- to cope with inaccurate state information and the reaction time of the attenuation control system.

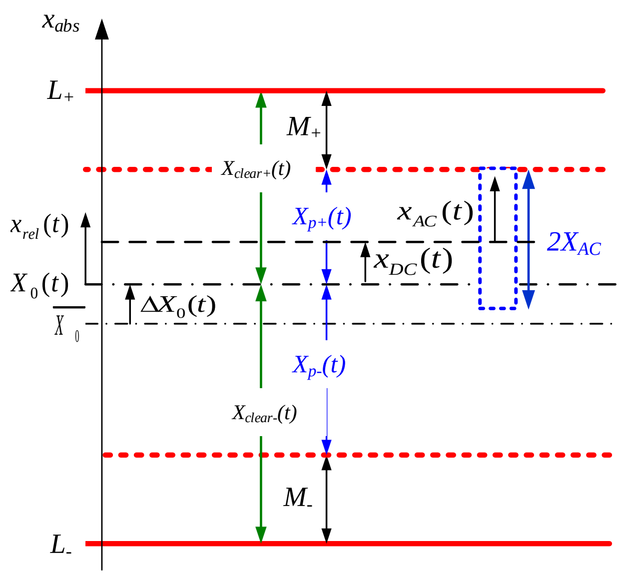

Figure 3 DC-component $x_{DC}(t)$ and AC-component $x_{AC}(t)$ of the voice coil displacement and the influence of an offset $\Delta X_0(t)$ in the voice coil rest position

While reliable protection under all circumstances is the primary goal, the protection system shall reduce the input signal only to prevent an overload situation. An overreaction of the protection system would reduce acoustical output, which is related to the AC displacement:

$$x_{AC}(t) = x_{rel}(t) - x_{DC}(t) \tag{5}$$

The DC component $x_{DC}(t)$ shifts the voice coil away from the instantaneous rest position $X_0(t)$ but generates no acoustical output. The DC component $x_{DC}(t)$ which may be in the same dimension as the AC components $x_{rel}(t)$ is generated by transducer nonlinearities rectifying the audio signal [21]. Contrary to the voice coil rest position $X_0(t)$, the DC component $x_{DC}(t)$ is a fast varying signal component depending on the instantaneous properties of the audio signal. This is a critical problem in transducers with a soft suspension, especially in micro-speakers, as investigated by Vignon et. al. [17].

A useful criterion for assessing the performance of the protection system is the nominal maximum amplitude $X_{AC}(t)$ which limits the desired AC-signal

$$-X_{AC} \le x_{AC}(t) \le X_{AC} \tag{6}$$

while assuming that the audio signal $w(t)$ has a symmetrical probability density function $pdf(w(t))$.

The end-of-line test completing the manufacturing process can provide valuable statistical information which describe the initial properties (t=0) of devices under tests (DUTs) such as the mean initial voice coil rest position

$$\overline{X_0} = \frac{1}{N_D} \sum_{n_{DUT}}^{N_D} X_0(t=0, n_{DUT}) \tag{7}$$

and the mean initial positive and negative clearances

$$\overline{X_{clear\pm}} = \frac{1}{N_D} \sum_{n_{DUT}}^{N_D} X_{clear\pm}(t=0, n_{DUT}) \tag{8}$$

averaged over a representative number N_D of DUTs. The Golden Reference DUT selected in end-of-line testing has properties close to those initial mean values.

The initial value of the maximum positive and negative protection limits of the displacement

$$\begin{aligned} X_{p+}(t=0) &= \overline{X_{clear+}} - M_+ \\ X_{p-}(t=0) &= \overline{X_{clear-}} - M_- \end{aligned} \tag{9}$$

are the initial clearances reduced by the safety margins M_+ and M_-.

The transducer manufacturer is interested in maximizing the nominal maximum amplitude X_{AC} by placing the voice coil at the optimum rest position

$$\overline{X_0} = (L_+ + L_-)/2 \tag{10}$$

close to the geometrical center of the boundaries L_+ and L_- generating symmetrical clearances:

$$\overline{X_{clear+}} \approx \overline{X_{clear-}} \approx (L_+ - L_-)/2 \tag{11}$$

Furthermore, the transducer manufacturer spends significant effort to reduce the asymmetries in the nonlinear stiffness $K_{ms}(x)$ and force factor $Bl(x)$ and to keep the generated DC displacement $x_{DC}(t)$ small. Unfortunately, production spread and the influence of acoustical load in the audio system, aging of the suspension, gravity and climate may generate an offset in the rest position:

$$\Delta X_o(t) = X_0(t) - \overline{X_0} \tag{12}$$

This offset reduces the clearance on one side of the coil and generates asymmetries in the nonlinearities which may additionally increase the DC component $x_{DC}(t)$.

3. OVERVIEW ON PROTECTION SCHEMES

3.1. Attenuation Control

The attenuation control element in Figure 1 activates the high-pass filter which attenuates the low frequency components in the audio signal $w(t)$ to keep the following peak value of the displacement signal within the protection limits X_{p+} and X_{p+}. The mechanical attenuation control element receives the instantaneous displacement signal $x_{rel}(t)$ depending on the attenuated audio signal $v(t)$. This feedback structure can cope with the time-variant and nonlinear transfer responses of the attenuator and transducer and the properties of the instantaneous audio signal $w(t)$.

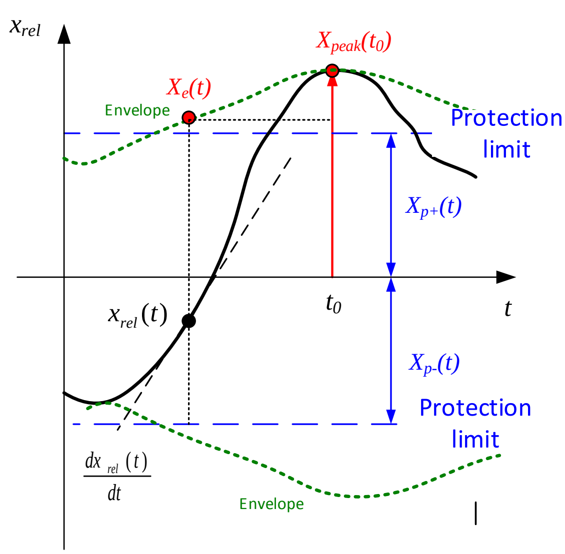

Figure 4 Anticipation of the peak displacement at an overshoot case

The attenuation control system has to cope with the restoring force and inertia of the mass-spring system generating a low-pass characteristic between terminal voltage $u(t)$ and voice coil displacement $x_{rel}(t)$. The inertia of the moving mass causes a further increase of the displacement $x_{rel}(t)$ even if the electrical input signal is switched off. The displacement maximum $x_{peak}= x_{rel}(t_0)$ occurs at the instant of time t_0 where the gradient (velocity) becomes zero and all kinetic energy W_{kin} of the coil is stored as potential energy W_{pot} in the suspension. Considering the total energy W_{tot} in the mass-spring system without external excitation and dissipation

$$W_{tot} = W_{kin} + W_{pot} = const.$$

$$\left(\frac{dx_{rel}(t)}{dt}\right)^2 \frac{M_{ms}}{2} + x_{rel}^2(t)\frac{K_{ms}}{2} = x_{peak}^2 \frac{K_{ms}}{2} \tag{13}$$

the maximum displacement

$$X_{peak}(t) = \sqrt{x_{rel}^2(t) + \left(\frac{1}{2\pi f_s}\frac{dx_{rel}(t)}{dt}\right)^2} \tag{14}$$

can be anticipated based on the instantaneous displacement $x_{rel}(t)$ at an earlier time ($t<t_0$) and the resonance frequency f_s. If the protection system would only be activated when the instantaneous displacement passes one of the protection limits $X_{p\pm}$, an overshooting over these limits could not be prevented. In order to avoid a mechanical overload the protection limits have to be reduced by a safety margin M_{act} corresponding to the expected overshoot value O_{act} in percent. Any latency in the feedback loop will further increase the activation margin M_{act}.

The undershoot value U_{act} describes the overreaction of attenuation control where the maximum displacement $x_{rel}(t_0)$ stays below the protection limits. This case does not endanger the transducer of an overload but reduces the acoustical output.

The dangerous overshooting over the protection limits and the undesired undershooting without overload can be significantly reduced by attenuating the input signal before the mechanical overload occurs. The attenuation control system gets more time to find the optimum attenuation value and to avoid impulsive distortion in the audio signal degrading sound quality. There are two different solutions for this problem:

3.1.1.Attenuation with Delay

The easiest solution provides a second attenuator which reduces the delayed audio signal *w(t-τ)* as shown in Figure 5. The attenuation control receives the audio signal $w(t)$ without any time delay via the first attenuator and compares the modeled displacement $x_{rel}(t)$ with the protection limits $X_{p\pm}$ and attenuates the input signal $w(t)$ and the delayed input signal $w(t-\tau)$ simultaneously. If the time delay τ is large enough the modeled displacement x_{rel} may temporally exceed the allowed protection limits but the voice coil of the transducer will stay within the permissible limits. This approach requires a transducer model and increases the latency in the audio path, which is undesired in many applications.

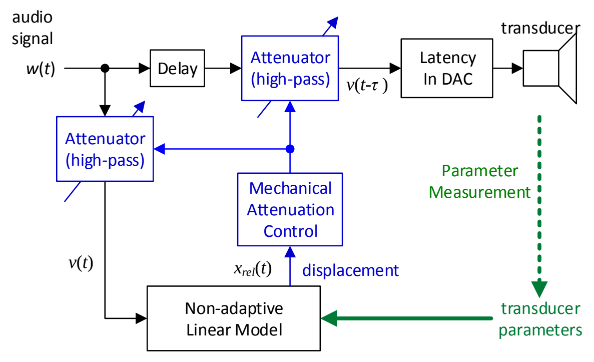

Figure 5 Protection based on transducer states predicted by a non-adaptive linear model (LM)

3.1.2. Anticipation of the Peak Displacement

The latency in the audio stream can be reduced or even completely be avoided by anticipating the peak displacement X_{peak} as suggested by Klippel [9]. This invention exploits the inertia of the transducer generating a smooth displacement waveform corresponding to a low-pass filtered audio signal.

The energy balance in Eq. (14) may be used to anticipate peak value X_{peak} but this approach neglects the influence of the audio signal $w(t)$ and the effect of the attenuator.

A more accurate estimate of the peak value X_{peak} is the instantaneous envelope

$$X_{peak}(t) = \sqrt{x_{rel}^2(t) + x_c^2(t)} \tag{15}$$

using the analytic continuation $x_c^2(t)$ calculated by the Hilbert transformation [19]

$$x_c(t) = \frac{1}{\pi}\int_{-\infty}^{+\infty}\frac{x_{rel}(\tau)}{t-\tau}d\tau \tag{16}$$

or approximated by all-pass filters [20]. The envelope provides accurate estimates of the peak value for steady-state excitation condition. To cope with impulsive audio signals more advanced prediction models are required which describe the acceleration and deceleration of the coil in transient excitation phases [11].

3.2. State Information

The attenuation control element needs accurate information on the instantaneous voice coil position as early as possible to activate the attenuator in time. This information can be provided either by direct state measurement or by indirect state modeling combined with a parameter measurement.

3.2.1. Measurement

A displacement sensor depicted in Figure 6 provides the measured voice coil position $x_{meas}(t)$ with minimum delay to the mechanical attenuation control system.

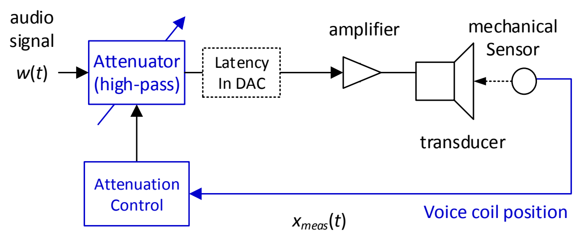

Figure 6 Protection based on measured transducer states

Averaging of the measured signal $x_{meas}(t)$ over a large time window ($T_m >> 1/f$) gives the instantaneous voice coil rest position

$$X_{0,meas}(t) = \frac{1}{T_m}\int_0^{T_m} x_{meas}(t-\tau)d\tau \tag{17}$$

,

and the voice coil displacement:

$$x_{rel}(t) = x_{meas}(t) - X_{0,meas}(t) \tag{18}$$

The initial mean rest position defined by

$$\overline{X_0} \approx \overline{X_{0,meas}} = \frac{1}{N_D}\sum_{n_{DUT}}^{N_D} X_{0,meas}(t=0, n_{DUT}) \tag{19}$$

can be determined by measuring the Golden Reference DUT representing the mean production yield.

The varying rest position detected by state measurement gives the instantaneous clearances

$$\begin{aligned} X_{clear+}(t) &= \overline{X_{clear+}} - \Delta X_o(t) \\ X_{clear-}(t) &= \overline{X_{clear-}} + \Delta X_o(t) \end{aligned} \tag{20}$$

depending on the measured offset in the voice coil rest position

$$\Delta X_o(t) = X_{0,meas}(t) - \overline{X_{0,meas}} \tag{21}$$

defined as the difference between instantaneous and initial rest position.

The total safety margins for the state measurement technique

$$\begin{aligned} M_+ &= M_m + M_{act} \\ M_- &= M_m + M_{act} \end{aligned} \tag{22}$$

only comprise a first component M_m coping with the measurement error

$$e_{meas}(t) = x_{abs}(t) - x_{meas}(t) \tag{23}$$

and the activation margin M_{act} required to cope with inertia of the moving transducer parts and the delay in the audio path (DAC) and measurement path (ADC). The margin M_{act} for state measurement can only be reduced by anticipating the peak value in accordance with chapter 3.1.2.

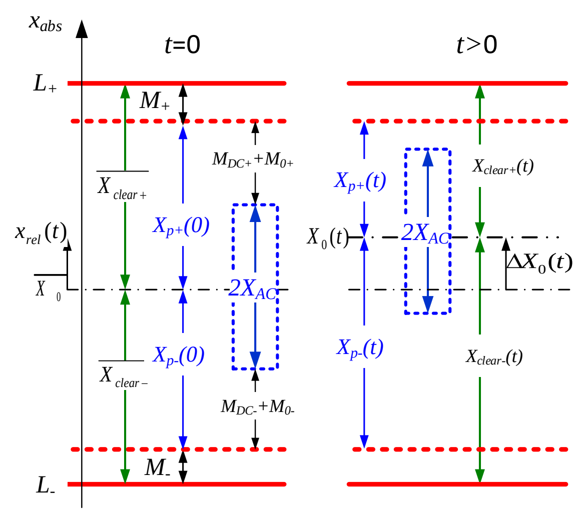

Figure 7 Protection limits for state measurement (SM) and adaptive nonlinear modeling (ANM).

The maximum positive and negative protection limits of the displacement

$$\begin{aligned} X_{p+}(t) &= X_{p+}(0) - \Delta X_0(t) \\ &= \overline{X_{clear+}} - M_+ - \Delta X_0(t) \\ X_{p-}(t) &= X_{p-}(0) + \Delta X_0(t) \\ &= \overline{X_{clear-}} - M_- + \Delta X_0(t) \end{aligned} \tag{24}$$

are not constant but vary with the offset $\Delta X_0(t)$ over time as illustrated in Figure 7. The voice coil offset $\Delta X_0(t)$ and a DC displacement $x_{DC}(t)$ requires no additional safety margins $M_0\pm$ and $M_{DC}\pm$, respectively, because they are directly monitored in the measured absolute voice coil position. However, both effects reduce the

AC displacement and the acoustical output if the coil move to one of the boundaries.

The nominal maximum amplitude

$$X_{AC} = Min\begin{pmatrix} (1-U_{act})X_{p-}(0) - M_{DC-} - M_{0-} \\ (1-U_{act})X_{p-}(0) - M_{DC+} - M_{0+} \end{pmatrix} \qquad (25)$$

can be derived from the minimum of the negative and positive protection limits $X_p\pm$ reduced by the undershoot factor U_{act} and margins $M_{DC}\pm$ and $M_0\pm$ describing the maximum shift of the voice coil rest position $\Delta X_0(t)$ and maximum DC displacement.

3.3. Transducer Modeling

The attenuation control system in section 3.1.1 requires an accurate transducer model to provide the voice coil displacement $x_{rel}(t)$ earlier before this state variable is generated by the transducer. A linear model with fixed parameters is the simplest solution but neglects the time variance and transducer nonlinearities. Adaptive schemes provides updated estimates of the transducer parameters and simplify the initial identification. Nonlinear modeling and active linearization provides the highest accuracy and copes with DC displacement and offset of the rest position.

3.3.1. Non-adaptive Linear Modeling

The voice coil displacement

$$x_{rel}(t) = x_{lin}(t) \in S_{LDE}(v(t), \mathbf{P}_{lin}) \qquad (26)$$

is a part of the solution S_{LDE} of a set of linear differential equations (LDE) with time invariant parameters $\mathbf{P}_{lin}(t) = \mathbf{P}_{lin}(t=0)$. This model can be realized as a digital IIR-filter receiving the attenuated audio signal $v(t)$ as an input and generating a linear approximation $x_{lin}(t)$ of the displacement x_{rel}.

The protection limits

$$\begin{aligned} X_{p+}(t) &= X_{p+}(0) = \overline{X_{clear+}} - M_+ \\ X_{p-}(t) &= X_{p-}(0) = \overline{X_{clear-}} - M_- \end{aligned} \qquad (27)$$

are time invariant and depend only on the initial clearances and the total safety margins:

$$\begin{aligned} M_+ &= M_{AC} + M_{0+} + M_{DC+} + M_{act} \\ M_- &= M_{AC} + M_{0-} + M_{DC-} + M_{act} \end{aligned} \qquad (28)$$

The first three components of the total margin correspond to the positioning error

$$\begin{aligned} & x_{abs}(t) - (x_{lin}(t) + \overline{X_0}) \\ & = e_{AC}(t) + \Delta X_o(t) + x_{DC}(t) \end{aligned} \qquad (29)$$

caused by modeling error $e_{AC}(t)$, offset in voice coil rest position $\Delta X_o(t)$ and a DC displacement $x_{DC}(t)$ generated dynamically by transducer nonlinearities.

The AC margin

$$M_{AC} = M_{AC,spread} + M_{AC,time} \qquad (30)$$

copes with the modeling error of the AC displacement

$$e_{AC}(t) = e_{spread}(t) + e_{time}(t) + e_{dist}(t) + e_{res}(t) \qquad (31)$$

considering the variation of the transducer properties due to production spread and temporal changes of the transducer parameters over time. The nonlinear signal $e_{dist}(t)$ comprises harmonics and intermodulation distortion. The nonlinear distortion at frequencies above resonance frequency provide only a small contribution to the total displacement and can be neglected in the context of mechanical protection. However, the nonlinear distortion at excitation frequencies ($f < f_s$) may cause a nonlinear compression C of the fundamental component (typically 2 dB < C < 6 dB). This effect is not considered in the safety margin in Eq. (30) because it reduces the acoustical output and relaxes the mechanical load. The residual error $e_{res}(t)$ represents the visco-elastic behavior, hysteresis and other mechanisms which are not considered in current modeling.

The margins

$$\begin{aligned} M_{0+} &= M_{0+,spread} + M_{0+,time} \\ M_{0-} &= M_{0-,spread} + M_{0-,time} \end{aligned} \qquad (32)$$

correspond to the varying rest position

$$X_0(t) = \overline{X_0} + X_{0,spread}(DUT) + X_{0,time}(t) \qquad (33)$$

depending on the particular device (DUT) and time t.

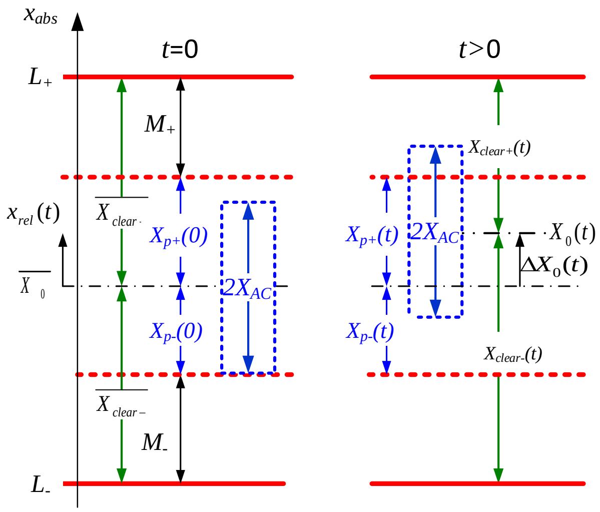

Figure 8 Protection limits for non-adaptive linear modeling (LM) and adaptive linear modeling (ALM).

The linear model used in LM protection scheme cannot model monitor the nonlinear compression and has the tendency to predict a larger displacement and to activate the attenuator without an overload. The resulting undershoot reduces the acoustical output and the nominal maximum amplitude

$$X_{AC} = Min\begin{pmatrix}(1-U_{act})F_{com}X_{p+}(0), \\ (1-U_{act})F_{com}X_{p-}(0)\end{pmatrix} \tag{34}$$

which is the minimum of the negative and positive protection limits $X_p\pm$ reduced by the reduction factor

$$F_{com} = 10^{-C/20dB} \tag{35}$$

corresponding to the nonlinear compression C in dB and the undershoot factor U_{act} of the attenuation control.

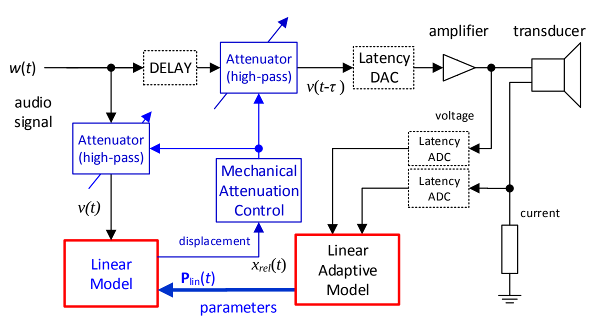

Figure 9 Protection based on transducer states predicted by an adaptive linear model (ALM)

3.3.2. Adaptive Linear Modeling

The protection system based on linear adaptive modeling (ALM) as shown in Figure 9 provides the voice coil displacement

$$x_{rel}(t) = x_{lin}(t) \in S_{LDE}(v(t), \mathbf{P}_{lin}(t)) \tag{36}$$

as the solution of the linear differential equation using time varying linear parameters $\mathbf{P}_{lin}(t)$ which are permanently updated by measured voltage and current at the transducer terminals while reproducing an audio signal.

Like the LM scheme, the ALM protection scheme uses constant protection limits $X_{p\pm}(t)$=const. defined in Eq. (27) and the safety margins

$$\begin{aligned} M_+ &= M_{0+,spread} + M_{0+,time}M +_{DC+} + M_{act} + M_m \\ M_- &= M_{0-,spread} + M_{0-,time} + M_{DC-} + M_{act} + M_m \end{aligned} \tag{37}$$

corresponding to the positioning error

$$\begin{aligned} &x_{abs}(t) - (x_{lin}(t) + \overline{X}_0) \\ &= e_{AC}(t) + \Delta X_o(t) + x_{DC}(t) + e_m(t) \end{aligned} \tag{38}$$

comprising error components as discussed earlier related to Eq. (29) for the LM protection scheme. The margin M_m represents the adaptive modeling error $e_m(t)$ which may be increased as long as the model has not learned the varying transducer properties, climate conditions and sudden changes in the acoustical environment. The margin M_m in Eq. (37) is much smaller than the margin M_{AC} required in the LM scheme because variations of the linear parameters are identified after a short learning time. The AC error $e_{AC}(t)$ in Eq. (38) describes the nonlinear compression of the AC signals while the time variance of the linear parameters is compensated by the adaptive scheme. The DC displacement $x_{DC}(t)$ and the offset $\Delta X_0(t)$ provide the largest contribution to the safety margins.

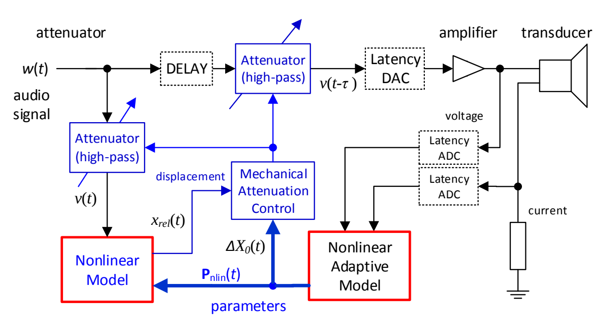

Figure 10 Protection based on transducer states predicted by an adaptive nonlinear model (ANM)

Consequently, the ALM protection scheme usually provides a larger nominal maximum amplitude X_{AC} in accordance with Eq. (34) compared to the LM scheme.

3.3.3. Adaptive Nonlinear Modeling

The ANM protection scheme generates the most accurate estimate of the voice coil displacement

$$x_{rel}(t) = x_{nlin}(t) \in S_{NDE}(v(t), \mathbf{P}_{nlin}(t)) \quad (39)$$

based on the solution S_{NDE} of the nonlinear differential equations (NDE). The nonlinear model in protection system is provided with the time varying parameters $\mathbf{P}_{nlin}(t)$ provided by a second adaptive model monitoring voltage and current at the transducer terminals as depicted in Figure 10.

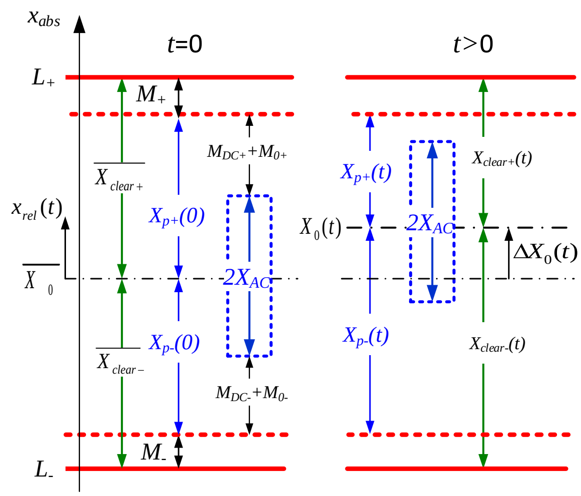

Figure 11 Protection limits for state modeling with Adaptive Nonlinear Modeling (ANM).

Based on the permanently updated parameters $\mathbf{P}_{nlin}$ the nonlinear model is capable of predicting the DC displacement, the nonlinear compression of the AC signals and time variant changes of the transducer properties.

Thus the total safety margins

$$\begin{aligned} M_+ &= M_{act} + M_m \\ M_- &= M_{act} + M_m \end{aligned} \quad (40)$$

require only the activation margin M_{act} and the adaptive modeling margin M_m corresponding to the positioning error:

$$x_{abs}(t) - x_{nlin}(t) = \overline{X}_0 + e_m(t) \quad (41)$$

The adaptive modeling error $e_m(t)$ comprises a transient component which will vanish after convergence of the parameter updating and a residual component which corresponds to limitations of the nonlinear modeling.

The attenuation control receives the instantaneous offset $\Delta X_0(t)$ from the nonlinear adaptive model and determines the time variant clearances $X_{clear\pm}(t)$ and protection limits $X_{p\pm}(t)$ according to Eqs. (20) and (24), respectively.

However, a large offset $\Delta X_0(t)$ in the rest position and a dynamic DC displacement $x_{DC}(t)$ generated by rectification of the audio signal will increase the asymmetry of the protection limits (for $t > 0$) as illustrated on the right hand side of Figure 11 and reduce the nominal maximum amplitude X_{AC} in accordance with Eq. (25) and the acoustical output. Both state measurement (SM) and adaptive nonlinear modeling (ANM) provide a similar accuracy in the predicted and measurement displacement. However, the ANM scheme only can cope with latency in the feedback of the transducer information because the parameters vary much slower than the activation time of the attenuation control.

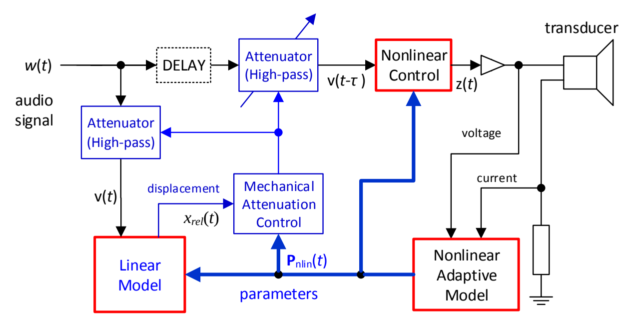

Figure 12 Protection based on Adaptive Nonlinear Control (ANC)

3.3.4. Adaptive Nonlinear Control

The ANC scheme combines mechanical protection with nonlinear transducer control to equalize, linearize and stabilize the transducer for any audio input. The control is based on a feed-forward filter structure [22] having a mirror symmetry to the nonlinear transducer model. This filter, performing an inverse preprocessing of the input signal $v(t\text{-}\tau)$, requires the parameters $\mathbf{P}_{nlin}(t)$ provided by the nonlinear adaptive model as illustrated in Figure 12. Using a DC coupled amplifier, the control can also generate a small DC voltage based on the identified offset $\Delta X_0(t)$ to keep the coil at the desired initial rest position $\overline{X}_0$ as discussed in [23].

By compensating all signal distortion caused by time variant and nonlinear properties the series connection nonlinear controller and transducer becomes a linear system with constant properties. Thus the voice coil displacement

$$x_{rel}(t) = x_{lin}(t) \in S_{LDE}(v(t), \mathbf{P}_{nlin}(0)) \quad (42)$$

can be modeled by a linear IIR-filter which is part of the solution S_{LDE} of the linear differential equation (LDE) using the initial parameters $\mathbf{P}_{\mathrm{nlin}}(t=0)$.

The error in the voice coil position

$$x_{abs}(t) - x_{lin}(t) = \overline{X_0} + e_m(t) \tag{43}$$

requires only safety margins M_{act} and M_m for the activation of the attenuator and the adaptive parameter measurement in accordance with Eq. (40). Synthesizing a DC voltage which compensates the offset in rest position

$$\Delta X_0(t) = X_0(t) - \overline{X_0} \approx 0 \tag{44}$$

the clearances

$$\begin{aligned} X_{clear+}(t) &= \overline{X_{clear+}} \\ X_{clear-}(t) &= \overline{X_{clear-}} \end{aligned} \tag{45}$$

and the protection limits

$$\begin{aligned} X_{p+}(t) &= X_{p+}(0) = \overline{X_{clear+}} - M_+ \\ X_{p-}(t) &= X_{p0}(0) = \overline{X_{clear-}} - M_- \end{aligned} \tag{46}$$

correspond to the initial values corresponding to the target performance of the transducer.

If the designed prototype and the mean production yield (Golden DUT) exhibits symmetrical clearances, the ANC protection scheme provides the maximum acoustical output corresponding to the nominal maximum amplitude

$$X_{AC} = Min\begin{pmatrix} (1-U_{act})X_{p+}(0), \\ (1-U_{act})X_{p-}(0) \end{pmatrix} \tag{47}$$

considering the undershoot factor U_{act} as illustrated in Figure 13.

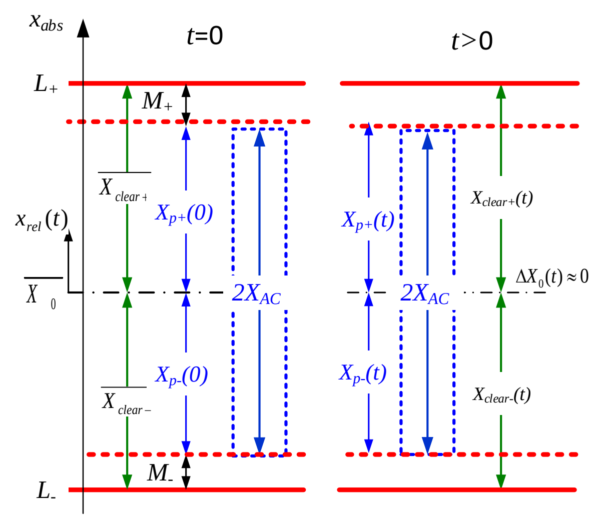

Figure 13 Protection limits for Adaptive Nonlinear Control (ANC).

4. SPECIFICATION OF PROTECTION PARAMETERS

After presenting a general framework for the important protection schemes this section addresses the communication between transducer manufacturer, software provider and system integrator as illustrated in .

4.1. Transducer Characteristics

The transducer manufacturer provides the clearances, margins and transducer parameters required by the system integrator to adjust the selected protection scheme to the particular transducer. The end-of-line test provides the "*Golden Reference DUTs*" that are the best representatives of the production yield. The mean clearance values $\overline{X_{clear-}}$ and $\overline{X_{clear+}}$ are determined for this Golden DUT by measuring the maximum displacement using a laser displacement sensor without generating impulsive distortion commonly known as "*rub&buzz*" in the sound pressure output signal. Starting at small amplitudes the test is repeated with increased amplitudes until both the crest factor of the high-pass filtered distortion exceeds 12 dB and the peak distortion value exceeds 1%. The crest factor of the impulsive distortion [21] plotted versus voice coil displacement indicates at which voice coil position the impulsive distortion is generated. This information is the basis for finding the root cause (bottoming, voice coil rubbing, limitations by the suspension, …) and a meaningful specification of the clearances.

The minimum of the upper and lower clearance values corresponds to the mechanical peak displacement

$$X_{mech} = Min\left(\overline{X_{clear+}}, \overline{X_{clear-}}\right) \qquad (48)$$

as defined by IEC standard [26].

The value X_{mech} or a value specified by the customer is used as the target displacement X_{target} for adjusting the stimulus [21] in other tests required to define the safety margins.

The positive and negative safety margins $M_{DC\pm}$ of the maximum DC displacement can be measured by exciting the Golden DUT with sinusoidal signals at target displacement.

A long-term test also known as "power test" is required to measure the shift $\Delta f_{s,spread}$ of the resonance frequency over time and to specify the corresponding margin

$$M_{AC,time} = X_{target}\left(\frac{\Delta f_{s.time}}{f_s}\right)^2 \qquad (49)$$

which assesses the increase of the peak displacement due to aging and fatigue increasing the suspension compliance.

The long-term test also provides the safety margin $M_{0,\pm}$ which corresponds to the shift of the voice coil rest position over time.

The end-of-line test provides the safety margins $M_{0,spread\pm}$ and $M_{AC,spread\pm}$ representing the production spread of the voice coil rest position and resonance frequency f_s. If those parameters are checked by a pass-fail decision, the margin $M_{0,spread\pm}$ corresponds to the production limit of the permissible offset of voice coil rest position and the margin

$$M_{AC,spread} = X_{target}\left(\frac{\Delta f_{s,spread}}{f_s}\right)^2 \qquad (50)$$

corresponding to the allowed spread $\Delta f_{s,spread}$ of the resonance frequency f_s used in the pass-fail decision.

The mean value of the lumped parameters measured in end-of-line testing or the measured parameters of the Golden DUT give the initial parameters $\mathbf{P}(t=0)$ provided to the system integrator.

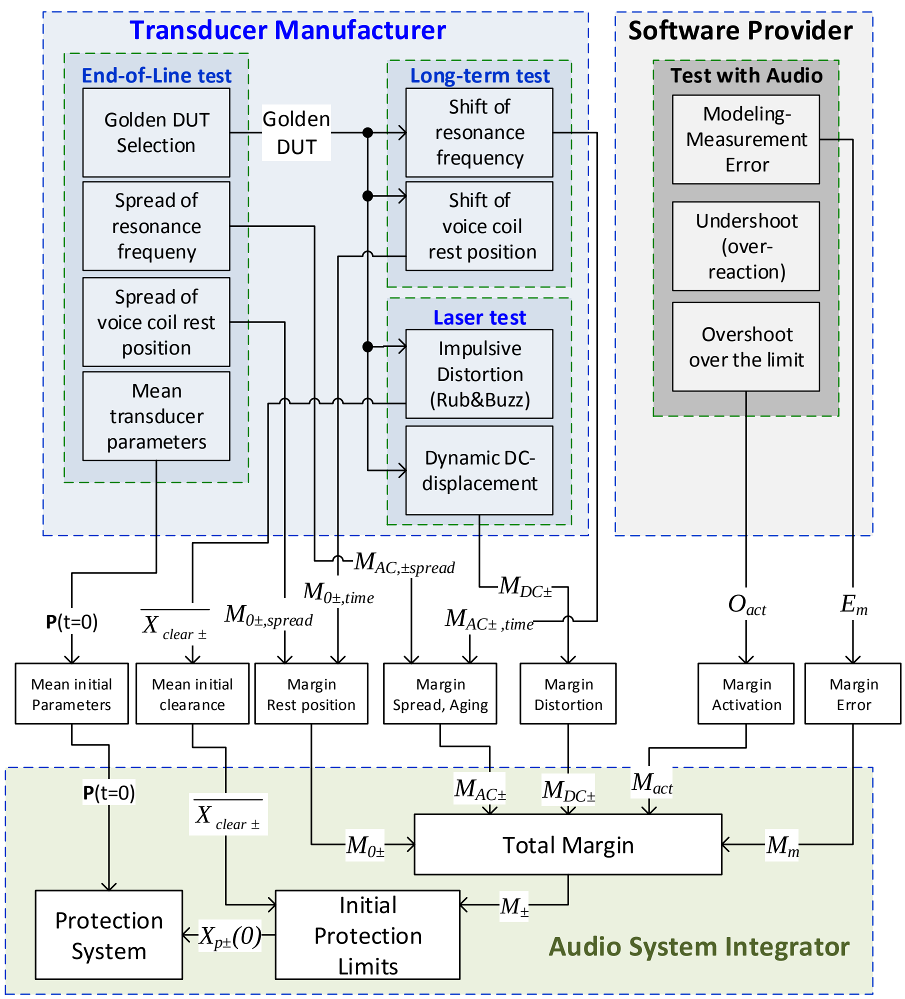

Figure 14 Work flow for the determination of the protection parameters

4.2. Software Provider

The developer of the protection algorithm specifies the overshoot, undershoot as well as the adaptation and measurement error of the protection system by relative percentage values O_{act}, U_{act} and E_m, respectively. Multiplying overshoot O_{act} and error E_m with the target peak displacement X_{target} in Eq. (48) gives the activation margin M_{act} and the error margin M_m in mm, respectively.

4.3. System Integrator

The initial protection parameters $X_{p+}(t=0)$ and $X_{p-}(t=0)$ used by the software algorithm can only be determined by the system integrator who has access to hardware and software characteristics.

The protection schemes and adjustment procedure have been discussed so far only to avoid damage and bottoming, voice coil rubbing and other impulsive distortion, which are inacceptable by the end user. This approach satisfies the current needs in personal audio and communication systems where maximum loudness, reliability, cost, weight, height and other geometrical constraints are more important than sound quality.

However, transducers developed for home, automotive, professional and other applications have a much larger clearance and usually use a spider in the mechanical suspension, which improves the stability with respect to voice coil rubbing. Here the intermodulation distortion generated by the motor geometry (voice coil height and gap depth limits) also limits the maximum output. The maximal peak displacement X_{max} defined in accordance with IEC [26] and AES [25] is limited by a certain amount of nonlinear distortion acceptable for the particular application (typically 10 %).

The protection schemes presented in the paper can also be used as a compressor of the low frequency signal components to keep the generated harmonic and intermodulation distortion below permissible limits. The system integrator determines the maximal peak displacement X_{max} based on the decay of the nonlinear force factor $Bl(x)$ and compliance $C_{ms}(x)$ as proposed in [24] and replaces the initial clearances by the X_{max} value if this value limits the maximum output.

5. DISCUSSION

The performance of the different protection schemes presented in section 3 are illustrated on a data set given in Table 1 representing typical micro-speakers as used in smartphones.

Table 1: Specification of the transducer

Transducer Characteristics	**Symbol**	**Value**
Initial clearances	$\overline{X}_{clear+}$	500 µm
	$\overline{X}_{clear-}$	450 µm
Rest position offset margins	M_{0+}	20 µm
	M_{0-}	50 µm
DC-displacement margin	M_{DC+}	10 µm
	M_{DC-}	80 µm
Linear AC-Displacement margin	M_{AC}	150 µm
Maximum nonlinear compression	C	2 dB

The initial clearances found on the Golden DUT are almost symmetrical in the example showing that the coil's initial rest position is well placed between the boundaries. A high negative safety margin M_{DC} is required to cope with negative DC component in the displacement generated by the rectification of the audio signal. The rectification may be caused by a distinct asymmetry in the force factor characteristics Bl(x) as found in most micro-speakers which use no voice coil former to place the coil symmetrically in the magnet gap. The nonlinear stiffness characteristic $K_{ms}(x)$ may also have a significant asymmetry corresponding to the shape of a single corrugation role of the diaphragm used as suspension in micro-speakers. The DC displacement may also cause a permanent offset in the voice coil rest position due to the viscoelastic memory effect in the material which is visible when the audio signal vanishes and no DC force is generated by the rectification process anymore.

Table 2: Specification of the protection algorithms

Characteristics (Symbol)	**Protection Scheme**

		SM	LM	ALM ANM ANC
Additional delay of the audio signal (in ms)	τ	0	3	3
Maximum measurement or adaptation error (in percent)	E_m	2	0	10
Maximum overshoot (in percent)	O_{act}	30	5	5
Maximum undershoot (in percent)	U_{act}	50	10	10

The software provider specifies an additional time delay and other relative performance characteristics for the protection algorithm. These characteristics are independent of the transducer but based on assumptions and limitations of the state measurement, adaptive identification and attenuation control. Table 2 gives typical characteristics describing the performance of the protection algorithms. In the current example all protection schemes (LM, ALM, ANM, ANC) which are based on modeling the voice coil displacement are using a relatively large time delay τ= 3ms in the audio stream to cope with the inertia of the coil and to keep the overshooting and undershooting low as discussed in section 3.1.1. An additional time delay cannot be applied to the protection scheme based on state measurement (SM) resulting in a poorer performance with respect to overshooting and undershooting. The anticipation of the peak value according to section 3.1.2 is not used in this example but this technique is capable of avoiding any additional delay in the audio signal with superior performance of the attenuation control.

5.1. Protection Limits

The spreadsheet in Table 3 illustrates how the system integrator generates the initial protection limits $X_{p+}(0)$ and $X_{p-}(0)$ and the nominal AC displacement X_{ac} based on the information from transducer and software supplier.

The high margins M_{0-} and M_{DC-} representing the offset of the voice coil rest position and the maximum DC-displacement, respectively, cause an asymmetry in the total margins and in the initial protection limits with $X_{p-}(0)<X_{p+}(0)$ for the protection schemes based on linear modeling (LM and ALM).

The state measurement (SM) and nonlinear protection schemes (ANM, ANC) do not require the margins M_{0-} and M_{DC-} because those schemes cope with a varying coil offset and a DC component in different ways:

The protection schemes without nonlinear control (SM and ANM) can only reduce the protection limits $X_{p+}(t)$ and $X_{p-}(t)$ according to Eq. (24) if such a behavior is detected. The minimum values of the protection values $Min(X_{p+})$ and $Min(X_{p-})$ describe the worst case scenario. The nonlinear adaptive control (ANC) compensates actively for this undesired behavior and keeps the coil within the initial (and constant) protection limits $X_{p+}(0)$ and $X_{p-}(0)$.

Only the non-adaptive linear modeling (LM) requires a high margin M_{AC} for coping with the time variant mechanical compliance $C_{ms}(x=0,t)$ due to visco-elasticity, aging, climate and other influences.

The activation margin M_{act} and the measurement/ adaptation margin M_m are calculated by multiplying the overshoot O_{act} and error E_m, respectively, given in Table 2 with target peak displacement X_{target}= 0.450 mm based on the minimum clearance in Table 1.

Table 3: Spreadsheet for calculating the protection limits and nominal amplitude

Protection Scheme	**Side**	**Transducer Margins**			**Protection Margins**		**Total Margin**	**Protection Limits**		**Nominal Amplitude**
	upper	M_{0+}	M_{DC+}	M_{AC}	M_{act}	M_m	M_+	$X_{p+}(0)$	$Min(X_{p+})$	X_{AC}
	lower	M_{0-}	M_{DC-}				M_-	$X_{p-}(0)$	$Min(X_{p-})$	
SM (state measurement)	+				135	9	144	356	336	140
	-						144	306	256	
LM (linear, non-	+	20	10	150	22		202	298	298	106

adaptive modeling)	-	50	80				302	148	148	
ALM (adaptive, linear modeling)	+	20	10		22	45	97	403	403	200
	-	50	80				197	253	253	
ANM (adaptive, nonlinear, modeling)	+				22	45	67	433	413	227
	-						67	383	333	
ANC (adaptive, nonlinear control)	+				22	45	67	433	433	344
	-						67	383	383	

The margin M_m required to cover the temporal modeling error before adaptive protection schemes (ALM, ANM, ANC) have been converged is usually larger than the margin M_m representing the error in state measurement (SM).

5.2. Acoustical Output

The last column in Table 3 describes the nominal AC amplitude for each protection scheme in the worst case scenario which corresponds with the acoustical output at low frequencies.

The linear modeling with fixed parameters (LM) gives the lowest output because it requires a large margin M_{AC} to cope with aging and climate.

The state measurement (SM) without anticipation of the peak value according to section 3.1.2 has also a relatively low performance due to the high activation margin required to cope with overshooting and undershooting of the attenuation control.

The nominal AC amplitude X_{AC} found in the simple linear modeling (LM) can be doubled by using adaptive parameter identification (ALM) which requires no margin M_{AC} for the time variance of the linear parameters. Considering the transducer nonlinearities in the modeling (ANM) increases the negative initial protection limit P_{p-} because the margins M_{0-} and M_{DC+} are not required anymore. The ANM automatically reduces the negative protection limit $X_{p-}(t)$ if an offset $\Delta X_0(t)$ is found. However, the nominal amplitude X_{AC} is not much higher than the value found in ALM protection scheme because this characteristic describes the worst case scenario with a DUT having maximum offset $\Delta X_0(t)$ and generating maximum DC displacement. The adaptive nonlinear control (ANC) gives the largest nominal amplitude X_{AC} because it linearizes the transducer and shifts the coil to the original rest position $\overline{X}_0$.

6. CONCLUSION

The theory and the practical consideration show that the protection limits $X_{P\pm}$ and the nominal AC amplitude X_{AC}, which describes the maximum acoustical output in the worst case depend on transducer properties and the performance of the protection scheme. Those characteristics can only be generated by the system integrator based on reliable information received from transducer manufacturer and software provider. The system integrator can replace clearance values by X_{max} if the audibility of the nonlinear distortion limit the maximum displacement.

The most important transducer characteristics are the lumped parameters, clearance values and safety margins describing the variation of parameters and stability of the transducer (DC displacement). It is useful to refer all those information to Golden Reference Units which should represent the production average during end-of-line testing.

The most important characteristics of the protection algorithm are the measurement or adaptation error for the critical state variables, overshooting and undershooting of the activation control system and latency added by the protection systems.

Linear modeling with fixed transducer parameters is not capable of coping with nonlinear and time variant processes and requires large safety margins, which reduces the acoustical output considerably.

The adaptive protection schemes can be realized at low cost by monitoring voltage and current at the transducer terminals without using a mechanical sensor.

The adaptive nonlinear control (ANC) generates not only the largest AC amplitude X_{AC} and acoustical output but also reduces the nonlinear distortion. The active linearization and stabilization of the transducer also needs a reliable mechanical protection system because any overshooting over the protection limits will cause an unpredictable increase of the nonlinear compensation signal and would require power amplifiers providing higher peak voltages. There is a mutual dependency and symbiosis between protection, equalization, linearization and stabilization, which are based on the same lumped parameter model and use the same parameters identified adaptively. Adaptive nonlinear control also provides new degrees of freedom to design smaller, cost-efficient transducers generating acoustic output more efficiently.

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