OPTIMAL DESIGN OF LOUDSPEAKERS WITH NONLINEAR CONTROL

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Nonlinear loudspeaker control opens new degrees of freedom in passive driver design. Transducers having a short voice coil overhang are more sensitive and efficient at low amplitudes than transducers using a linear motor. Nonlinear distortion generated at higher amplitudes can actively be compensated by nonlinear control. This paper investigates the behavior of real loudspeakers under control using artificial test signals and music as stimuli. Design criteria are presented for active loudspeaker systems considering SPL target, power consumption, thermal and nonlinear behavior and amplifier requirements.

INTRODUCTION

Currently loudspeakers become more and more active comprising a passive transducer, electronics for amplification and signal conversion and special control software implemented in a digital signal processor.
The design of the passive speaker is a very complex issue where the engineer solves electrical, mechanical and acoustical problems and searches for the best compromise possible under given restrictions. Although the same transducer principle has been used for years there are significant progress in materials, manufacturing technologies, numerical simulation (FEM, BEM, large signal performance), measurement and quality control.
Likewise there are new techniques for the amplification of the audio signal reducing size, weight and cost per Watt. Since there is enough electrical power (> 1 kW) available for most applications the heating of voice coil and the mechanical suspension limit the permissible input power and increase the need for active loudspeaker protection. Solutions that consider the physical mechanisms in the transducer in contrast to straightforward limiters open the field of nonlinear signal processing dedicated to transducers. The compensation of the nonlinear distortion is another new challenge.

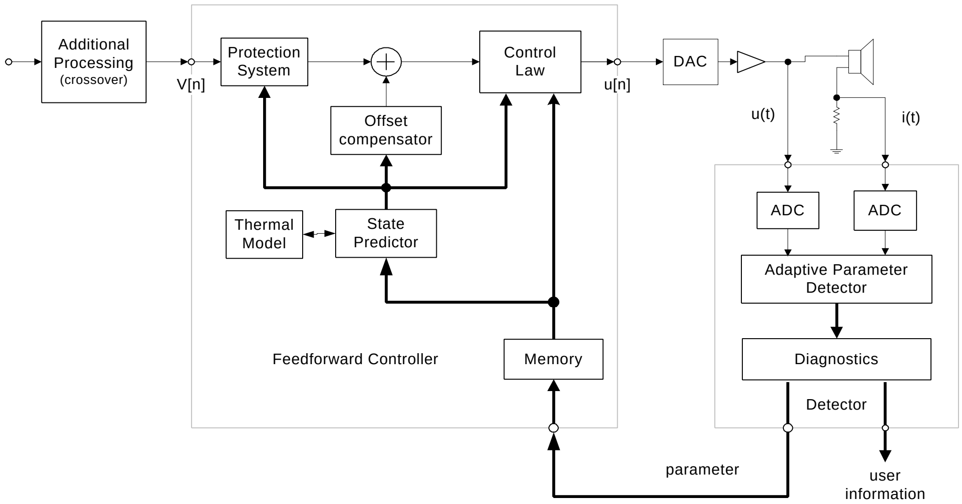

Fig. 1: Adaptive nonlinear control of loudspeaker systems

This paper addresses this particular subject and explores the practical benefit of nonlinear control. At first, a short introduction to the control technology is presented. In a second part the performance of a real loudspeaker is investigated with and without control while a normal audio signal is reproduced. This example will show the large signal behaviour of the loudspeaker, the operation of the controller and amplifier requirements. The last part of the paper searches for new ways to combine passive driver design and nonlinear control to produce more sound pressure output at sufficient sound quality.

1 NONLINEAR LOUDSPEAKER CONTROL

For many years engineers have been fascinated by the idea of using electronics to control and linearize loudspeaker system. Starting with servo control followed by generic polynomial filters based on Volterra-Series approach, finally special control architectures have been developed which exploit available knowledge about the physical mechanisms in loudspeakers [1 – 5]. An interesting approach is the adaptive control system [6] as shown in Fig. 1. It comprises a detector which identifies the loudspeaker parameters by monitoring electrical signals at the loudspeaker terminal only. The linear, nonlinear and thermal parameters are transferred to a second part which performs the pre-processing of the audio signal to realize the desired loudspeaker output.

The operation of the controller will be illustrated by using a real 5 inch driver intended for automotive applications operated here in a baffle.

The detector identifies the linear, nonlinear and thermal parameters of the driver while measuring voltage and current at the terminals. This identification can be accomplished with an ordinary audio signal or an artificial noise signal having sufficient bandwidth and amplitude. The initial identification also determines the valid working range defined by maximal peak displacement (for this driver X_{max} = 5mm) limited by the motor and suspension design and by the maximal input power P_{max} which is limited by the thermal dynamics (conduction, convection cooling).

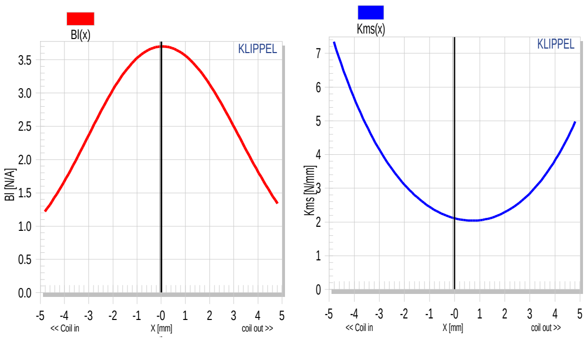

Fig. 2: Force factor Bl(x) and Stiffness $K_{ms}(x)$ versus voice coil displacement of loudspeaker under test

The nonlinear force factor characteristic Bl(x) shown in Fig. 2 reveals a motor having a coil height almost equal to the gap depth giving a distinct maximum which is almost perfectly centred at the rest position x=0. The stiffness $K_{ms}(x)$ of the suspension also shown in Fig. 2 has a distinct asymmetry, e.g. a negative displacement produces a higher restoring force than a positive displacement.

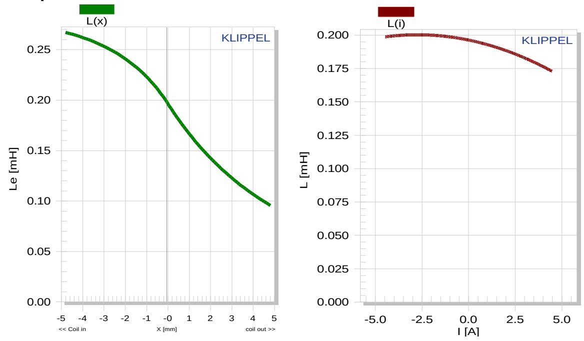

Fig. 3: Inductance L(x) versus displacement x and inductance L(i) versus current of the loudspeaker under test

The nonlinear variation of the voice coil inductance L(x) and L(i), versus displacement x and input current i are shown in Fig. 3. Since this driver has not got any means for shorting the magnetic ac-field (e.g. copper rings) the inductance falls with positive displacement where the coil moves away from the iron material. The variation of the inductance versus current is small because the ac-flux is much smaller than the dc-flux generated by the magnet.

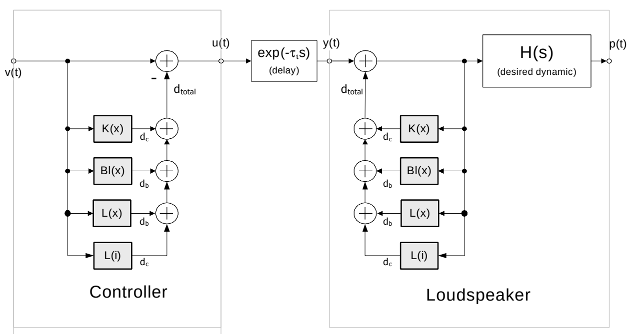

Fig. 4: Transducer-oriented control structure

The compensation of the signal distortion generated by the four nonlinearities and the linearization of the loudspeaker can be illustrated on the signal-flow chart depicted in Fig. 4. This block diagram shows only the most important features of the control technology. First the loudspeaker is represented by a linear subsystem H(s) and four nonlinear subsystems which produce the nonlinear distortion signals $d_c(t)$, $d_b(t)$, $d_l(t)$ and $d_{li}(t)$ which correspond with the nonlinear parameters $K_{ms}(x)$, Bl(x), L(x) and L(i), respectively. Note, that the nonlinear subsystems are not directly identical with the nonlinear parameters but comprise all four static nonlinearities and further linear subsystems to generate the state signals displacement, velocity and current. Without going into further details (interested readers are referred to [6]) the block diagram illustrates precisely that the generated distortion signals are added to the input signal y(t) and are fed back to the input of nonlinear subsystems. The distorted input signal is also supplied via the linear system H(x) to the loudspeaker output. The linear system is effectively a 2^{nd}-order or higher-order high-pass defined by traditional linear parameters (Thiele/Small parameters). If the amplitude of the stimulus is sufficiently small the nonlinear parameters will not vary with displacement x and current i and the nonlinear distortion are negligible compared to the input signal $v_i(t)$. Thus, the linear system H(s) alone determines the transfer behavior in the small signal domain. At higher amplitudes the feedback loop becomes active and the distortion components will also affect their own generation process. This generates some interesting effects such as nonlinear compression, instabilities, jumping effects (bifurcation) and chaos at higher amplitudes. It is also responsible for some complicated interactions between different nonlinearities. For example a poor suspension with an asymmetrical stiffness characteristic may produce a dc-displacement which moves the coil away from the Bl-maximum resulting in significant intermodulation distortion.

After having the nonlinear subsystems separated from the linear subsystems it is not difficult to design a nonlinear control system which compensates for the distortion and linearizes the overall transfer behavior. According to the mirror filter approach we have to do just the inverse processing in the electrical domain, that means we have to synthesize the same distortion signals $d_c(t)$, $d_b(t)$, $d_l(t)$ and $d_{li}(t)$ and to subtract them from the electrical control input v(t). All of the loudspeaker distortion (e.g the nonlinear terms in the differential equation) will be compensated and there will be a linear relationship between the control input v and the sound pressure p(t) and all related state signals (displacement x, velocity v …). This can be exploited for synthesizing the compensation distortion not from the control output signal u(t) (which is typical for observer based control architectures) but directly from the control input v. Thus we end up with the feed-forward control structure as shown in Fig. 4 which is just the mirrored transducer model. Since the mirror filter has almost no feedback in the nonlinear processing the controller behaves stable under almost all conditions (any choice of the nonlinear parameters !!). Additional time delay can be added in a feed-forward controller which simplifies the digital implementation and the use of digital-analogue converters.

1.1 Large Signal Performance of the Speaker

The following measurements in the large signal domain shall illustrate the complex behavior at high amplitudes.

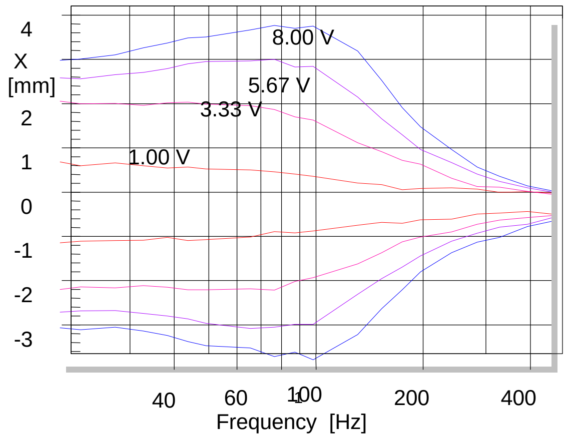

Fig. 5: Peak and bottom value of the displacement measured with a single tone stimulus varied in frequency and voltage.

The voice coil displacement is a most interesting signal because it describes the internal state of the loudspeaker and activates three of the most dominant loudspeaker nonlinearities.

Fig. 5 shows the positive and negative peak displacement while exciting the loudspeaker with a single tone varied in frequency and amplitude. Although the voltage is increased in equal steps the displacement rises at a decreasing step size. Predicting the peak displacement at 8 Volt and 20 Hz based on a linear model we would expect almost twice the value (7.5 mm) instead of 3.5

mm. Thus the loudspeaker loses sensitivity at certain frequencies in the large signal domain which can be quantified by a compression factor in dB as shown in Fig. 6.

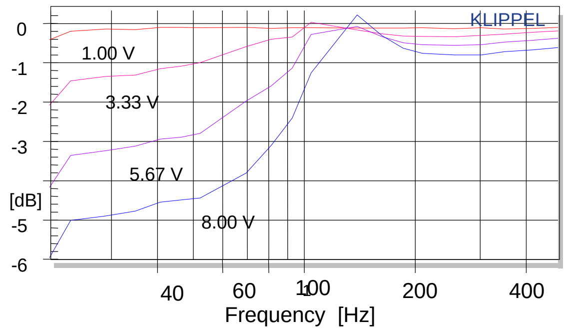

Fig. 6: Reduction of the fundamental component due loudspeaker nonlinearities measured with a single tone stimulus varied in frequency and voltage (measurement time < 1 s).

The measurements are performed by using a very short stimulus keeping the voice coil temperature almost at ambient conditions. Thus Fig. 6 reveals only the amplitude compression caused by driver nonlinearities but not by thermally increased voice coil resistance.

The nonlinear compression of the fundamental is dominant at low frequencies where voltage, current and displacement are in phase and the decay of force factor and compliance coincide. However, at the resonance frequency (f_s=140 Hz) the loudspeaker produces more output than predicted by a linear model. This is mainly caused by the loss of electrical damping and the 90 degree phase relationship between current and displacement.

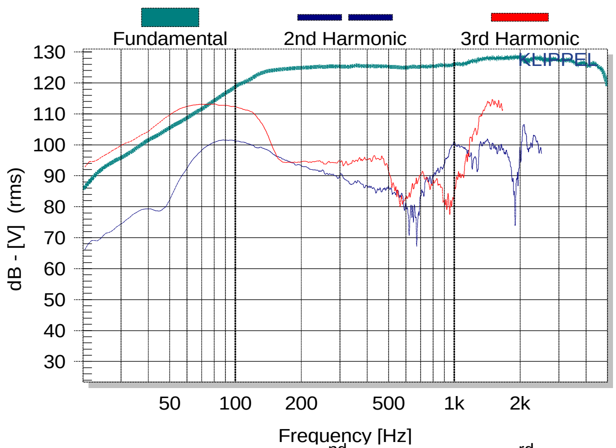

Fig. 7: Fundamental, 2^{nd}-order and 3^{rd}-order harmonic components in the sound pressure output of the test loudspeaker measured with a sweep at 8 Volts.

A common symptom of a nonlinear systems is the generation of harmonic distortion which is usually measured by a single tone varied versus frequencies. Fig. 7 shows the fundamental, the 2^{nd}-order and 3^{rd}-order component in dB (SPL) measured in the near field of the loudspeaker. This measurements reveals the effects of Bl(x) and K_{ms}(x) below the resonance frequency where the voice coil displacement is high. At higher frequencies it shows the effect of flux modulation due to L(i)-nonlinearity and nonlinear vibration of the break-up modes on the cone and suspension. The nonlinear distortion responses measured in the sound pressure output also depend on the properties of the linear system H(s) which includes the influence of the electro-mechanical transducer, cone vibration, radiation and the room if the measurements are performed in the far field. The dependency of the distortion measurements on microphone position and room properties can be avoided by transforming the distortion measured in the sound pressure output back to the electrical terminals giving equivalent input distortion [7]. The transformation can be accomplished by a simple filter with the inverse transfer function $H(s)^{-1}$ prior to the spectral analysis. The equivalent input distortion generated by the dominant nonlinearities K_{ms}(x), Bl(x), L(x) and L(i) and measured at an arbitrary microphone position are identical with the total distortion d_{total}(t) generated by the nonlinear subsystems in Fig. 4.

The concept of equivalent input distortion has a crucial importance for evaluation and design of nonlinear control system:

1. Nonlinear control can only compensate for equivalent input distortion. If the equivalent input distortion depends on the microphone position the nonlinearity may be distributed in 1a 3D space (over cone or sound field) and can not perfectly compensated by pre-processing of a single electric input signal. A similar problem is the equalization of the linear transfer function in a sound field considering speakers directivity and room reflections.
2. Equivalent input distortion can be expressed in Volts and can be directly compared with the electrical input signal. This is important for investigating the physical limits of the nonlinear control considering power handling, peak voltage and thermal load of the loudspeaker and power amplifier.

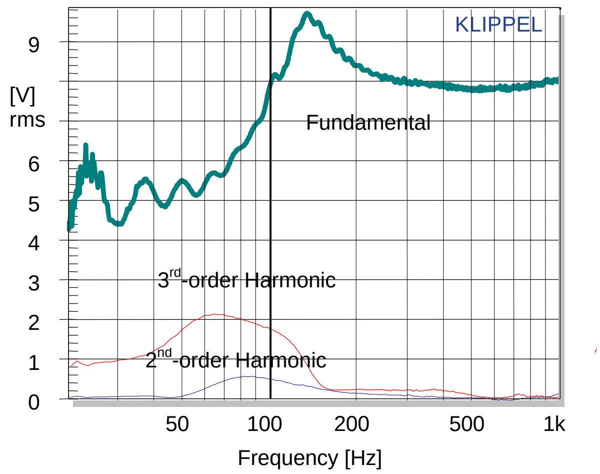

Fig. 8: Equivalent electrical components at the loudspeaker terminals: fundamental (thick line), 2^{nd}-order (dashed line) and 3^{rd}-order harmonic component (thin line).

For example, Fig. 8 shows the equivalent fundamental component (thick line) and the 2^{nd}- and 3^{rd}-order harmonics measured by a sinusoidal sweep of 8 V rms. The curves are derived from the near-field measurement in Fig. 7 but are identical with measurement at other microphone positions (where the signal-to-noise ratio might be worse). The amplitude of the equivalent fundamental is less than 5 V for f < 50 Hz which is caused by nonlinear amplitude compression as shown in Fig. 6 where a negative fundamental component in the nonlinear distortion d_{total} is added to the input. However, at the resonance frequency f_s=140 Hz the distortion component d_{total} actually increases the fundamental and produces more acoustical output than a linear system.
The harmonics in Fig. 8 are less than 2 Volts which are about 20 – 30 % of the total input. For low excitation frequencies the high-pass characteristic of the linear filter H(s) attenuates the fundamental more than the harmonics (which fall into the pass-band) producing a highly distorted acoustical output.

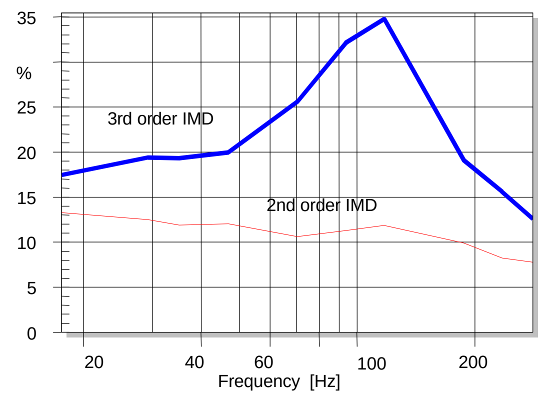

Fig. 9: 2^{nd}- and 3^{rd}-order intermodulation distortion according IEC 60268-5 measured with a two-tone stimulus (15 < f_1< 300, f_2=1000Hz)

The harmonic distortion have only significant values at low frequencies where the impact on sound quality is relatively low (if f_s < 100 Hz). However, force factor Bl(x) and inductance L(x) can generate significant amplitude modulation of any audio signal in the pass-band when a low frequency tone generates sufficient displacement. The amplitude modulation is much more critical than phase or frequency modulation distortion (caused by the Doppler effect) because it is detected by the fluctuation of the sound pressure level or variation of the envelope in the time domain and perceived as an unnatural roughness of the sound.

1.2 Loudspeaker under Control

After discussing the large signal behavior measured by using artificial test signals the state of the loudspeaker shall be investigated while an electrical control circuit pre-distort the stimulus in such a way that the voice coil displacement shows a linear relationship to the control input.
A short piece of popular music of 40 s length is used as stimulus during the experiments. The piece is repeated 15 times and the level is increased by 1 dB after each loop.

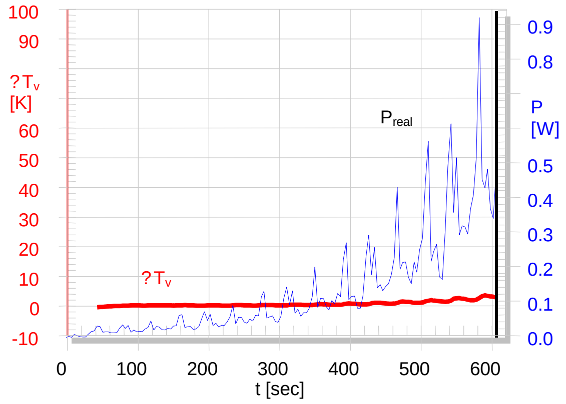

Fig. 10: Real input power (thin line) of the test loudspeaker and voice coil temperature (thick line) excited by a repeated music stimulus whereas amplitude is increased in 1dB steps.

Fig. 10 shows the real input power P_{real} of the loudspeaker and the voice coil temperature recorded during the reproduction of the music stimulus increased by 1 dB steps.

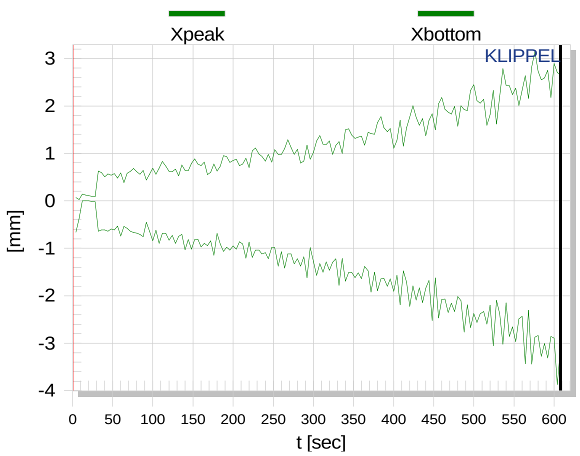

Fig. 11: Peak and bottom displacement of the voice coil while exciting the loudspeaker with a music stimulus played in a loop and increased in 1 dB steps.

Fig. 11 shows the peak and bottom displacement during the experiment. At the last loop where the input gain is maximal the negative bottom value almost reaches the maximal peak displacement X_{prot}= 4 mm where the electronic protection system would be activated and would attenuate the low frequency components automatically.

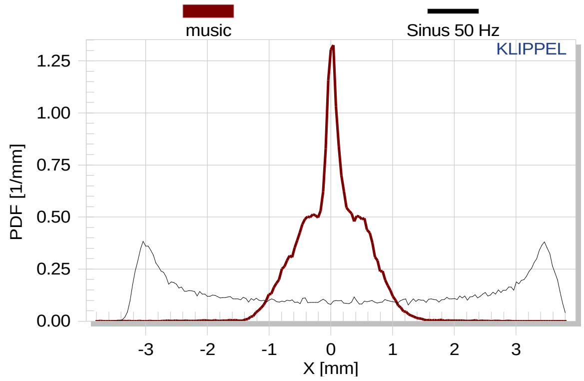

Fig. 12: Probability density function pdf(x) of voice coil displacement for a music signal (thick line) and a single tone at 50 Hz (thin line) while both stimuli produced the maximal peak displacement x_{peak}= 4 mm.

More information about the instantaneous displacement gives the probability density function pdf(x) in Fig. 12 which shows how often the voice coil is found at a certain displacement. For music there is a distinct maximum at the rest position and the coil stays most of the time in the gap. The incidents where the coil is at maximal displacement (x_{peak}= 4 mm) are rare and very short which results in a high crest factor of the voice coil displacement (> 15 dB). Thus, not much power (less than 1 W) is required to generate the peak displacement with a common music signal. Consequently, the increase of the voice coil temperature is negligible.
The situation changes for a single tone having a completely different pdf as shown in Fig. 12. Here the crest fact is low and the pdf becomes maximal at the peak displacement. About 10 Watts are required to produce x_{peak}= 4 mm at 50 Hz and the heating of the voice coil can not be neglected in long term operation.

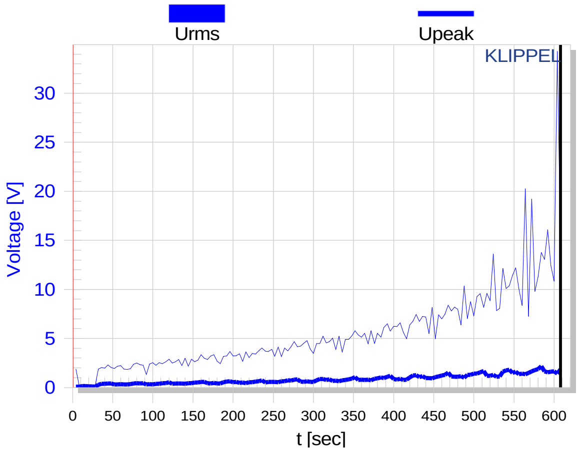

Fig. 13: Peak voltage (thin line) and rms voltage at the loudspeaker terminals with a music stimulus played in a loop and increased in 1 dB steps.

Fig. 13 shows the peak and rms-value of the voltage at the loudspeaker terminals measured with the music stimulus. In the small signal domain the control output has the same crest factor of about 15 dB as the original music signal supplied to the control input. However, at maximal input gain the controller becomes active and generates much higher voltage peaks while the rms value stays almost constant. Those voltage peaks are caused by the equivalent input distortion synthesized by the nonlinear control system in Fig. 4.

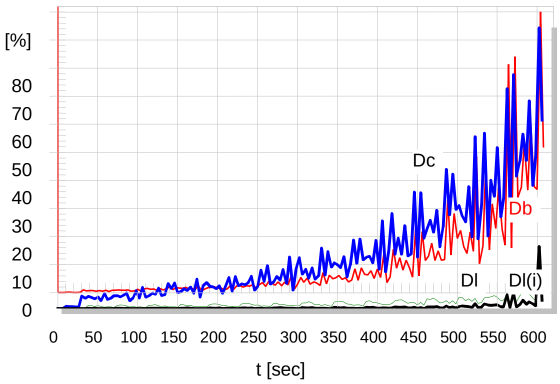

Fig. 14: Compensated nonlinear distortion components generated by stiffness (Dc), force factor (Db), displacement varying inductance (Dl) and current varying inductance (Dl(i)) measured with a repeated music stimulus increased in 1dB steps.

The digital control system makes it possible to measure the nonlinear distortion components on-line while reproducing an arbitrary signal. Fig. 14 shows the distortion component generated by force factor, stiffness and the inductance nonlinearities transformed via the linear system H(s) into sound pressure output. The ratio between the peak value of distortion component and the peak value of the total signal is a useful metric giving the instantaneous distortion in percent.

The distortion components recorded during the experiment give further interesting information on the loudspeaker. Clearly, the distortion rises with the gain of the input signal. At small amplitudes the distortion D_c of the suspension is dominant. It is mainly caused by 2nd-order distortion generated by the asymmetry in the $K_{ms}(x)$ characteristic. At higher amplitudes the distortion D_b from the force factor Bl(x) increases rapidly and come into the same order of magnitude as D_c. This is mainly caused by the symmetrical shape of the Bl-curve which produces 3rd-order distortion which rise at a higher rate than the 2nd-order. Of course these peak values do not say much about the audibility and the impact on sound quality. As mentioned above the distortion components from Bl(x) are mainly intermodulation which cover the whole audio band while the suspension distortion Kms(x) are bounded to lower frequencies.

The distortion generated by inductance nonlinearity L(x) and L(i) stays at much lower level because the resistance Re is still the dominant part in the electrical input impedance.

Comparing the distortion at the loudspeaker terminals with the equivalent distortion in the sound pressure output shows the true distortion reduction of active control system. Ideally the equivalent distortion of the loudspeaker with control should vanish completely but a distortion reduction of 10 – 20 dB can be accomplished in practice. This way of assessing the performance of the controller also considers that the controller compensates for the nonlinear compression of the fundamental component. Increasing the output of the current driver by 6 dB at low frequencies the loudspeaker itself will generate much more distortion than operated without control.

From this experiment the following observations can be summarized:

- The controller synthesizes not only harmonic and intermodulation components but also a fundamental component which compensates for nonlinear amplitude compression.

- The concept of equivalent input distortion is a convenient basis for comparing the distortion generated by the loudspeaker and the distortion synthesized by nonlinear control.

- The input power and the rms voltage highly depends on the spectral properties and the probability function of the stimulus.

- For music signals with a bell-shaped distribution of voice coil displacement the linearization does not increase the input power significantly.

- The active linearization requires power amplifiers which are capable of transferring a control output having a higher crest factor than the original stimulus at the control input.

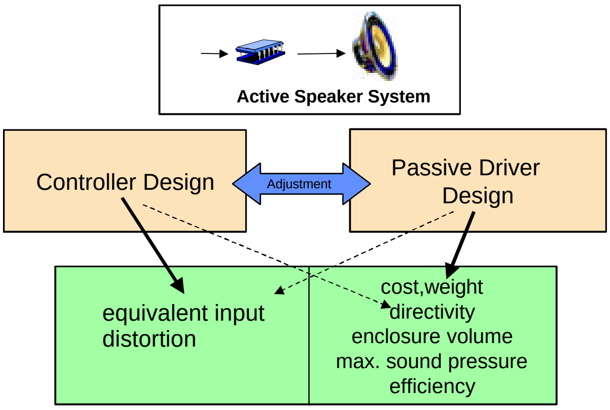

Fig. 15: Dedicated loudspeaker control gives new degrees of freedom for the passive driver design

2 OPTIMAL DESIGN OF ACTIVE LOUDSPEAKER SYSTEMS

This previous experiment has been performed on a normal driver designed for automotive applications where no nonlinear control circuit is currently available. The current driver is a compromise between realizing the required SPL output and sufficient sound quality. There are new degrees of freedom if the driver design focus on cost, weight, size, efficiency, SPL output and directivity. However, permissible nonlinear distortion, protection against thermal and mechanical overload, alignment giving a desired overall target response can be realized by electrical means. The cooperation between driver and system design as illustrated in Fig. 15 will be discussed by using different design choices.

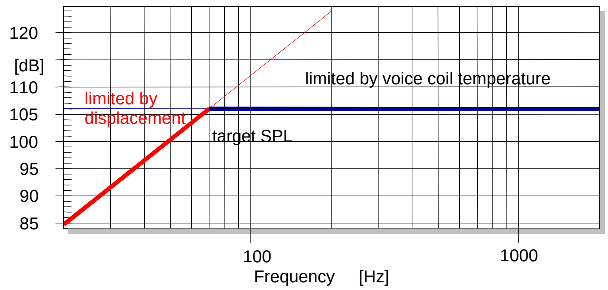

Fig. 16: Target of the large signal performance

2.1 Target Behavior

The design starts with defining the maximal SPL output level of the driver. At lower frequencies the output is limited by the peak displacement of the voice coil. Assuming that the suspension allows a free excursion of the coil up to +- 5mm peak and assuming an cone area of 314 cm^2 the volume velocity limits the sound pressure to 85 dB SPL at 20 Hz rising with 12 dB per octave up to the resonance frequency f_s=70 Hz as illustrated in Fig. 16. At higher frequencies the maximal SPL should be 106 dB which may be limited by the efficiency of the driver, the required peak voltage and thermal power handling.

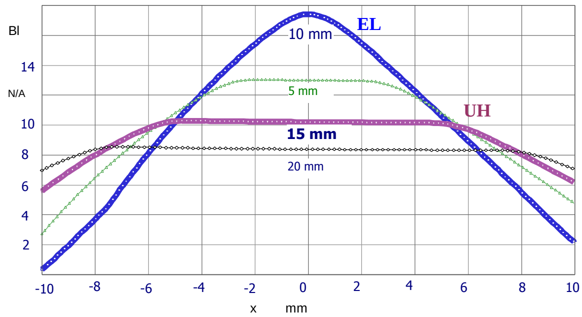

Fig. 17: Bl(x) profile while varying the gap depth while using the same voice coil height of 10 mm (Dodd [9])

2.2 Three Design Choices

This goal shall be realized by three design choices. The first design choice is a loudspeaker with a nonlinear motor design which is coupled with nonlinear control to provide active linearization. The other two design choices behave sufficiently linear in the used working range and no linearization is required. The only difference between the three design choices is the voice coil height and the gap depth. The first choice is an equal-length configuration EL where the coil length and gap depth is 10 mm each. The Bl profile has a distinct maximum of 17.2 N/A and falls almost linearly with positive and negative displacement as shown in Fig. 17. Using the same coil and magnet but increasing the gap depth gives the under-hang configuration UH where the force factor is about 10.2 N/A but almost constant within +- 5 mm. The third design choice uses the same gap and magnet as the equal-length configuration but a coil height of 20 mm. Assuming the same wire material, diameter and number of layers but neglecting the fringe field the force factor Bl(x=0)=17.2 N/A is identical with the first design choice. However, the resistance, inductance and moving mass of the coil is doubled. Besides the differences in the motor structure all three design choices uses the same suspension, cone and magnet. It is also assumed that all of the other nonlinearities Kms(x), Le(x), Le(i) are negligible. This can be accomplished by using a carefully designed suspension and proper shorting material for reducing the magnetic ac-field. All the tree design choices have the same thermal behavior represented by the thermal resistance of the voice coil.

DESIGN CHOICES	**EL** **equal-length**	**UH** **under-hang**	**OH** **over-hang**	
Coil Height	10	10	20	mm
Gap depth	10	15	10	mm
Moving Mass M_{ms}	50	50	80	g
Coil mass	30	30	60	g
Inductance Le	0.5	0.5	1	mH
Resonance Frequency fs	50	50	40	Hz
Resistance R_e	3.5	3.5	7	Ohm
Bl at rest position (x=0)	17.2	10.2	17.2	N/A
Bl at x=5mm	10	10.2	17.2	N/A
Thermal resistance R_{tv}	2	2	2	K/W
Linearization	required	-	-	
SPL at 500 Hz, 1m and 1V	89	84.1	78.6	dB

Table 1: Influence of the voice coil height and gap depth on loudspeaker parameters and other characteristics

Table 1 gives a summary on the linear, nonlinear and thermal parameters of the three design choices. The first design choice using an equal-length configuration (EL) gives the highest sensitivity 89 dB SPL in 1m distance at 500 Hz and 1V rms input but behaves nonlinearly at higher voice coil displacement. The under-hang configuration gives almost a linear behavior at the price of a 5 dB less sensitivity at low and high frequencies. The overhang configuration preserves the high Bl of the first design choice but has 10 dB less sensitivity at higher frequencies due to the doubled mass and increased electrical impedance.

3 LARGE SIGNAL PERFORMANCE OF THE DESIGN CHOICES

In this section the behavior of the three design choices is investigated for three most critical test signals and a typical music stimulus. While producing the target sound pressure output according to Fig. 16 the following criteria are compared:

The input power and the peak voltage are important for designing the power amplifier. The equivalent input distortion in Volts and a traditional distortion measure (THD or IMD) describes the contribution of the controller. Finally the amplitude compression of the driver is described by a compression factor C and the voice coil temperature after long-term excitation.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	7	7.5	4.5	W
Peak Voltage	12	10	14	V
EID comp. actively	1.5	≈ 0	≈ 0	V
THD comp. actively	43	≈ 0	≈ 0	%
Compression	0.6	≈ 0	≈ 0	dB
ΔT_{coil}	≈ 0	≈ 0	≈ 0	K

Table 2: Short-term reproduction of a single tone at f_1= 20 Hz generating peak displacement x_{peak}= 5 mm (measurement time < 1 s)

The first artificial stimulus is a single tone at 20 Hz producing maximal peak displacement. Here the drive unit with an overhang configuration requires only half of the input power because the high value and displacement independent force factor provides a good excitation. The high moving mass is not a disadvantage because the restoring force is much larger than the inertia. The equal-length configuration EL requires less power than the underhang configuration UH but the amplifier has to provide a higher peak voltage due to equivalent input distortion EID of 1.5 V. The nonlinear controller compensates not only for the THD of 43 % in the output signal but also for the amplitude compression C=0.6 dB of the fundamental component. Since the measurement has been performed with a short stimulus there is no increase of the voice coil temperature in all three cases.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	7	7.5	4.5	W
Peak Voltage	12	10	14	V
EID comp. actively	1.5	≈ 0	≈ 0	V
THD comp. actively	43	≈ 0	≈ 0	%
Compression	0.6	≈ 0	≈ 0	dB
ΔT_{coil}	12	15	9	K

Table 3: Long-term reproduction of a single tone at f_1= 20 Hz generating peak displacement x_{peak}= 5 mm (measurement time > 1 min)

Table 3 shows only minor changes if the measurement time is long and the voice coil is in thermal equilibrium. The amplitude compression at low frequencies is usually caused by the nonlinearities inherent in the driver but not by the heating of the coil.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	13	37	68	W
Peak Voltage	10	18	33	V
EID comp. actively	≈ 0	≈ 0	≈ 0	V
THD comp. actively	≈ 0	≈ 0	≈ 0	%
Compression	≈ 0	≈ 0	≈ 0	dB
ΔT_{coil}	≈ 0	≈ 0	≈ 0	K

Table 4: Short-term reproduction of a single tone f_2= 500 Hz generating 106 dB at 1 m (measurement time < 1 s)

Table 4 shows the state of the three loudspeakers while reproducing a single high-frequency tone f_2=500 Hz just in the audio band. Since there is not much voice coil displacement the equal-length configuration EL behaves almost linearly but generates the expected SPL with a fraction of the input power and with much less peak voltage. The high Bl of the overhang coil can not compensate for the increased moving mass and high input impedance compared with the under-hang configuration UH.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	14	45	100	W
Peak Voltage	11	23	56	V
EID comp. actively	≈ 0	≈ 0	≈ 0	V
THD comp. actively	≈ 0	≈ 0	≈ 0	%
Compression	0.9	2.4	4.7	dB
ΔT_{coil}	28	92	206	K

Table 5: Long-term reproduction of a single tone f_2=500Hz generating 106 dB at 1 m (measurement time > 1 min)

The advantage of the equal-length configuration EL and the disadvantage of the under-hang configuration UH becomes even worse if the coil is heated up to the steady-state temperature as shown in Table 5. Linearity realized by passive driver design increases significantly the power consumption and causes significant thermal amplitude compression.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	35	44	71.5	W
Peak Voltage	30	28	47	V
EID comp. actively	8	≈ 0	≈ 0	V
IMD comp. actively	35	≈ 0	≈ 0	%
Compression	2.5	≈ 0	≈ 0	dB
ΔT_{coil}	≈ 0	≈ 0	≈ 0	K

Table 6: Short-term reproduction of a two-tone stimulus generating for f_2= 500 Hz an SPL of 106 dB in 1 m and a peak displacement of x_{peak}= 5 mm by using f_1= 20 Hz
(measurement time < 1 s)

Applying a two-tone stimulus comprising a low frequency tone at 20 Hz which produces 5 mm peak displacement and a high-frequency tone at 500 Hz which represents the audio signal in the pass band. For this stimulus the motor with equal-length configuration generates 35 % intermodulation distortion of the high-frequency tone and an attenuation of the high frequency fundamental component by 2.5 dB. To compensate the nonlinear amplitude compression and the nonlinear distortion an equivalent input voltage of 8 Volt has to be synthesized by the active control system. Despite the active compensation the equal-length configuration requires the lowest power input and a moderate increase of the peak voltage.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	40	55	110	W
Peak Voltage	35	38	78	V
EID comp. actively	8	≈ 0	≈ 0	V
IMD comp. actively	43	≈ 0	≈ 0	%
Compression	4.7	2.7	6.3	dB
ΔT_{coil}	80	110	225	K

Table 7: Long-term reproduction of a two-tone stimulus generating for f_2= 500 Hz an SPL of 106 dB in 1 m and a peak displacement of x_{peak}= 5 mm by using f_1= 20 Hz
(measurement time > 1 min)

Applying the two-tone stimulus for a longer time the heating of the coil causes significant thermal amplitude compression in the linear design choices UH and OH. Despite the nonlinear compensation signal the equal-length configuration requires the lowest peak voltage and input power.

DESIGN CHOICES	**EL** equal-length	**UH** under-hang	**OH** over-hang	
Input Power	14	45	100	W
Peak Voltage	35	38	78	V
Compression	3.4	2.4	4.7	dB
ΔT_{coil}	28	92	206	K

Table 8: Long-term reproduction of a music stimulus generating 106 dB SPL in 1 m and a peak displacement of x_{peak}= 5 mm (measurement time > 1 min)

The large signal performance of the design choices for a normal audio stimulus (music) having a bell-shaped probability function as shown in Fig. 12 can be derived from previous simulations based on single-tone and two-tone stimulus. The long-term behavior such as power consumption, compression and voice coil temperature in Table 8 can be estimated by using the simulations based on the single 500 Hz tone presented in Table 5 because the probability of high displacement is low and only the high frequency component determines the power handling. However, the estimation of the peak voltage at the terminals requires the long-term simulation based on the two-tone stimulus in Table 7. The values in Table 8 show that the nonlinear motor with electrical distortion compensation gives the highest “effective” sensitivity, lowest power consumption, voice coil heating and thermal amplitude compression.

4 CONCLUSIONS

Nonlinear control opens new degrees of freedom in the passive driver design which can be used to realize small, lightweight loudspeaker systems which produce the expected sound pressure output at maximal efficiency and with sufficient sound quality. The important point is that the passive driver design should focus on features which can not be changed by electrical control such as

- vibration of the cone or membrane
- radiation into 3D space (directivity)
- interaction with room and acoustical environment
- size and weight of the transducer
- cost of manufacturing
- efficiency, sensitivity.

Electrical control can only help on issues close to the loudspeaker terminals where the nonlinearities are lumped in a single source adding distortion to a one-dimensional signal path. The concept of equivalent-input distortion may be helpful to distinguish problems (which may be linear or nonlinear) generated in the one-dimensional or multi-dimensional domain.

The paper showed that a driver with a nonlinear motor topology coupled with a nonlinear compensator increases the overall efficiency and sensitivity especially for music signals where the coil exploits the high force factor at the rest position. This reduces the power consumption and the heating of the coil which may be used to produce the required volume velocity by smaller cone area and reduced box size. Compensation of nonlinear distortion in music signals however increases the crest factor in the controller output. Linearization of an equal-length configuration to a moderate displacement where Bl(x) decreases to 50 % of the Bl-maximum put still moderate requirements on peak voltage provided by the power amplifier. Linearization of excessive motor nonlinearities (Bl(x) < 0.5 Bl(x=0)) is possible but requires efficient amplifiers (e.g. class D) which have low internal losses at low and medium output voltages but are capable of generating high peak voltage when nonlinear control becomes active.

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