KLIPPEL ANALYZER SYSTEM

Measurement Service Report

Database Name:	DUT-1234-example
Driver Name:	DUT-1234-example
Driver Comment:	recommended for drivers with a resonance 30 Hz < fs < 200 Hz Connect driver to SPEAKER 2 for LPM, for all other modules connect driver to SPEAKER 1

Content

Number	Name	Date	Time	Comment
1	1a LPM T/S parameter (laser)	03/31/09	16:07:30	One step method using a laser displacement sensor. Target: Measure small signal parameter and distortion with the use of a multitone stimulus. Adjust Setup: Set voltage in PP Stimulus for maximum SNR in I, U and X while keeping low distortions in I

Linear Parameter Measurement (LPM)

Overview

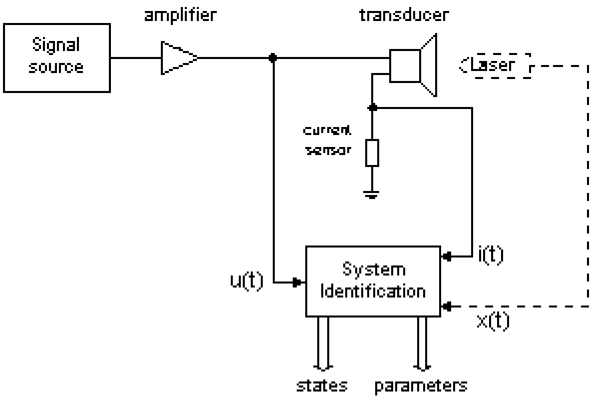
The Introductory Report illustrates the powerful features of the Klippel Analyzer module dedicated to the measurement of the linear speaker parameters. Additional comments are added to the results of a practical measurement applied to the speaker specified above.

After presenting short information to the measurement technique the report comprises the following results

- linear speaker parameters + mechanical creep factor
- electrical impedance response
- mechanical transfer response (voltage to voice coil displacement)
- acoustical transfer response (voltage to SPL)
- time signals of the speaker variables during measurement
- spectra of the speaker variables (fundamental, distortion, noise floor)
- summary on the signal statistics (peak value, SNR, headroom,…).

MEASUREMENT TECHNIQUE

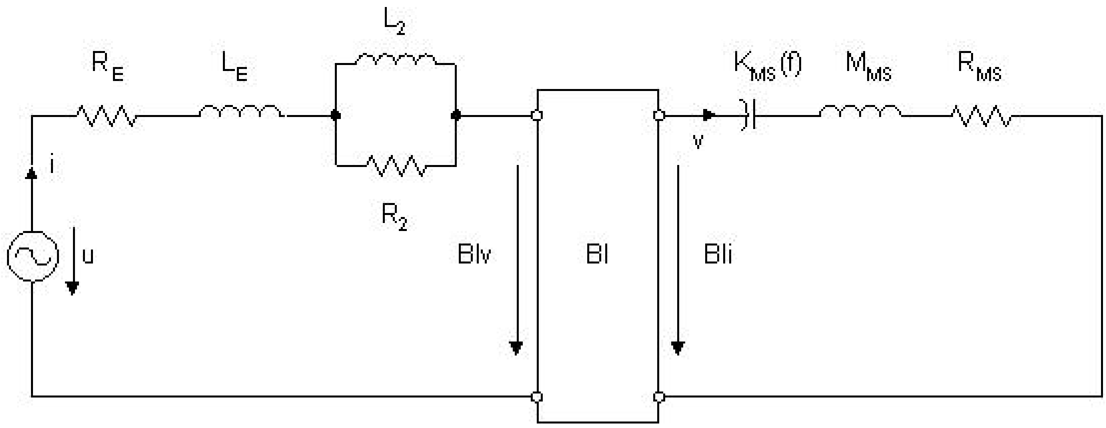
The measurement module identifies the electrical and mechanical parameters (Thiele-Small parameters) of electro-dynamical transducers. The electrical parameters are determined by measuring terminal voltage u(t) and current i(t) and exploiting the electrical impedance Z(f)=U(f)/I(f). The mechanical parameters can either be identified using a laser displacement sensor or by a second (comparative) measurement where the driver is placed in a test enclosure or an additional mass is attached to it. For the first method the displacement of the driver diaphragm is measured in order to exploit the function Hx(f)= X(f)/U(f). So the identification dispenses with a second measurement and avoids problems due to leakage of the test enclosure or mass attachment. Furthermore the suspension creep of the driver is identified giving more accuracy of the loudspeaker model at low frequencies.



Measurement Results

Linear electrical and mechanical parameters

The measurement module determines the components (Thiele-Small Parameters) of the linear loudspeaker model below describing the small signal behaviour of the driver.



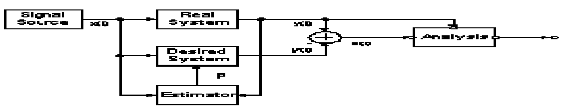
The table below shows the electrical and mechanical parameters of the linear driver model, the derived parameters (resonance frequency, loss factors etc.) and the parameter of the suspension creep factor.

Name	**Value**	**Unit**	**Comment**
Electrical Parameters			
Re	3.46	Ohm	electrical voice coil resistance at DC
Krm	0.0004	Ohm	WRIGHT inductance model
Erm	0.88		WRIGHT inductance model
Kxm	0.0037	Ohm	WRIGHT inductance model
Exm	0.73		WRIGHT inductance model
Cmes	592.96	µF	electrical capacitance representing moving mass
Lces	9.20	mH	electrical inductance representing driver compliance
Res	16.49	Ohm	resistance due to mechanical losses
fs	68.1	Hz	driver resonance frequency
Mechanical Parameters			
(using laser)			
Mms	11.308	g	mechanical mass of driver diaphragm assembly including air load and voice coil
Mmd (Sd)	10.482	g	mechanical mass of voice coil and diaphragm without air load
Rms	1.156	kg/s	mechanical resistance of total-driver losses
Cms	0.482	mm/N	mechanical compliance of driver suspension
Kms	2.07	N/mm	mechanical stiffness of driver suspension
Bl	4.367	N/A	force factor (Bl product)
Lambda s	0.067		suspension creep factor
Loss factors			
Qtp	0.742		total Q-factor considering all losses
Qms	4.187		mechanical Q-factor of driver in free air considering Rms only
Qes	0.879		electrical Q-factor of driver in free air considering Re only
Qts	0.726		total Q-factor considering Re and Rms only
Other Parameters			

Vas	11.3030	l	equivalent air volume of suspension
n0	0.391	%	reference efficiency (2 pi-radiation using Re)
Lm	88.13	dB	characteristic sound pressure level (SPL at 1m for 1W @ Re)
Lnom	88.75	dB	nominal sensitivity (SPL at 1m for 1W @ Zn)
rmse Z	2.26	%	root-mean-square fitting error of driver impedance Z(f)
rmse Hx	1.90	%	root-mean-square fitting error of transfer function Hx (f)
Series resistor	0.00	Ohm	resistance of series resistor
Sd	128.68	cm²	diaphragm area

Suspension creep factor

Some loudspeaker suspension materials exhibit significant creep (continued slow displacement under sustained force) in their dynamic behaviour. Therefore the traditional low-frequency loudspeaker model is expanded to incorporate suspension creep by replacing the simple linear compliance by the dynamic transfer function [1].

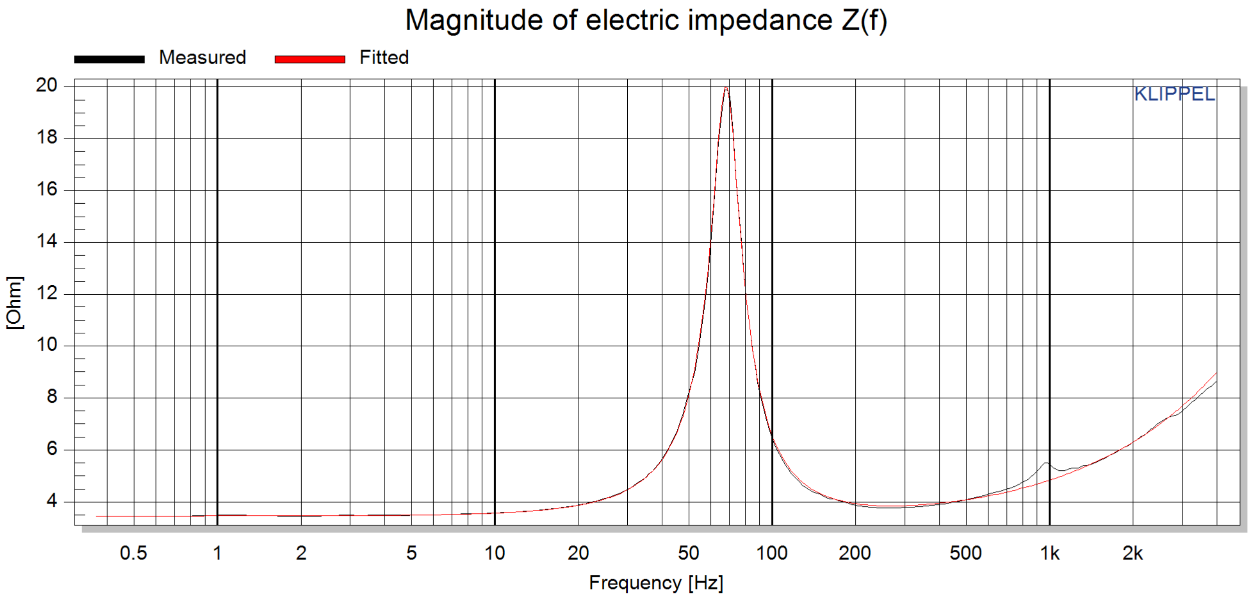


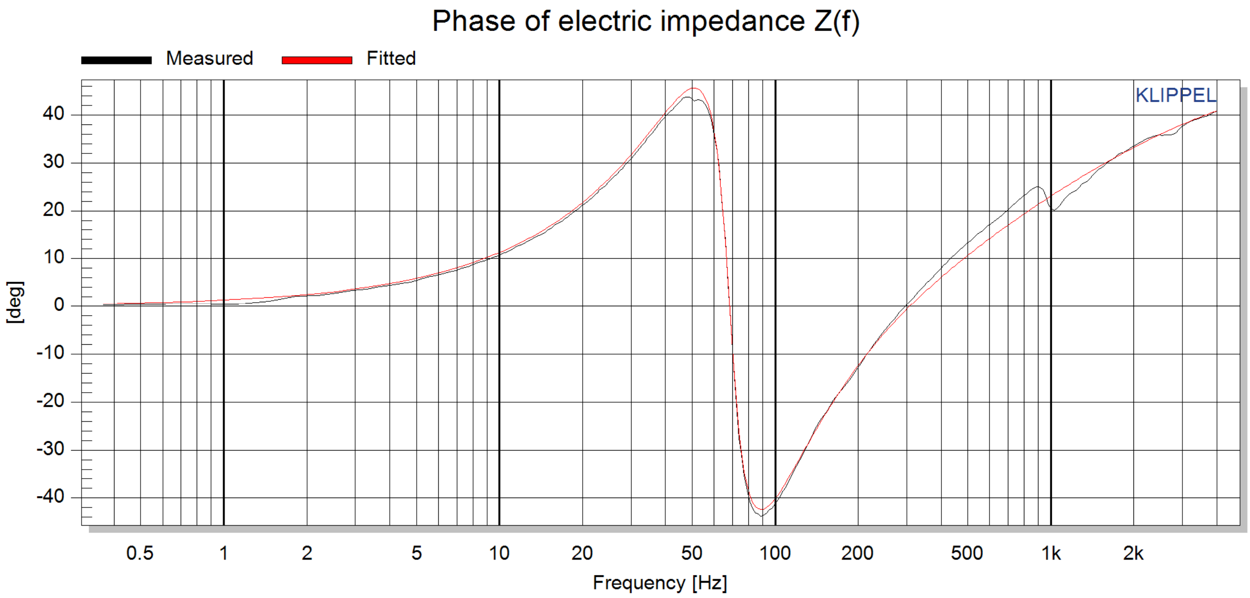
where C_{MS} is the linear compliance and f_s is the driver resonance frequency. There is a straight forward interpretation of the creep factor λ. The quantity λ 100% indicates the decrease of the compliance $C_{MS}(f_s)$ in percentages at low frequencies. For a frequency one decade below the resonance frequency f_s the compliance $C_{MS}(f_s)$ is decreased by λ 100% .

[1] Knudsen, M. H. and Jensen, J. G. *Low-frequency loudspeaker models that include suspension creep.* J. Audio Eng. Soc., Vol. 41, No. 1 / 2, 1993

Electrical Impedance

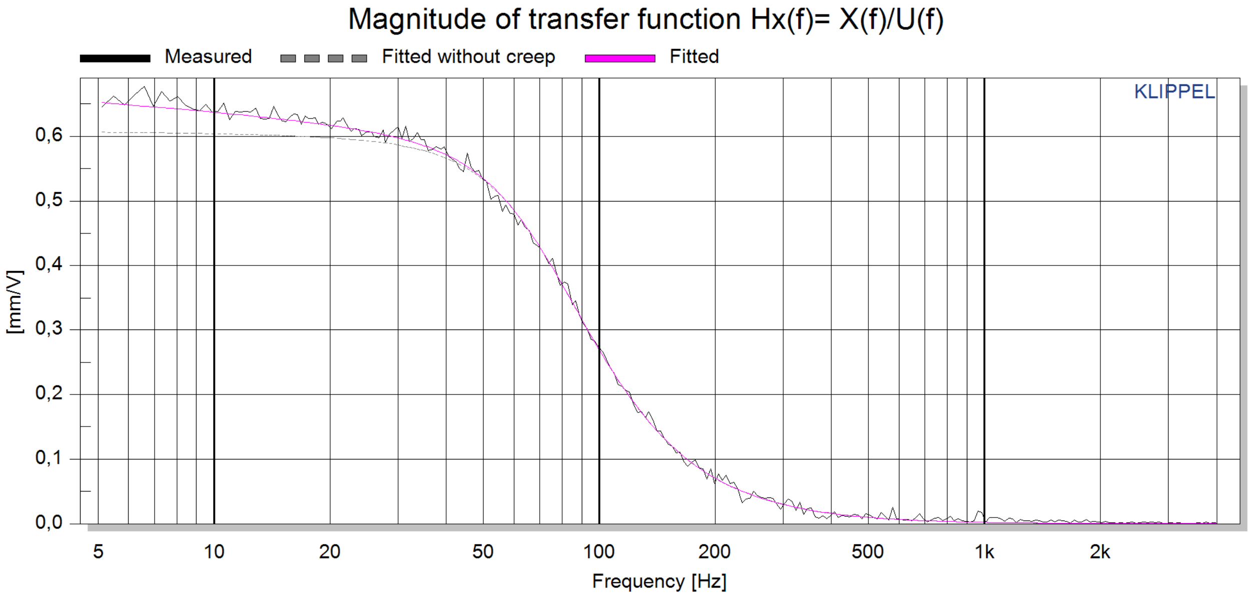
The two figures below show the magnitude and the phase response of the measured and estimated transfer function Z(f)= U(f)/I(f) where U is the terminal voltage and I is the current. The **solid** curve is the ratio of the measured spectra U(f), I(f) while the *thin* curve is the impedance of the linear driver equivalent circuit using the linear model and the identified electrical parameters shown





Displacement Transfer Function

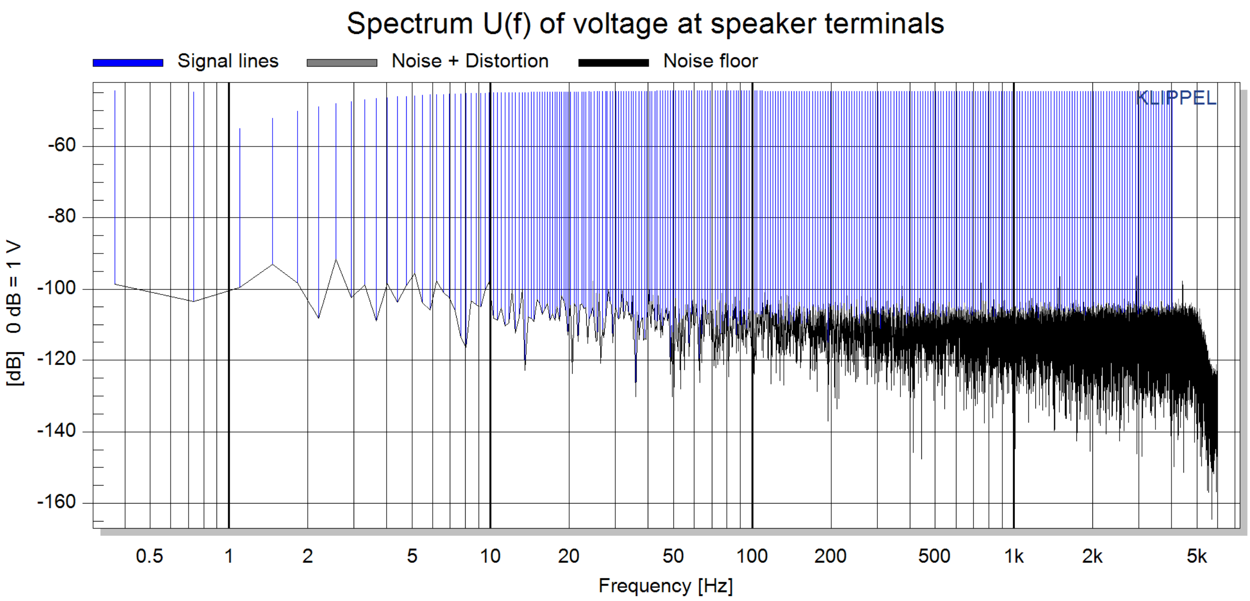
The figure below shows the magnitude of the measured and estimated transfer function Hx(f)= X(f)/U(f) between the voice coil displacement X and the terminal voltage U. The **solid black curve** is the ratio of the measured spectra X(f), U(f) while the ***thin black curve*** is the transfer function based on the linear driver equivalent circuit using the identified electrical and mechanical parameters as well as the creep parameter. The dashed red curve is based on the conventional model without considering the creep factor.



Spectra of measurement signals

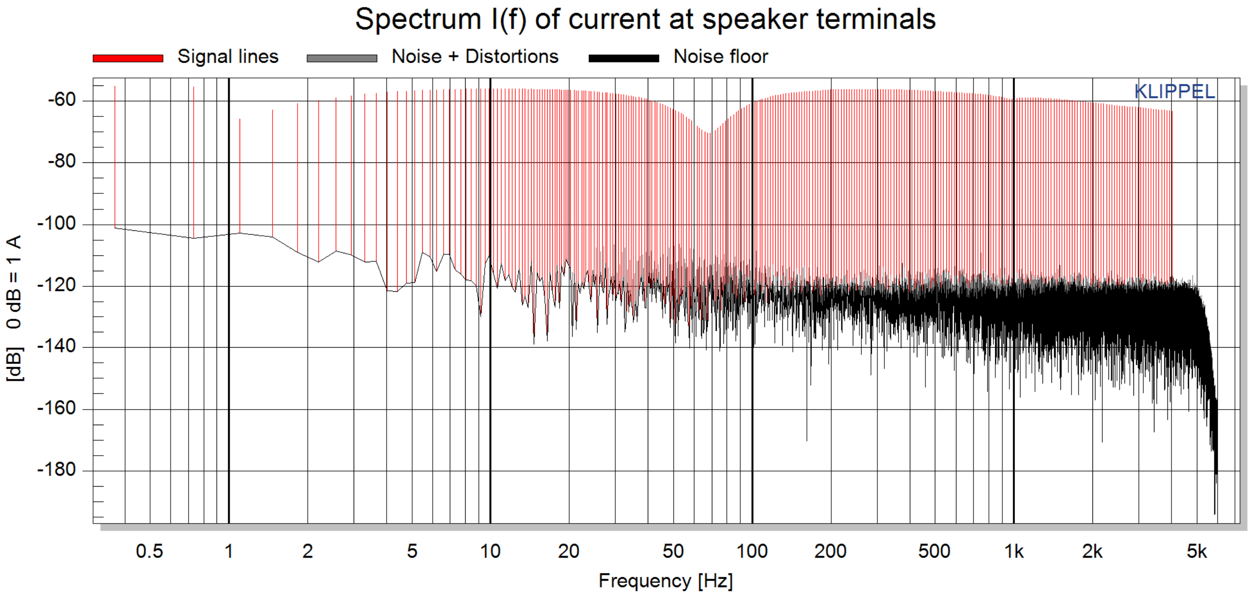
Voltage Spectrum

The diagram shows the multi-tone spectrum of the voltage at the terminals. The blue lines represent the fundamental components excited by the stimulus. The **black "noise floor"** lines represent the residual measurement noise caused by the voltage sensor. If the grey **"noise + distortions"** exceeds the residual noise floor we see the distortions generated by the nonlinearities of the power amplifier. This information is important for assessing the distortion of the speaker in the current, displacement and sound pressure below.



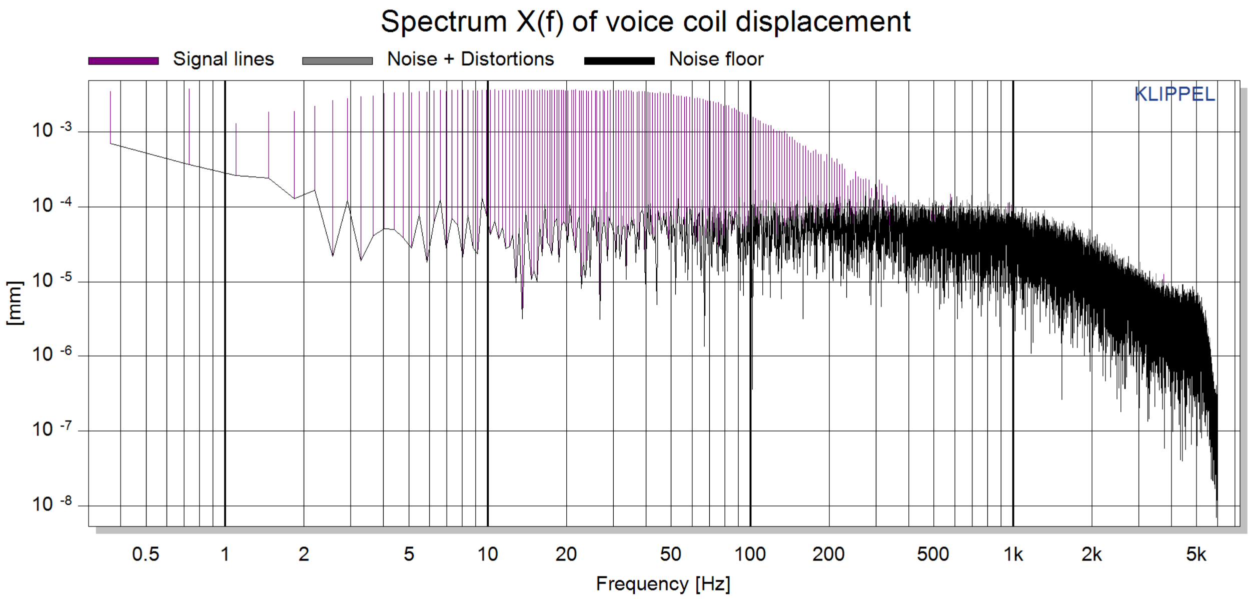
Current Spectrum

The diagram below shows the multi-tone spectrum of the current at the terminals. The red lines represent the fundamental components excited by the stimulus. Note the notch of the spectrum at the resonance frequency of the driver. The black **"noise floor"** lines indicate the residual noise caused by the measurement system (current sensor). If the grey **"noise + distortions" lines** exceeds the residual noise floor we see the distortions generated by the nonlinearities of the speaker (assuming that the power amplifier is sufficiently linear).



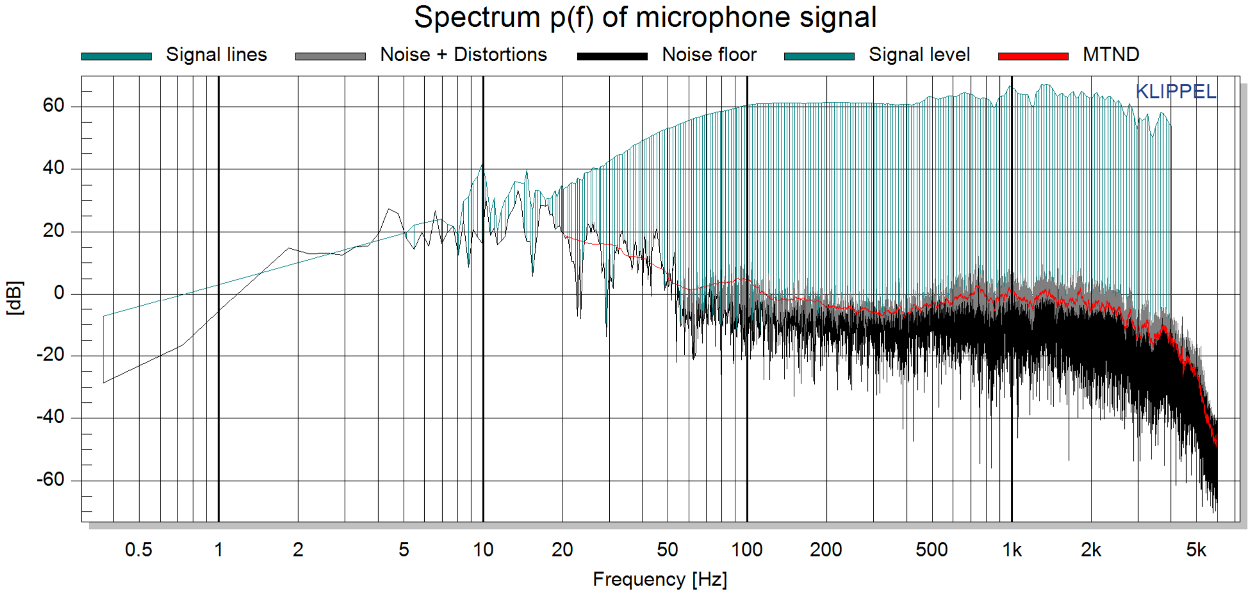
Displacement Spectrum

The diagram below shows the multi-tone spectrum of the voice coil displacement measured with the laser sensor. The **violet lines** represent the fundamental components excited by the stimulus. Note the 12 dB/octave decay of the displacement spectra above the resonance frequency of the laser. The black **"noise floor"** lines indicate the measurement noise caused by the resolution of the used Laser Sensor Head. Increasing the number of averaging will further reduce the residual noise line. If the grey **"noise + distortions"** exceeds the residual noise floor we see the distortions generated by the nonlinearities of the speaker. These components are independent on the number of averaging.



Sound Pressure Spectrum

The diagram shows the multi-tone spectrum of the sound pressure measured with the microphone. The **green lines** represent the fundamental components excited by the stimulus.The black **"noise floor"** lines indicate the ambient noise during the measurement. The grey **"noise + distortions"** are the nonlinear distortion components generated by the speaker.



Signal Characteristics

The table below summarizes important statistical characteristics (peak values, head rooms, SNR ratio, …) of the state variables (voltage, current, displacement and sound pressure). This information is helpful for assessing the working point of the driver (Small - Large Signal Domain) and to detect any malfunction operation (microphone or laser not connected).

Name	Value	Unit	Comment
Reduce Fmax to 20* fs to improve impedance fitting			
Voltage			
U pp	0.76	V	peak to peak value of voltage at terminals
U ac	0.10	V rms	AC part of voltage signal
U dc	-0.00	V	No proper amplifier output, is amplifier switched off?
U head	57.1	dB	digital headroom of voltage signal
U SNR+D	41.0	dB	ratio of signal to noise+distortion in voltage signal
fu noise	1.5	Hz	frequency of noise+distortion maximum in voltage signal
Current			
I pp	0.15	A	peak to peak value of current at terminals
I ac	0.02	A rms	AC part of current signal
I dc	0.00	A	
I head	37.2	dB	digital headroom of current signal
I SNR+D	37.1	dB	ratio of signal to noise+distortion in current signal
fi noise	1.1	Hz	frequency of noise+distortion maximum in current signal
Displacement			
X pp	0.25	mm	peak to peak value of displacement signal

X ac	0.04	mm rms	AC part of displacement signal
X dc	-0.04	mm	
X head	56.6	dB	digital headroom of displacement signal
X SNR+D	26.9	dB	ratio of signal to noise+distortion in displacement signal
fx cutoff	199.2	Hz	frequency of highest valid line in displacement signal
SPL			
p pp	152.12	mV	peak to peak value of microphone signal
p ac	18.62	mV rms	AC part of microphone signal
p head	34.1	dB	digital headroom of microphone signal
p sum level	84.8	dB	sum level of microphone signal
p mean level	58.3	dB	mean level of microphone signal
Measurement			
f sample	12000	Hz	sample frequency
N stim	32768	samples	stimulus length
cal x laser	-0.115538		Laser calibration factor

Number	Name	Date	Time	Comment
2	1b LPM T/S para. (added mass)	03/31/09	16:23:30	Two step method using added mass. Target: Check accuracy of laser measurement with identical setup. Adjust setup: export setup from laser LPM, import setup in PP Im/Export, insert weight of added mass in PP Method

Linear Parameter Measurement (LPM)

Added mass method

An additional added mass measurement can be used to verify the laser measurement based results.
This can be helpful to check the laser calibration or check the optical properties of the diaphragm.
Results can be compared in the dedicated database.

Number	Name	Date	Time	Comment
3	2 LSI Nonlinear Parameters	03/31/09	17:13:35	Target: Measure Nonlinear Parameter Adjust Setup: - select the start gain "G small" in relation to the speaker and the gain of the amplifier (PP Protection) - Tlim is also used as target temperature during temperature mode

Large Signal Identification (LSI)

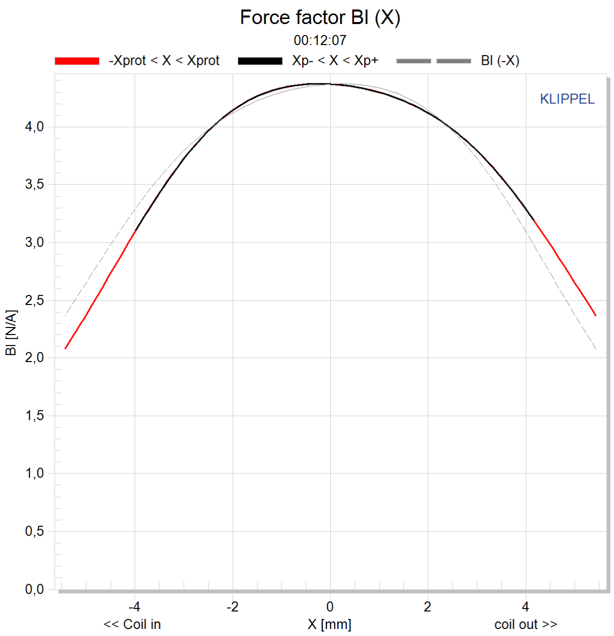
Overview

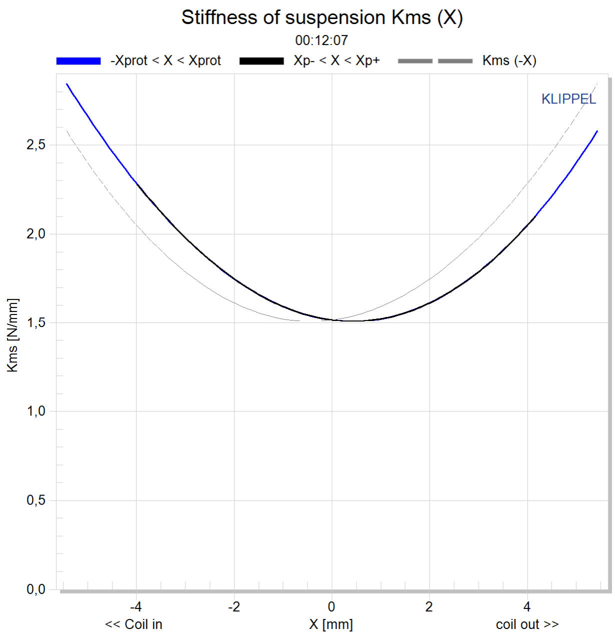
The Introductory Report illustrates the powerful features of the Large Signal Identification Module (LSI):
< nonlinear speaker parameters versus displacement and current
< coefficients of the power series expansion of the nonlinear parameters
< derived speaker parameters such as resonance frequency and loss factors
< parameters at the rest position (parameters for linear modeling)
< parameter variation versus time
< state variables of the speaker (temperature, displacement, …)
< contribution of each nonlinearity to the total distortion (distortion analysis)
< suggestions for loudspeaker improvements (remedy parameters)

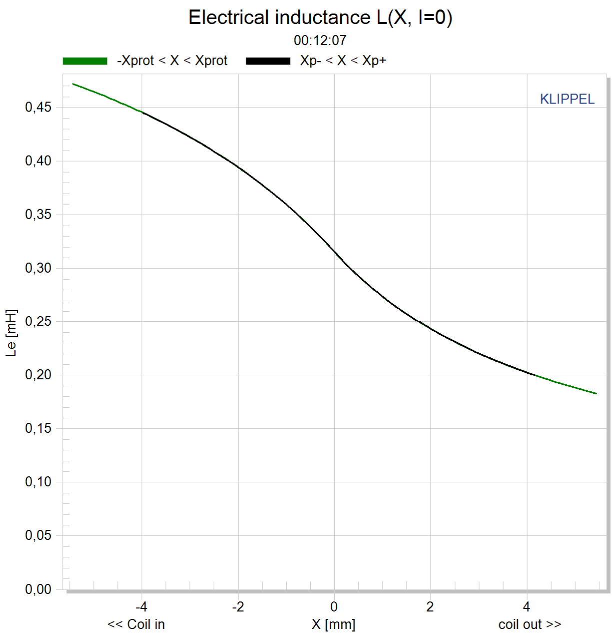
Nonlinear Characteristics

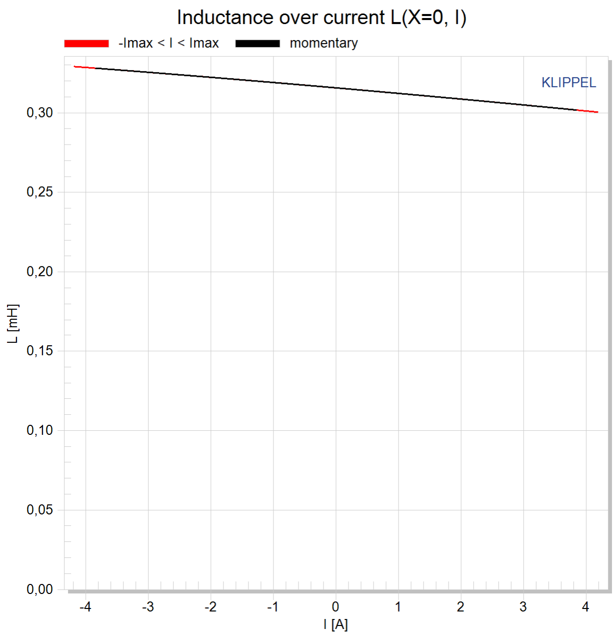
The dominant nonlinearities are modelled by variable parameters such as

Bl(x)	instantaneous electro-dynamic coupling factor (force factor of the motor) defined by the integral of the magnetic flux density B over voice coil length l as a function of displacement
$K_{MS}(x)$	mechanical stiffness of driver suspension a function of displacement
$L_E(i)$	voice coil inductance as a function of input current (describes nonlinear permeability of the iron path)
$L_E(x)$	voice coil inductance as a function of displacement









A solid line represents the used working range $x_{-peak} < x < x_{+peak}$ between the minimal and maximal peak displacement occurred in last update interval of the measurement. The dotted line shows the allowed working range $x_{max} < x < x_{max}$ identified by the automatic gain adjustment by using predefined limit values.

Nonlinear and thermal Parameters

Alternatively the nonlinear parameters may be represented by the coefficients of the power series expansion. This representation uses a minimal set of parameters and simplifies the export of the nonlinear parameters to the numerical simulation module (SIM) or any other design tool. The displacement X_{pse} describes the working range $-X_{pse} < X < X_{pse}$.

The displacement limits X_{BL}, X_C, X_L and X_d describe the limiting effect for the force factor Bl(x), compliance $C_{ms}(x)$, inductance $L_e(x)$ and Doppler effect, respectively, according to the threshold values Bl_{min}, C_{min}, Z_{max} and d_2 used. The thresholds Bl_{min}= 82 %, C_{min}=75 %, Z_{max}=10 % and d_2=10% generate for a two-tone-signal ($f_1=f_s$, $f_2=8.5f_s$) 10 % total harmonic distortion or 10 % intermodulation distortion

Symbol	Number	Unit	Comment
Displacement Limits			thresholds can be changed in Processing property page
X Bl @ Bl min=82%	3.2	mm	Displacement limit due to force factor variation
X C @ C min=75%	3.2	mm	Displacement limit due to compliance variation
X L @ Z max=10 %	1.9	mm	Displacement limit due to inductance variation
X d @ d2=10%	15.2	mm	Displacement limit due to IM distortion (Doppler)
Thermal Parameters			
alpha			Heating of voice coil by eddy currents
alphaOrg			Heating of voice coil by eddy currents (without limits)
Rtv		K/W	thermal resistance coil ==> pole tips
rv		Ws/Km	air convection cooling depending on velocity
Rtm		K/W	thermal resistance magnet ==> environment
tau m		min	thermal time constamt of magnet
Ctm		Ws/K	thermal capacity of the magnet
tau v		s	thermal time constant of voice coil
Ctv		Ws/K	thermal capacity of the voice coil
Thermal State			
delta Tw		K	Temperature increase in Warm Resistance Mode
delta Tc		K	Temperature increase in Convection Mode
delta Te		K	Temperature increase in Eddy Mode
Pcoil(warm)		W	Pcoil in warm mode
Pcoil(conv)		W	Pcoil in convection mode
Ptv(mag.beg)		W	power heating the coil at beginning of magnet mode
Ptv(mag.mid)		W	power heating the coil sampled in the middle of magnet mode
Ptv(mag.end)		W	power heating the coil at end of magnet mode

Ptm(mag.beg)		W	power heating the magnet at beginning of magnet mode
Ptm(mag.mid)		W	power heating the magnet sampled in the middle of magnet mode
Ptm(mag.end)		W	power heating the magnet at end of magnet mode
Power Series			
Bl0 = Bl (X=0)	4.4026	N/A	constant part in force factor
Bl1	0.0012134	N/Amm	1st order coefficient in force factor expansion
Bl2	-0.074105	N/Amm^2	2nd order coefficient in force factor expansion
Bl3	0.0010370	N/Amm^3	3rd order coefficient in force factor expansion
Bl4	-2.4617e-005	N/Amm^4	4th order coefficient in force factor expansion
Bl5		N/Amm^5	5th order coefficient in force factor expansion
Bl6		N/Amm^6	6th order coefficient in force factor expansion
Bl7		N/Amm^7	7th order coefficient in force factor expansion
Bl8		N/Amm^8	8th order coefficient in force factor expansion
L0 = Le (X=0)	0.31670	mH	constant part in inductance
L1	-0.038392	mH/mm	1st order coefficient in inductance expansion
L2	0.00035884	mH/mm^2	2nd order coefficient in inductance expansion
L3	0.00045082	mH/mm^3	3rd order coefficient in inductance expansion
L4	6.3053e-006	mH/mm^4	4th order coefficient in inductance expansion
L5		mH/mm^5	5th order coefficient in inductance expansion
L6		mH/mm^6	6th order coefficient in inductance expansion
L7		mH/mm^7	7th order coefficient in inductance expansion
L8		mH/mm^8	8th order coefficient in inductance expansion
C0 = Cms (X=0)	0.65619	mm/N	constant part in compliance
C1	0.012077	1/N	1st order coefficient in compliance expansion
C2	-0.015211	1/Nmm	2nd order coefficient in compliance expansion
C3	-0.00031915	1/Nmm^2	3rd order coefficient in compliance expansion
C4	0.00018816	1/Nmm^3	4th order coefficient in compliance expansion
C5		1/Nmm^4	5th order coefficient in compliance expansion
C6		1/Nmm^5	6th order coefficient in compliance expansion
C7		1/Nmm^6	7th order coefficient in compliance expansion
C8		1/Nmm^7	8th order coefficient in compliance expansion
K0 = Kms (X=0)		N/mm	constant part in stiffness
K1	-0.035289	N/mm^2	1st order coefficient in stiffness expansion
K2	0.040861	N/mm^3	2nd order coefficient in stiffness expansion
K3	0.00036726	N/mm^4	3rd order coefficient in stiffness expansion
K4	-1.0769e-005	N/mm^5	4th order coefficient in stiffness expansion
f1	-0.010834	1/A	coefficient (1) of L(I) Inductance over current (flux modulation)
f2	-0.000168	1/A^2	coefficient (2) of L(I) Inductance over current (flux modulation)

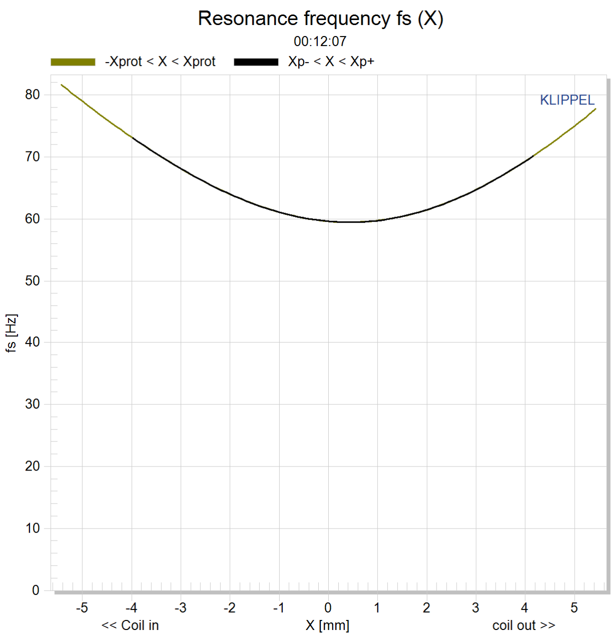
Xpse	5.4	mm	-Xpse < X < Xpse, range where power series is fitted

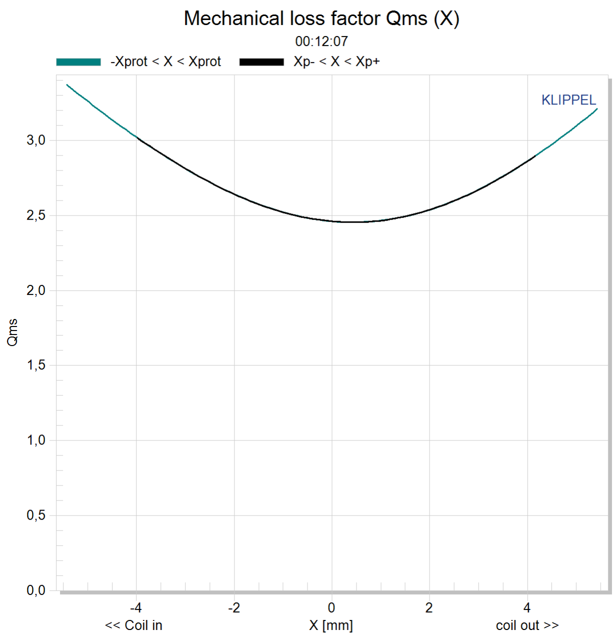
Derived Loudspeaker Parameters

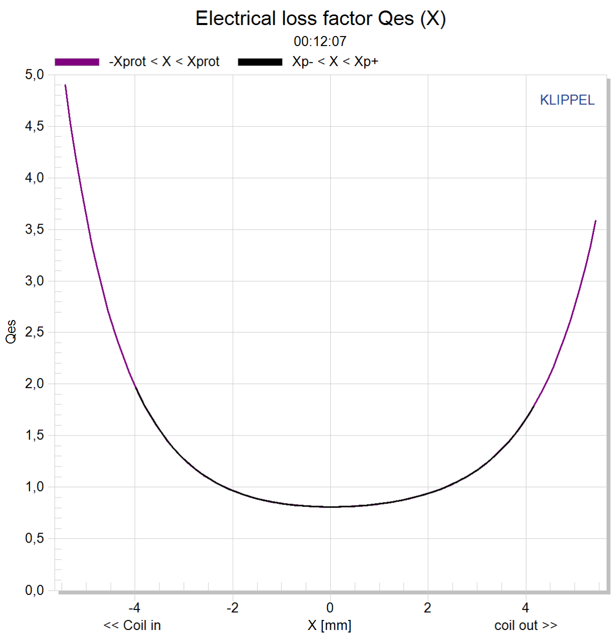
For the analysis and synthesis of loudspeaker system it is convenient to use special transducer parameters:

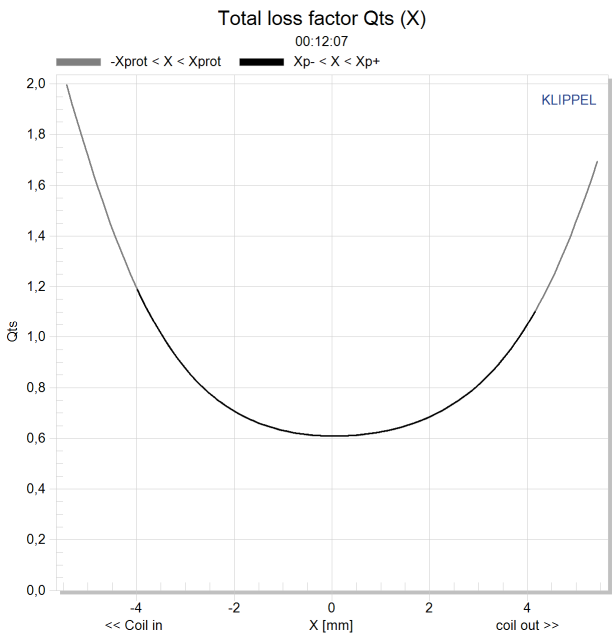
f_s (x)	instantaneous resonance frequency of the transducer varying with voice coil displacement
Q_{MS}(x)	mechanical loss factor of the transducer at f_s considering driver non-electrical resistances only
$Q_{ES}(T_V, x)$	electrical loss factor by considering the electrical resistance $R_E(T_V)$ only,
$Q_T(T_V, x)$	total loss factor at f_s and voice coil temperature T_V considering mechanical and electrical resistances R_{MS} and $R_E(T_V)$ only.

In contrast to linear modelling most of these parameters are not constant but depend on the instantaneous state of the transducer (displacement x, the voice coil temperature T_V).









Parameters at the Rest Position

The value of the nonlinear parameters at the rest position (*x=0*) may be used as input for the traditional linear modelling and may be referred as “linear parameters”. Please note that these parameters depend on the instantaneous state of the driver (voice coil temperature, peak value of displacement) and are presented for three different modes of operation:

Mode	Properties
LARGE+WARM	the transducer is operated in the large signal domain, the peak value of the displacement is high ($\|x\| < x_{max}$), the variation of the parameters is not negligible, the voice coil temperature is increased ($\Delta T_V > 0$) due to heating.
LARGE+COLD	the transducer is operated in the large signal domain, the peak value of the displacement is high ($\|x\| < x_{max}$), the variation of the parameters is not negligible, the effect of heating is compensated while considering the cold voice coil resistance $R_e(\Delta T_V = 0)$.
SMALL SIGNAL	the transducer is operated in the small signal domain, the amplitude of the excitation signal is sufficiently small, the displacement is small in comparison to the allowed maximal displacement ($\|x\| << x_{max}$), the variations of the nonlinear parameters are negligible, the increase of voice coil temperature is negligible ($\Delta T_V \approx 0$), the effects of the nonlinear, thermal and time-varying mechanisms are negligible, the transducer behaves almost linear.

Symbol	Large + Warm	Large + Cold	Small Signal	Unit	Comment
Note:					for accurate small signal parameters, use LPM module
Delta Tv = Tv-Ta	30	0	0	K	increase of voice coil temperature during the measurement
Xprot	5.4	5.4	0.7	mm	maximal voice coil excursion (limited by protection system)
Re (Tv)	3.81	3.42	3.42	Ohm	voice coil resistance considering increase of voice coil temperature Tv
Le (X=0)	0.32	0.32	0.27	mH	voice coil inductance at the rest position of the voice coil
L2 (X=0)	0.38	0.38	0.36	mH	para-inductance at the rest position due to the effect of eddy current
R2 (X=0)	1.48	1.48	1.26	Ohm	resistance at the rest position due to eddy currents
Cmes (X=0)	631	631	625	µF	electrical capacitance representing moving mass
Lces (X=0)	11.29	11.29	9.49	mH	electrical inductance at the rest position representing driver compliance
Res (X=0)	10.41	10.41	9.34	Ohm	resistance at the rest position due to mechanical losses
Qeps (X=0,					electrical Q-factor considering Zl (fs, Tv)

Tv)	0.81	0.73	0.88		only
Qtp (X=0, Tv)	0.61	0.56	0.64		total Q-factor considering all losses
Qms (X=0, Tv)	2.46	2.46	2.40		mechanical Q-factor considering Rms only
Qes (Tv)	0.81	0.73	0.88		electrical Q-factor considering Re (Tv) only
Qts (X=0, Tv)	0.61	0.56	0.64		total Q-factor considering Re (Tv) and Rms only
fs	59.6	59.6	65.4	Hz	driver resonance frequency
Mms	10.809	10.809		g	mechanical mass of driver diaphragm assembly including voice-coil and air load
Rms (X=0)	1.645	1.645	2.045	kg/s	mechanical resistance of total-driver losses
Cms (X=0)	0.66	0.66	0.50	mm/N	mechanical compliance of driver suspension at the rest position
Kms (X=0)	1.52	1.52	2.01	N/mm	mechanical stiffness of driver suspension at the rest position
Bl (X=0)	4.37	4.37	4.37	N/A	(imported) force factor at the rest position (Bl product)
Vas	15.3895	15.3895	11.5977	l	equivalent air volume of suspension
N0	0.386	0.431	0.353	%	reference efficiency (2Pi-sr radiation using Re)
Lm	88.0	88.5	87.6	dB	characteristic sound pressure level
Sd	128.68	128.68	128.68	cm²	diaphragm area

Parameter Variation versus Time

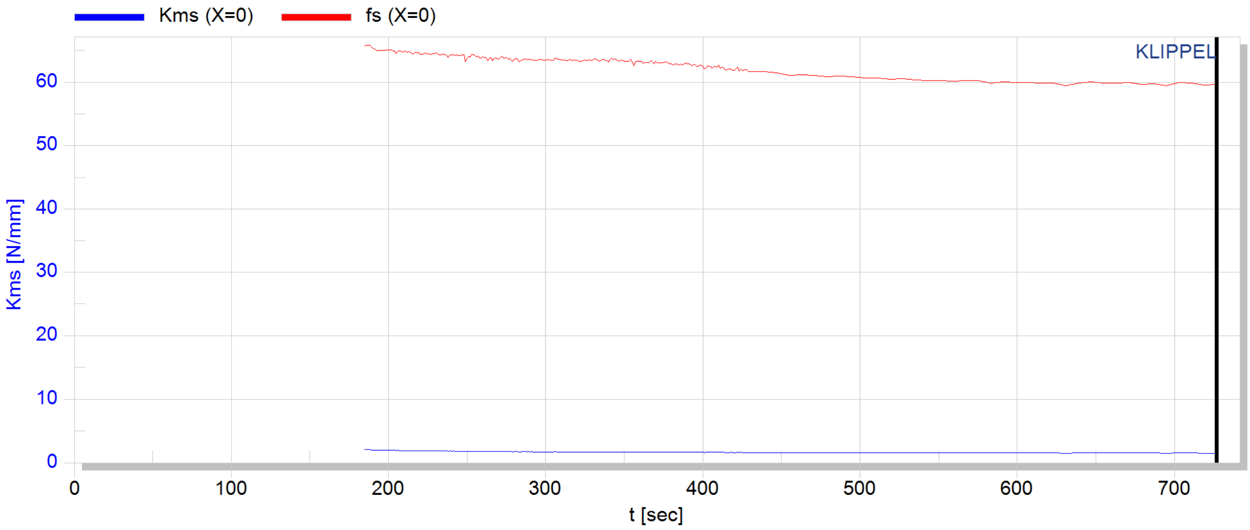
The instantaneous estimates of the speaker parameters are sampled during the measurement and stored in a database. Note the difference between the initial identification which starts in the small signal domain and determines the maximal amplitude limits (x_{max}, P_{max}, …) of the safe range of operation and the final long term measurement where the amplitude of the signal is almost constant.

Temporal Variations of the Stiffness $K_{MS}(t, x=0)$

The properties of the mechanical suspension depend not only on the instantaneous displacement x but also on the maximal peak value of displacement in the last few seconds and vary with time. There are reversible and non-reversible processes due to creep, aging and relaxation.

Stiffness Kms (t) and resonance frequency fs (t) at rest position X=0

00:12:07



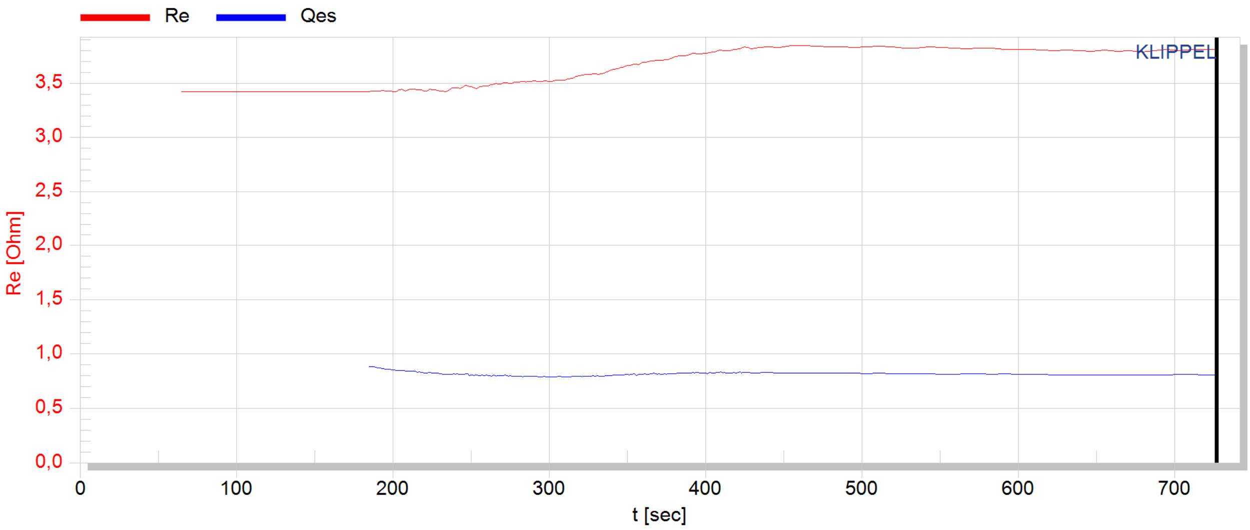
Temporal Variations of the Voice Coil Resistance $R_E(t)$

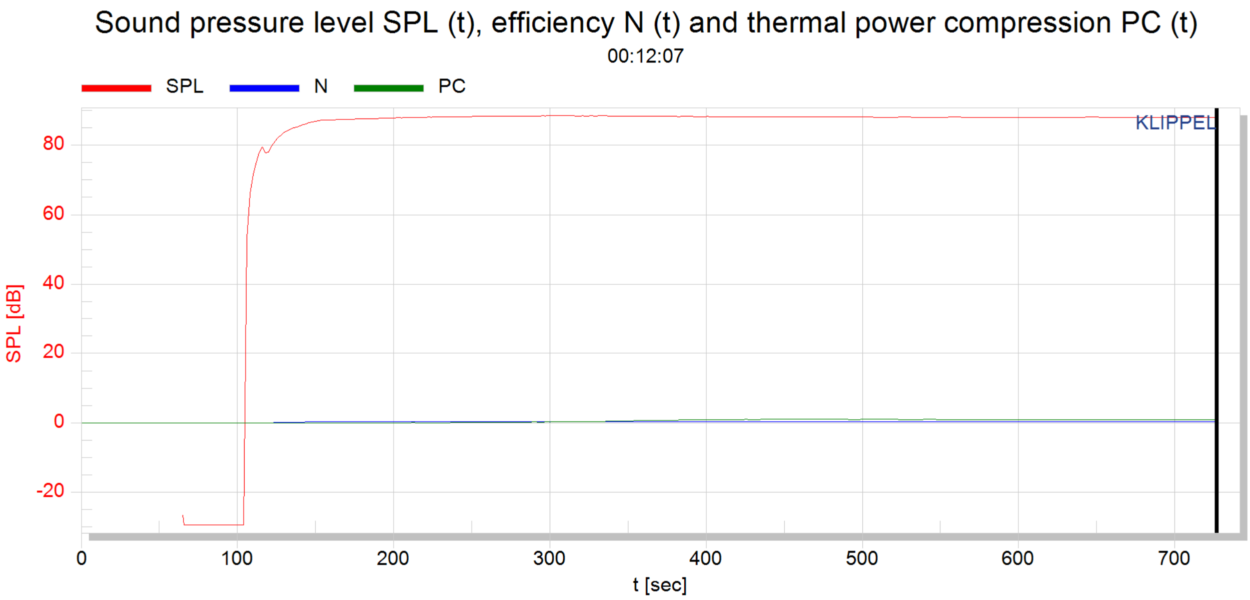
The voice coil resistance $R_E(t)$ varies during the measurement due to heating of the voice coil. This variations affect

$Q_{ES}(t, x=0)$	electrical loss factor
$h_0(t)$	efficiency
PC	power compression factor is expressed in dB and describes the loss of efficiency in the pass-band of the transducer where the electric resistance R_E dominates the electric input impedance.
SPL(t)	reference sound pressure level

Electrical resistance Re (t) and electrical loss factor Qes (t)

00:12:07





Transducer State

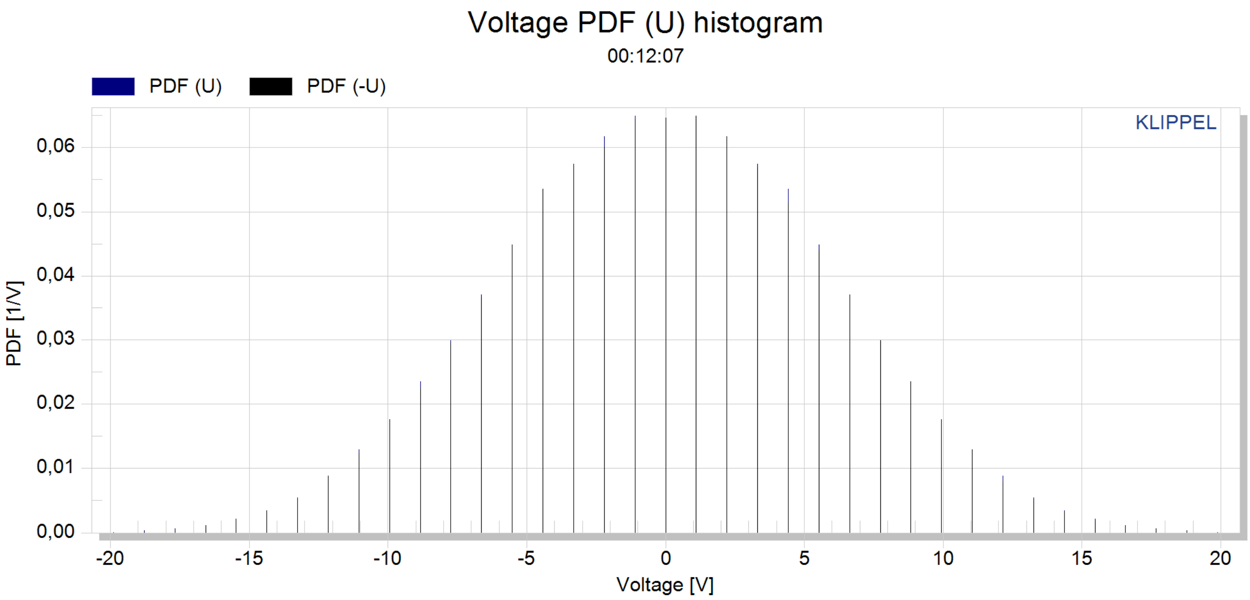
The state information describes the progress of system identification and important transducer variables in the last update interval of the measurement.

Symbol	Value	Unit	Comment
DBG: Data LOD:	172/1		LOD of results
DBG: Module LOD:	184/1		LOD version in current module
------	------	------	------
Date	2009-03-31		
Time	17:01:18		
Serial number	150		
Mode	Nonlinear Mode 5(7)		
Record	429/429		
Laser	signal reliable		
t	00:12:07	h:min:s	measurement time
Time remaining	00:02:53	h:min:s	recalculated at thermal mode(a)
Ei (t)	3.4	%	error current measurement
Ex (t)	1.1	%	error laser measurement
Eu (t)	6.7	%	error amplifier check
Delta Tv (Delta Tlim)	30.3 (90.0)	K	increase of voice coil temperature (limit)
Blmin (Bllim)	49.9 (50.0)	%	minimal force factor ratio (limit)
Cmin (Clim)	54.6 (50.0)	%	minimal compliance ratio (limit)
P (Plim)	6.5275 (50.00)	W	real electrical input power (limit)
Lmin	58.5	%	minimal inductance ratio
Pn	8.732492	W	nominal electrical input power
P Re	5.393398	W	Power heating voice coil
P Mech	1.859188	W	....

Irms	1.189	A	rms value of the electrical input current
Urms	5.910	V	rms value of the electrical voltage at the transducer terminals
Ipeak	3.836	A	peak value of the electrical input current
Upeak	20.921	V	peak value of the electrical voltage at the transducer terminals
PC	0.95	dB	thermal power compression factor
Glarge (Gmax)	18.8 (26.0)	dB	gain of the excitation amplitude increased in the large signal domain (maximum)
Mech. system		abs.	import used to identify mechanical system in absolute quantities
Xdc	0.05	mm	dc component of voice coil excursion measured in the last update intervall
Xpeak	5.26	mm	positive peak value of voice coil excursion measured in the last update intervall
Xbottom	-4.49	mm	negative peak value (bottom) of voice coil excursion measured in the last update intervall
Xp+	4.2	mm	upper limit of displacement range (99% probability)
Xp-	-4.0	mm	lower limit of displacement range (99% probability)
Xprot	5.4	mm	maximal voice coil excursion allowed by protection system
v rms	0.5	m/s	voice coil velocity
Distortion			
Db	18.1	%	distortion factors representing contribution of nonlinear force factor
Dl	5.7	%	distortion factor representing contribution of nonlinear inductance
Dc	20.3	%	distortion factor representing contribution of nonlinear compliance
D l(i)	0.7	%	distortion factor representing contribution of L(I) nonlinearity
Thermal			
R tc (v) (new)		K/W	
R tc (v) (old)		K/W	
R th total	5.61	K/W	Delta Tv / P Re
dbg: passedmode			23. . . . . . .16.15.14.13.12.11. .9.8.7.6.5.4. . . . .

Voltage Probability Density Function pdf(u)

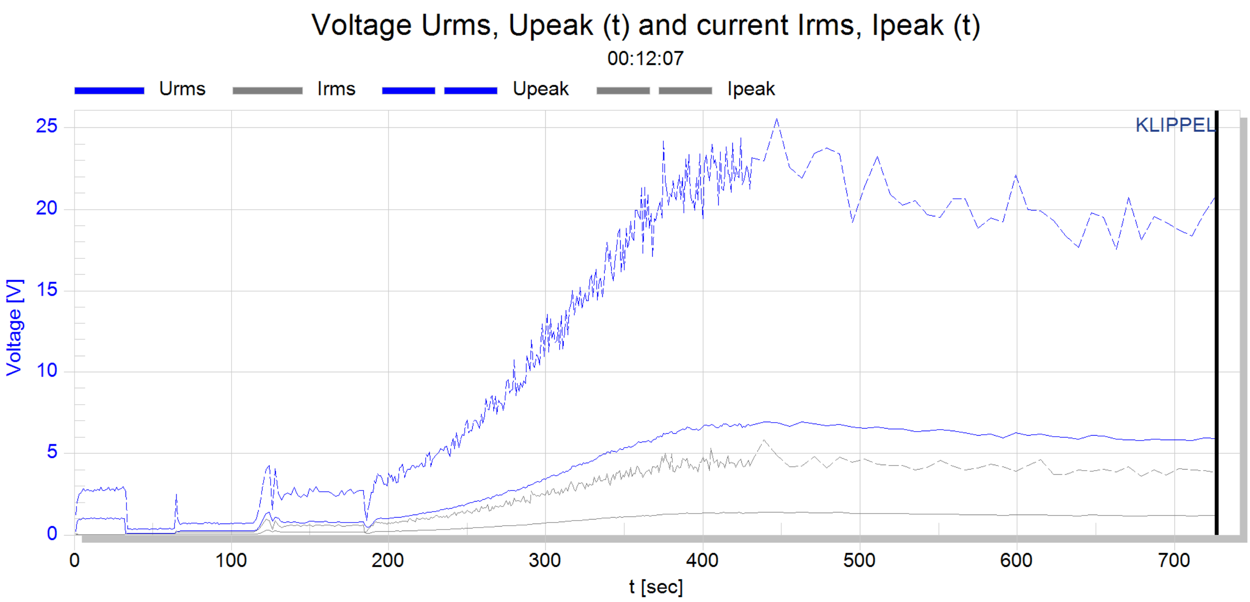
The probability density function of the voltage pdf(u) reflects the properties of the excitation signal (noise) and of the power amplifier used. If the power amplifier is not limiting and does not generate a DC-component in the output signal the pdf(u) is almost perfectly symmetrical. The positive and negative peak values, the rms-value and the crest-factor of the signal can be derived from the properties of the pdf(u).



Voltage $u_{peak}(t)$ and Current $i_{peak}(t)$

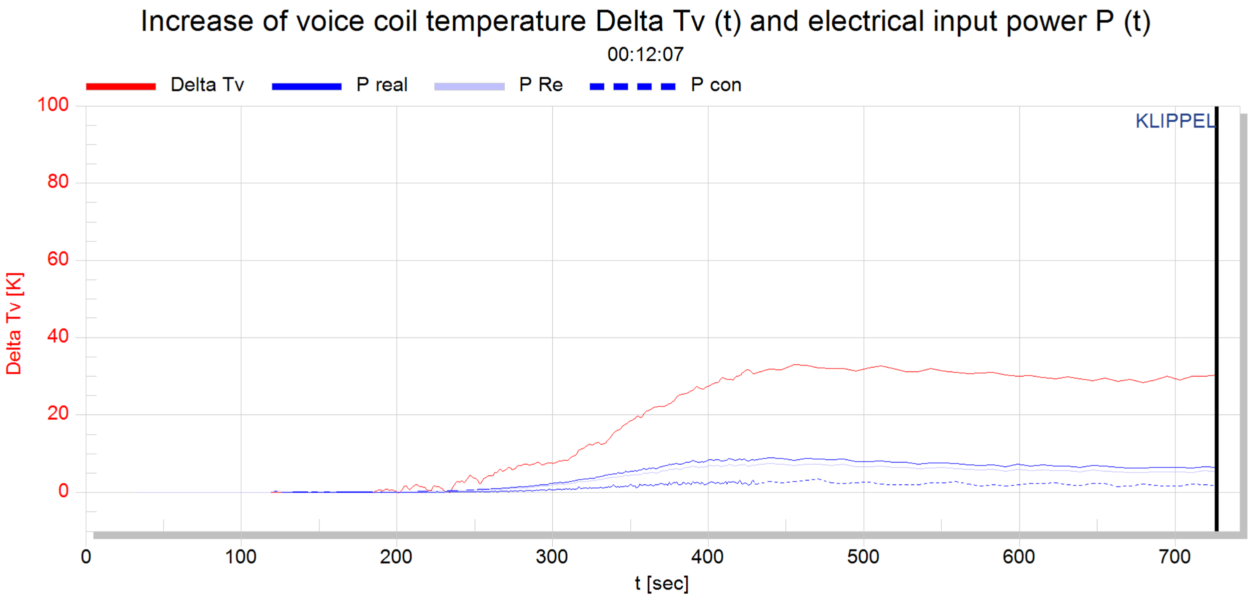
The electric signals at the transducer terminals are represented by

i_{peak}	peak value of the electric input current,
u_{peak}	peak value of the electric voltage at the transducer terminals.



Voice Coil Temperature D $T_V(t)$ and Power $P(t)$

The increase of the voice coil temperature $\Delta T_V(t)$ in comparison to the electric input power P(t) versus measurement time t shows the thermal characteristic of the transducer.



The different modes of operation can easily be identified in the time plot.

In the Amplifier Mode 1(7) the loudspeaker is disconnected and the gain, polarity and distortion of the power amplifier is checked. Here the amplitude of the current is zero.

In the Resistance Mode 2(7) the dc-resistance of the voice coil is measured by a pulsed noise signal.

In the Linear Mode 3(7) the loudspeaker is connected and noise at low amplitude is used as stimulus. The transducer is operated in the small-signal domain. The temperature of the voice coil at the end of this phase is used as reference temperature T_A which equals the ambient temperature.

In the Fast Mode 4(7) the amplitude of the stimulus is increased and the limits of the allowed working range are detected automatically. The voice coil temperature T_V increases with the input power. Both state signals are used as protection variables and are compared with the limit values P_{lim} and T_{lim} defined by the user.

In the Nonlinear Mode 5(7) the learning speed is reduced and the nonlinear curves are measured at highest precision.

The Thermal Mode 6(7) consist of special heating and cooling phases to identify the thermal parameters

a) a) heating up to ?T_{lim} by using a band-pass filtered signal 400 Hz – 1 kHz

b) b) cooling down phase to measure coil capacity

c) c) heating up to ?T_{lim} by using a band-pass filtered signal 10 Hz – 1kHz to measure convection cooling

d) d) cooling down phase

e) e) heating up to ?T_{lim} by using a band-pass filtered signal 400Hz – 2.5 kHz to measure heating by eddy currents.

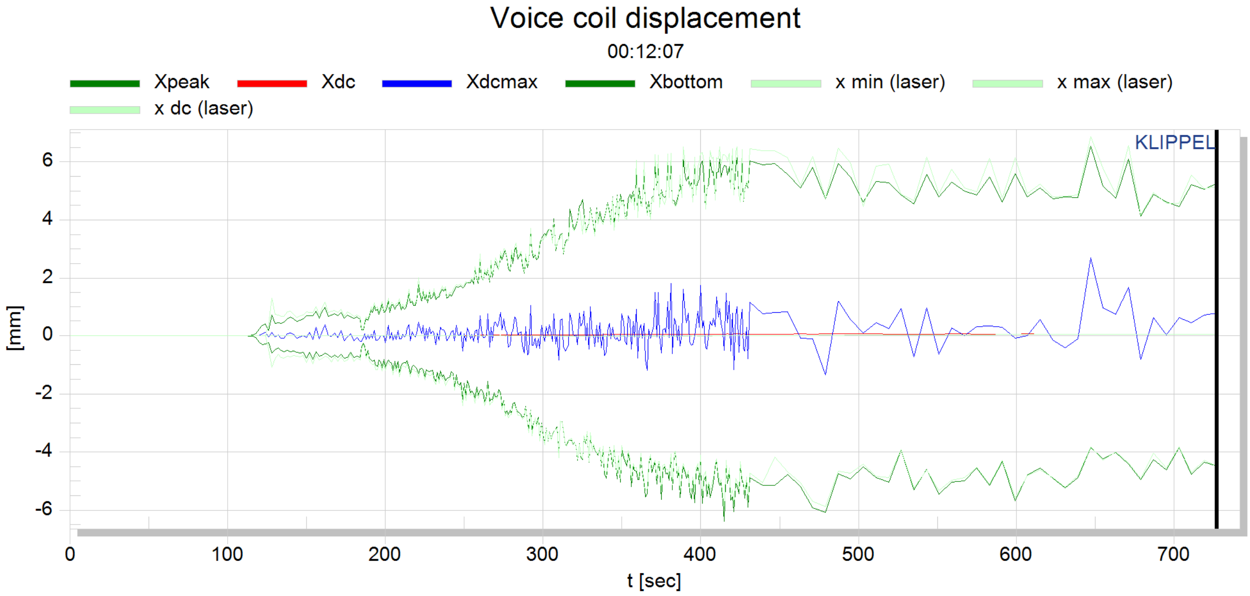
f) f) heating up to ?T_{lim} by using a band-pass filtered signal 400Hz – 2.5 kHz to measure resistance and capacity of the magnet (duration 50 min)

The Final Mode 7(7) uses the same signal as in the Nonlinear Mode but the learning speed is significantly reduced. It may run for ever to measure long term variations of the loudspeaker parameters.

Displacement *x(t)*

The displacement signal versus measurement time is represented as

$x_{peak}(t)$	positive peak of the voice coil displacement in the update interval,
$x_{dc}(t)$	averaged dc-value *in* voice coil excursion,
$x_{bottom}(t)$	negative peak value (bottom value) of the voice coil displacement in the updated interval,
$x_{dcmax}(t)$	maximal dc-value in voice coil excursion $x_{dcmax}(t)=(x_{peak}(t)+x_{bottom}(t))/2$



Asymmetrical nonlinearities produce not only second- and higher-order distortions but also a dc-part in the displacement by rectifying low frequency components.

For an asymmetric stiffness characteristic the dc-components moves the voice coil for any excitation signal in the direction of the stiffness minimum.

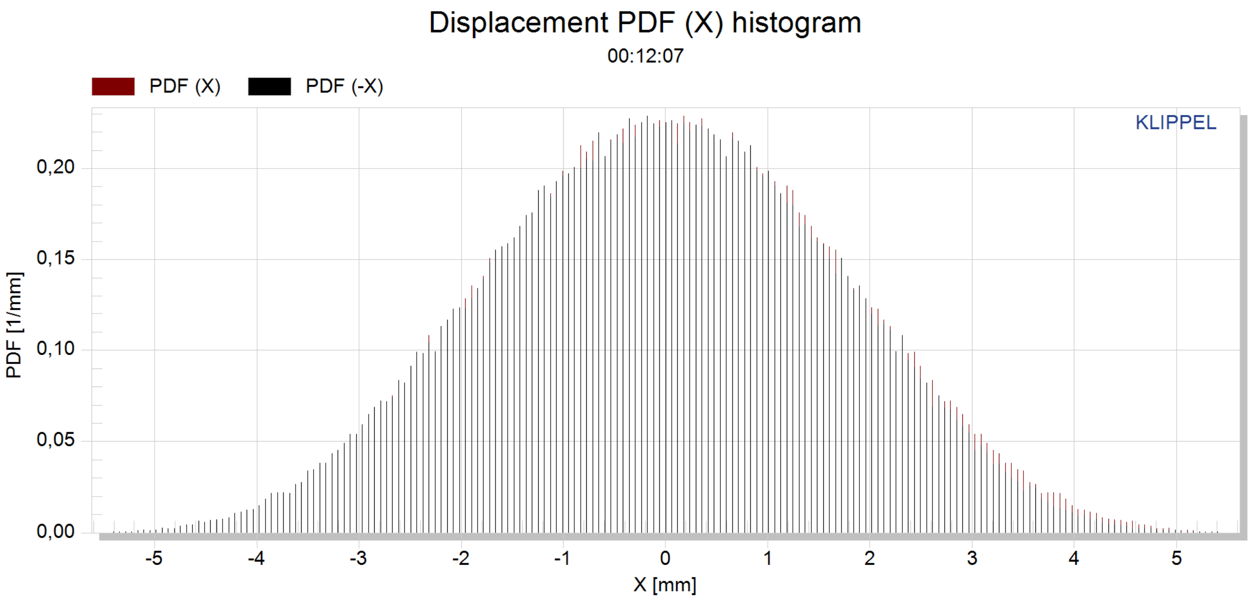
For an asymmetric force factor characteristic the dc-component depends on the frequency of the excitation signal. A sinusoidal tone below resonance ($f<f_S$) would generate or force moving the voice coil always in the force factor maximum. This effect is most welcome for stabilizing voice coil position. However, above the resonance frequency ($f>f_S$) would generate a dc-component moving the voice coil in the force factor minimum and may cause severe stability problems.

For an asymmetric inductance characteristic the dc-component moves the voice coil for any excitation signal in the direction of the inductance maximum.

Please note that the dynamically generated DC-components cause interactions between the driver nonlinearities. A optimal rest position of the coil in the gap may be destroyed by an asymmetric compliance or inductance characteristic at higher amplitudes. The module "Large Signal Simulation (SIM)" allows systematic investigation of the complicated behavior.

Displacement Probability Density Function *pdf(x)*

The probability density function of the displacement signal *pdf(x)* depends on the properties of the excitation signal (noise) and on the behavior of the transducer as well: An asymmetrical *pdf(x)* indicates a dynamic generation of a dc-component in the displacement caused by asymmetric transducer nonlinearities. The *pdf(x)* plays an important role as a weighting function in the nonlinear system identification and shows in which region of the displacement the nonlinear parameters are measured with highest precision.

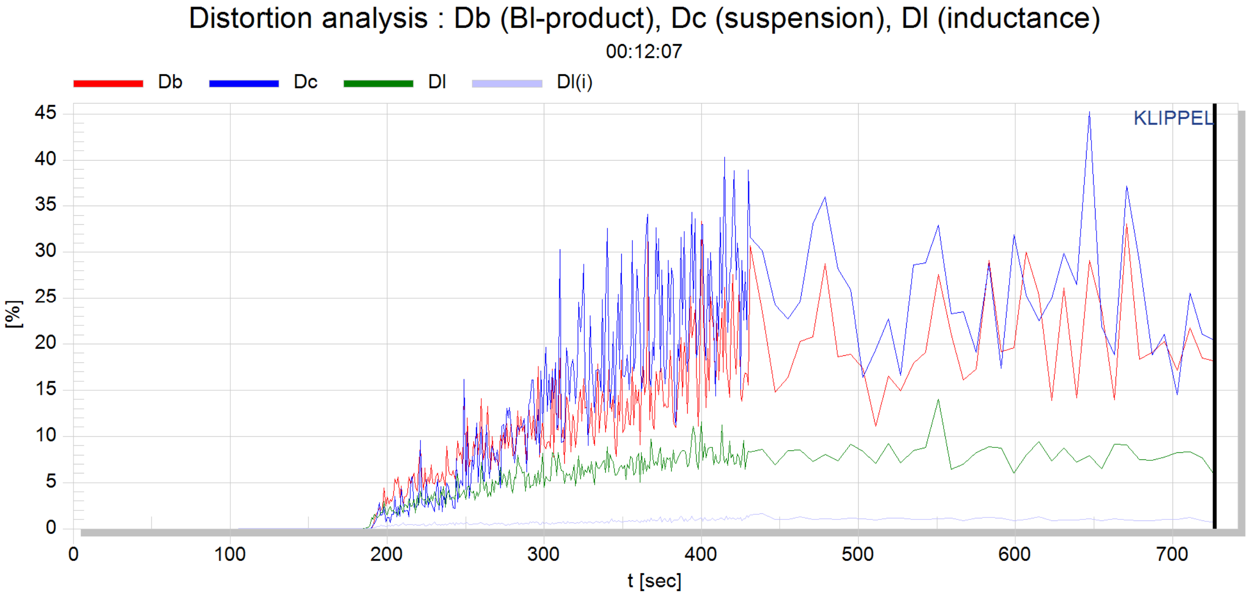


Distortion Analysis

The Distortion Analysis shows the contribution of each nonlinearity to the total distortion in the reproduced output signal for the audio-like excitation signal used during parameter measurement. The identified digital model of the transducer makes it possible to measure the peak value of the distortion components generated by force factor, compliance and inductance and to relate each value to the peak value of the total output signal (sound pressure):

d_b	relative degree of distortion generated by nonlinear force factor Bl(x)
d_L	relative degree of distortion generated by nonlinear inductance L_e(x)
d_C	relative degree of distortion generated by nonlinear compliance C_{ms}(x)
$d_{l(i)}$	relative degree of distortion generated by nonlinear inductance L_e(i) – permeability nonlinearity (flux modulation)

The distortion analysis is performed simultaneously with the parameter identification. The relative degrees of distortion are expressed in percent and presented versus measurement time (in seconds) for the transducer under test:



Each degree is a one-number representation of the distortion summarizing all of the harmonic and intermodulation components. Please note that the amount of distortion depend on the spectral properties of the excitation signal.

Remedies for Transducer Nonlinearities

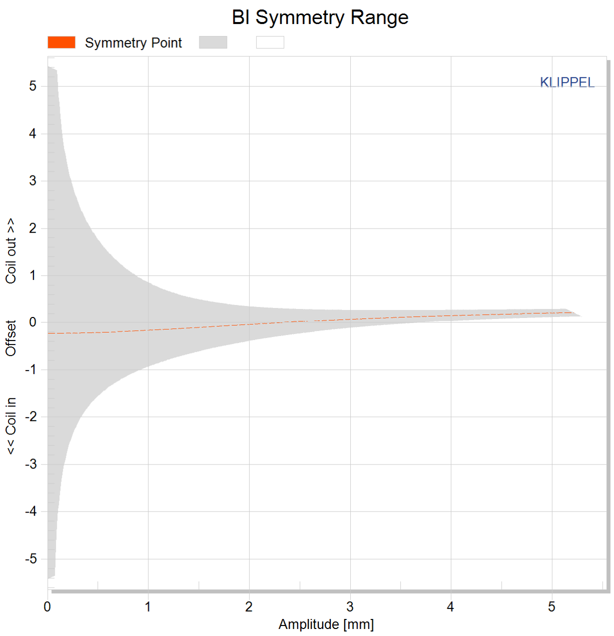
Bl Symmetry $x_b(x)$

This curve shows *the* symmetry point in the nonlinear Bl-curve where a negative and positive displacement $x=x_{peak}$ will produce the same force factor

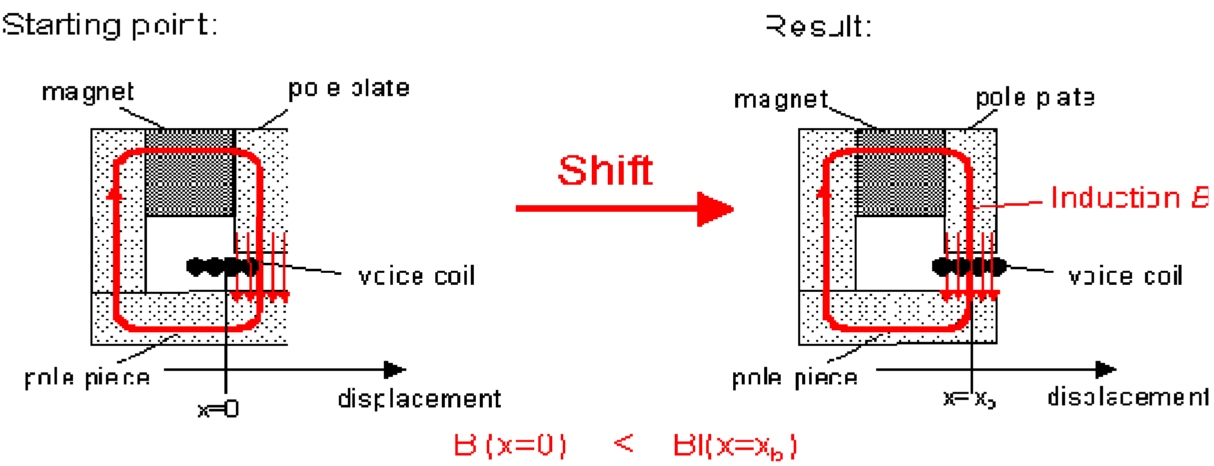
$Bl(x_b(x) + x) = Bl(x_b(x) – x)$.

If the shift $x_b(x)$ is independent on the displacement amplitude x then the force factor asymmetry is caused by an offset of the voice coil position and can be simply compensated.

If the optimal shift $x_b(x)$ varies with the displacement amplitude x then the force factor asymmetry is caused by an asymmetrical geometry of the magnetic field and can not completely be compensated by coil shifting.



Improving the rest position of the Coil



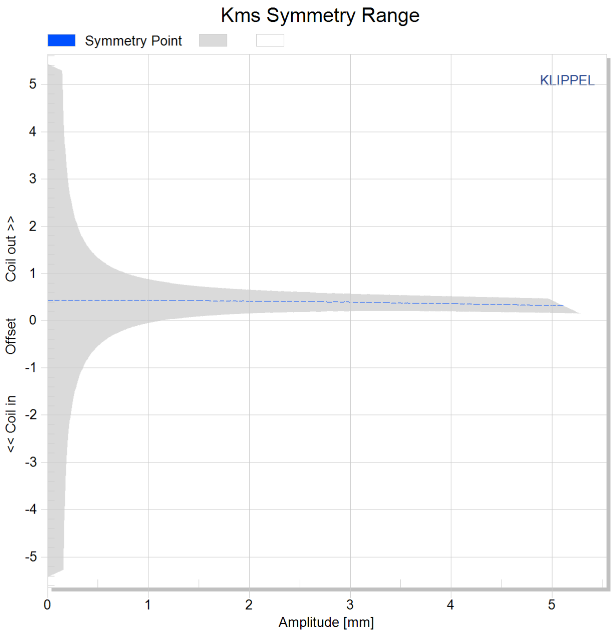
C_{ms} Symmetry $x_c(x)$

This curve shows *the* symmetry point in the nonlinear compliance curve where a negative and positive displacement $x=x_{peak}$ will produce the same compliance value

$C_{ms}(x_c(x) + x) = C_{ms}(x_c(x) - x).$

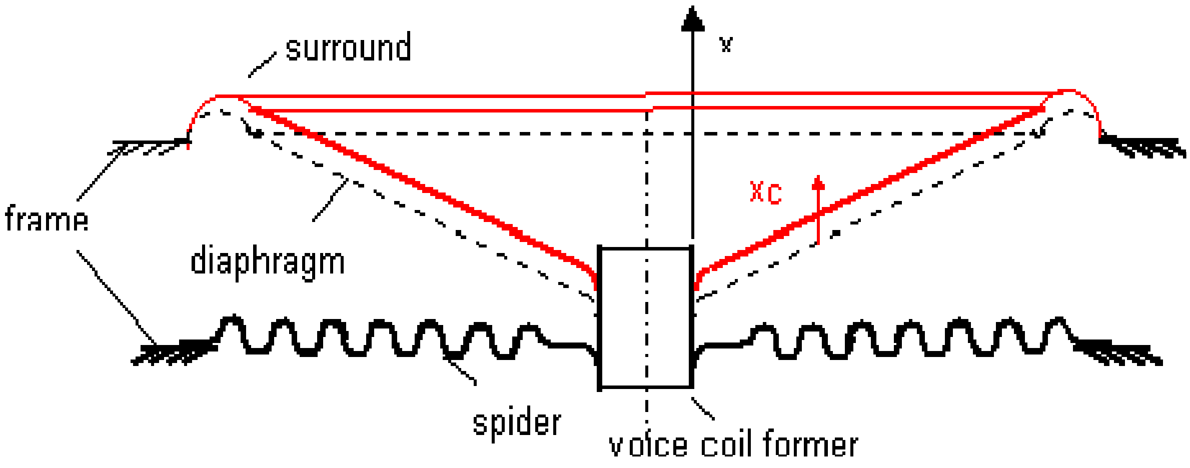
A high value of the symmetry point $x_c(x)$ at small displacement amplitudes $x \approx 0$ indicates that the rest position does not agree with the minimum of the stiffness characteristic. This may be caused by an asymmetry in the geometry of the spider (cup form) or surround (half wave).

A high value of the symmetry point $x_c(x)$ at maximal displacement $x \approx x_{max}$ may be caused by asymmetric limiting of the surround.



Adjustment of spider and surround

The asymmetry of the suspension may be reduced by reducing the distance x_R between spider and outer rim of the surround.



Please find more information in Application Note 1,2,3 (03/2009)

Number	Name	Date	Time	Comment
4	3a DIS X Fundamental, DC	03/31/09	17:29:50	Target: - Measure dc displacement generated dynamically - Measure amplitude compression Adjust Setup: - maximal voltage in PP Stimulus (use maximal Urms

				from the LSI or Umax) - adjust f start << fs << f stop

3D-Distortion Measurement (DIS)

Measurement of

VOICE COIL DISPLACEMENT

versus frequency and amplitude (linear 2D-plot)

Overview

The Introductory Report illustrates the powerful features of the 3D-Distortion Measurement Module which is a software module of the Klippel Analyzer System. Additional comments are added to the results of a practical measurement applied to the speaker specified above.

After presenting short information to the measurement technique the report comprises the following results

> Magnitude of the fundamental displacement component versus voltage and frequency of the sinusoidal excitation tone

> DC-component in voice coil displacement versus voltage and frequency

> Increase of voice coil temperature due to heating

> Spectral Components in the measured displacement at one sample of voltage and frequency combination

Measurement Method

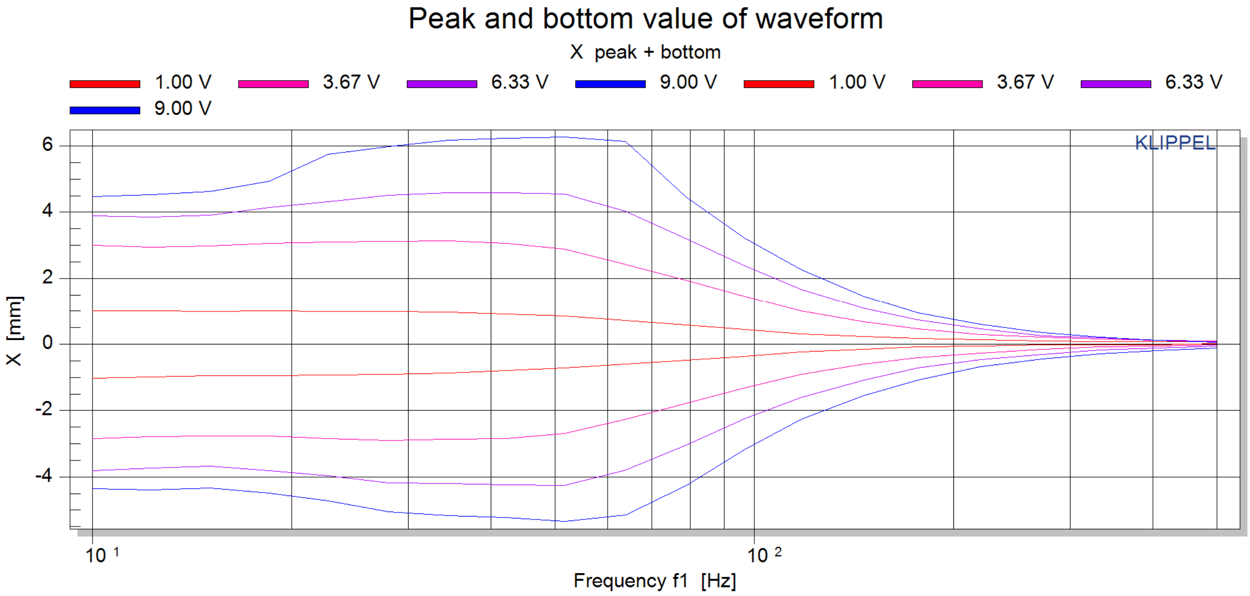
During the measurement the speaker is excited by a two-tone signal with variable frequencies f_1 and f_2 and terminal voltage u_1 and u_2, respectively, to measure a 3D response versus frequency and amplitude. The samples of voltage and frequency variation of both tones may be linearly or logarithmically spaced. In the example presented here we use a linear spacing and a graphical representation on linear axes.

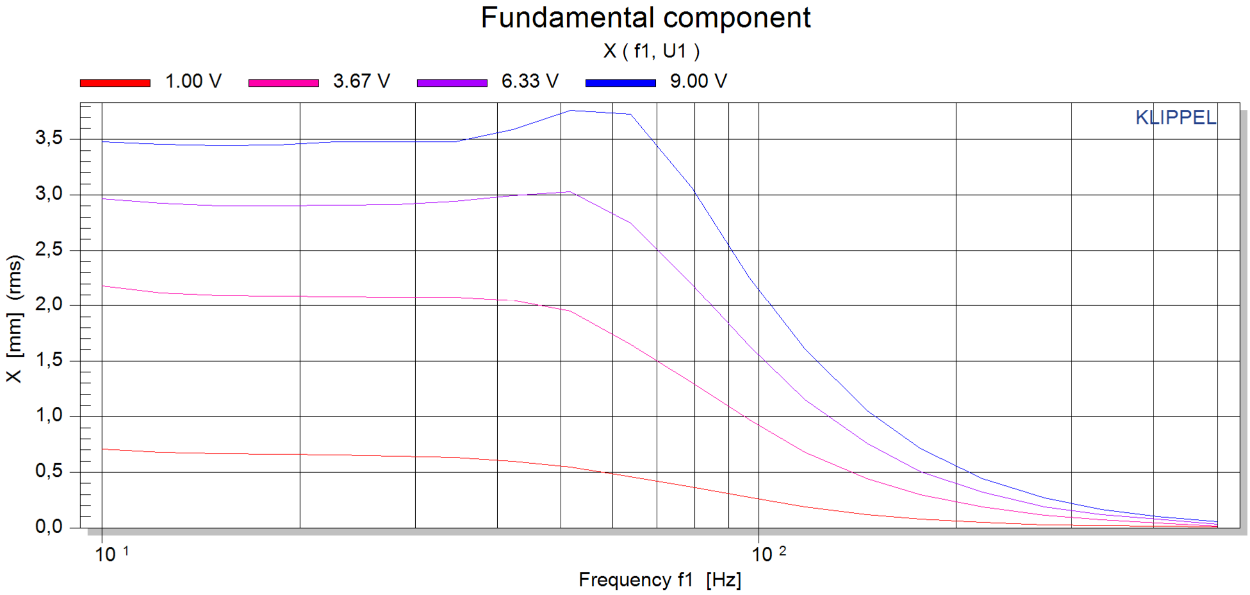
The hardware of the Distortion Analyzer 1 allows the acquisition of two analog input signal at high precision (100 dB SNR) at 96kHz sampling frequency. Additional measurement of the electrical signals at the speakers terminals is the basis for controlling the amplitude of the excitation signal just at the loudspeaker terminals to be independent of the gain setting of power amplifier and to be able to monitor the instantaneous voice coil temperature during the measurement. This information is used by a protection system which pauses the measurement if either the heating of the coil or the total harmonic distortion exceeds a user-defined threshold.

A spectral analysis allows to determine the magnitude of the fundamental, dc-component, harmonics and sum- and difference-tone intermodulation components up to 8^{th}-order.

In this report all of the results are represented as a 2D-plot comprising multiple curves versus frequency at specified amplitude. Alternatively, the results may be presented as a 2D-plot versus amplitude at specified frequency or as a 3D-plot versus amplitude and frequency.

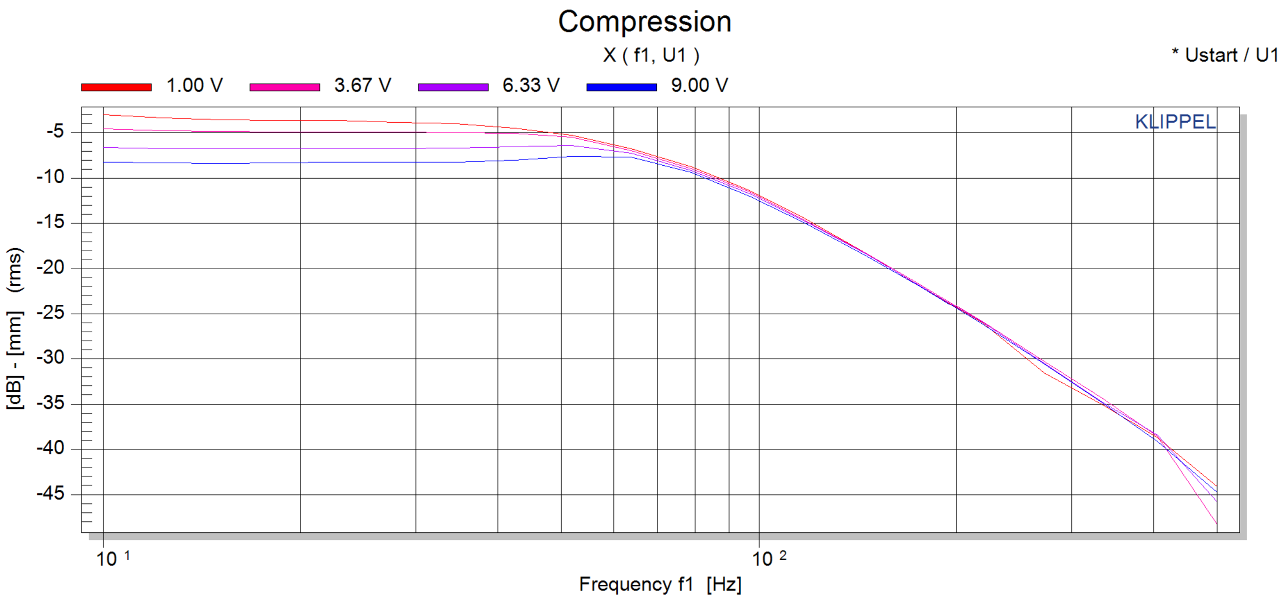
Fundamental





The 2D-plot shows the displacement versus frequency at specified amplitudes on linear axes to show the compression effect. Only at sufficiently low amplitudes there is a linear relationship between input and output signals.

COMPRESSION OF THE FUNDAMENTAL

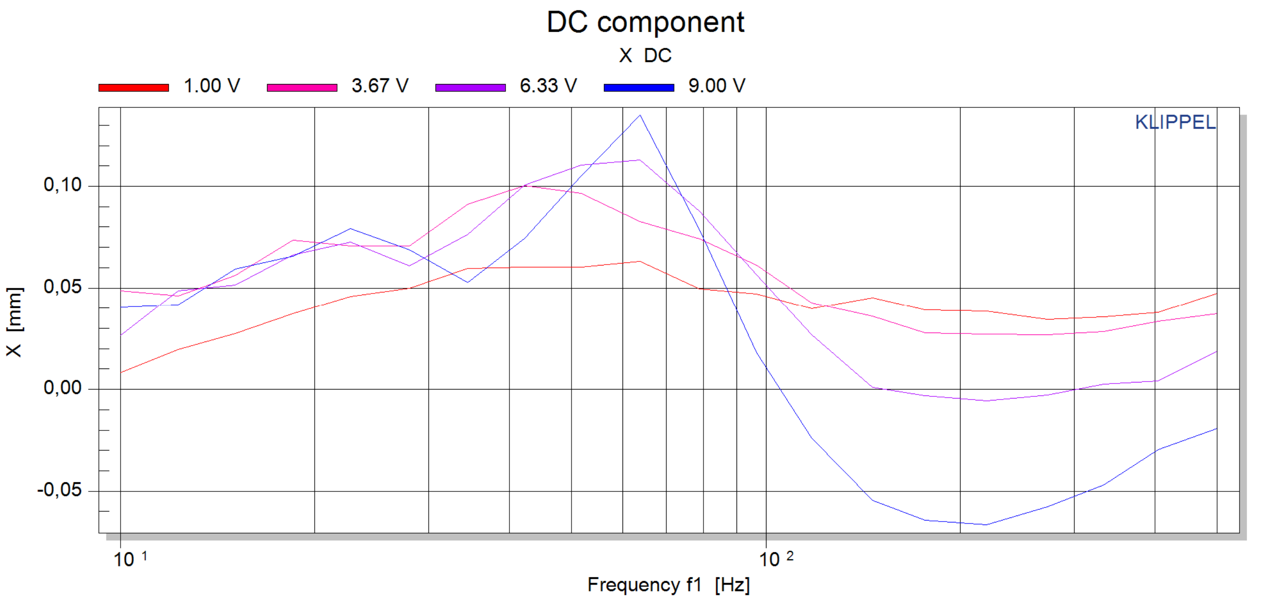


DC Component

The dc-component in the displacement will be generated dynamically if nonlinearities (Bl(x)-product, compliance Cms(x) and Le(x)) have an asymmetrical characteristic. At the resonance frequency the asymmetry of the stiffness is usually the dominant source for DC. Above the resonance frequency the force factor (Bl(x)-product) contributes more and more. If the motor nonlinearity is large and the compliance is low the system becomes instable f=2fs pushing the coil literally out to the gap. A generated dc-component will destroy the optimal working point. For example a dc-component generated by asymmetric stiffness may shift the voice coil position producing substantial Bl-distortion at higher frequencies eventually.

Thus ensuring a small DC component is a essential requisition for ensuring stable performance and low distortion in the large signal domain.

The figure below shows the dc component in voice coil displacement for varied voltage U and frequency f. of the sinusoidal excitation tone.



Please find more information in Application Note 13 (09/2008)

Number	Name	Date	Time	Comment
5				Target:

	3b DIS Motor stability	03/31/09	17:35:43	- Check stability of motor - Measure dc displacement at f1 = 1.5 fs (where Xdc is maximal) Adjust Setup: - Set frequency f1 = 1.5 fs - Maximal voltage in PP Stimulus (High voltages are no problem if the Protection is activated.)

3D-Distortion Measurement (DIS)

Measurement of

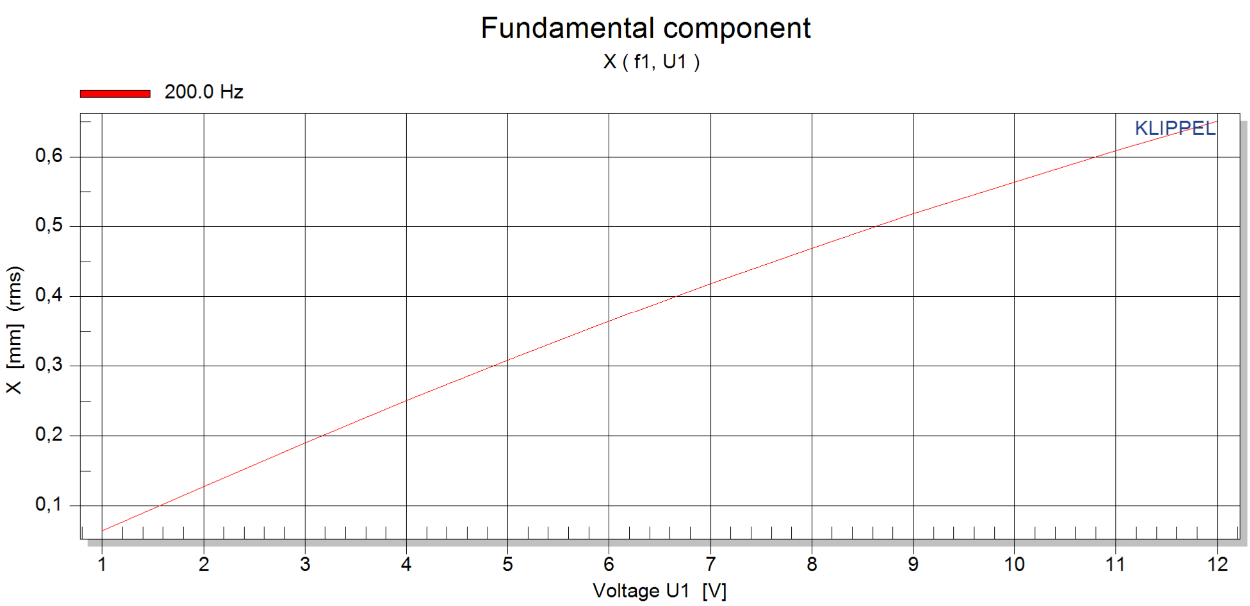
VOICE COIL DISPLACEMENT

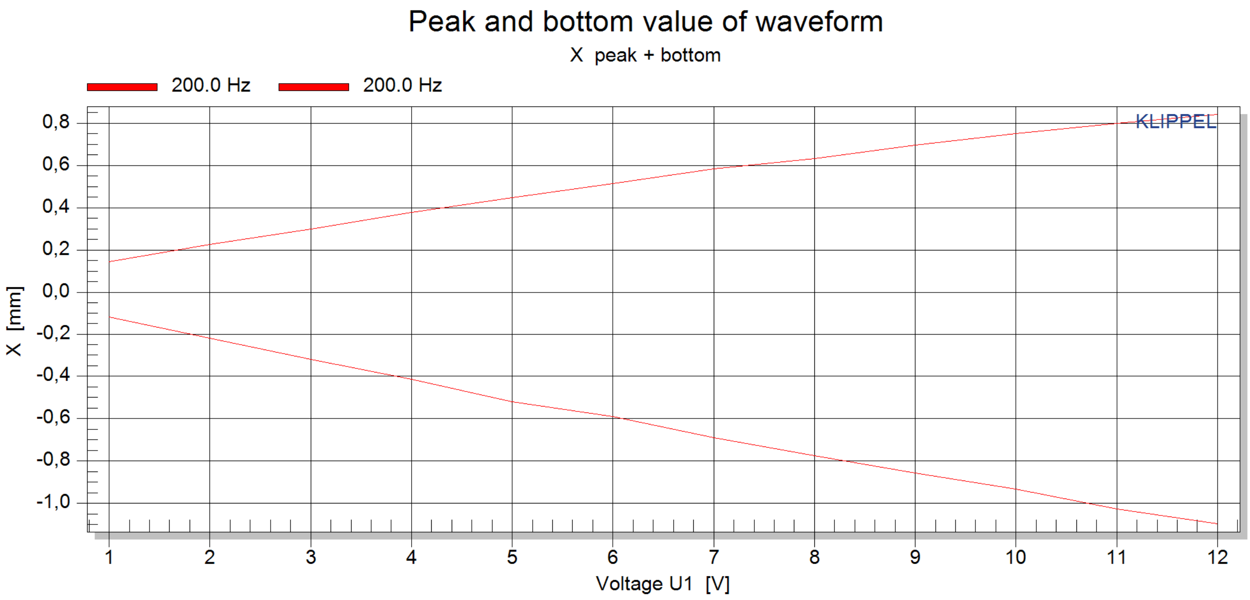
versus amplitude at one critical frequency (linear 2D-plot)

Measurement Method

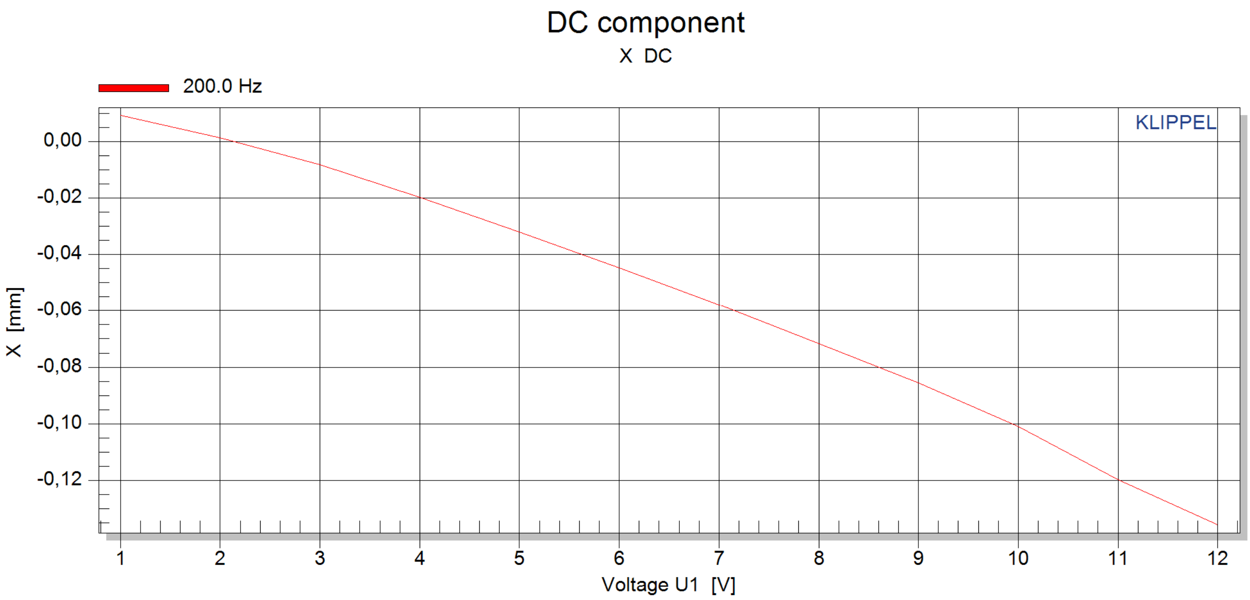
During the measurement the speaker is excited by a single-tone signal with fixed frequencies f_1 versus the stepwise increased terminal voltage u_1.

Fundamental



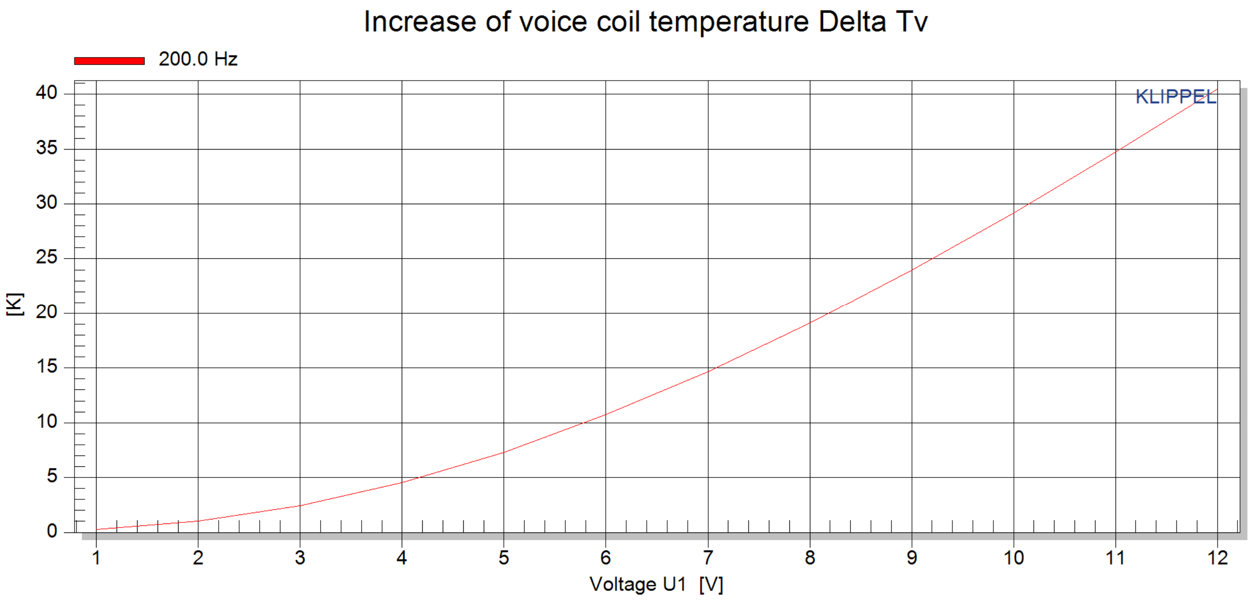


DC Component



Voice Coil Temperature

Used to protect the measured speaker from high voltages.



State Variables at last sweep point

Symbol	**Value**	**Unit**	**Comment**
SELECTED SWEEP POINT			
f1	199.95	Hz	first excitation tone
U1 (f1)	12.00	V rms	voltage first excitation tone
Delta Tv	40.5	K	increase of voice coil temperature
X peak	0.84092	mm	peak value of X
X bottom	-1.0953	mm	bottom value of X
X head	27.406	dB	headroom of X
X ac	0.65102	mm rms	ac part of X
	-3.7282	dB	ac part of X
PROTECTION			
Delta Tv lim	60.000	K	allowed increase of voice coil temperature

Please find more information in Application Note 14 (09/2008)

Number	**Name**	**Date**	**Time**	**Comment**
7	4a DIS IM Dist. (bass sweep) P	03/31/09	17:44:02	Target: Measure Intermodulation distortion in current and sound pressure by using a variable bass tone fs/4 < f1 < 4fs and a fixed voice tone f2 >> fs Adjust Setup: - Maximal voltage in PP Stimulus (use Urms f. LSI or Umax) - Microphone sensitivity
				Target: Change display mode to distortion versus current

8	4b DIS IM Dist. (bass sweep) I	03/31/09	17:44:02	Adjust Setup: - copy and paste measured operation 4a - set State Signal to Current in PP Display - change operation name to "4b DIS IM Dist. (bass sweep) I"

3D-Distortion Measurement (DIS)

Intermodulation Distortion

in radiated SOUND PRESSURE and CURRENT
versus frequency and amplitude (linear 2D-plot)

Overview

The Introductory Report illustrates the powerful features of the 3D-Distortion Measurement Module which is a software module of the Klippel Analyzer System. Additional comments are added to the results of a practical measurement applied to the speaker specified above.

After presenting short information to the measurement technique the report comprises the following results

Second- and third-order intermodulation component versus amplitude and frequency in a 2D-plot

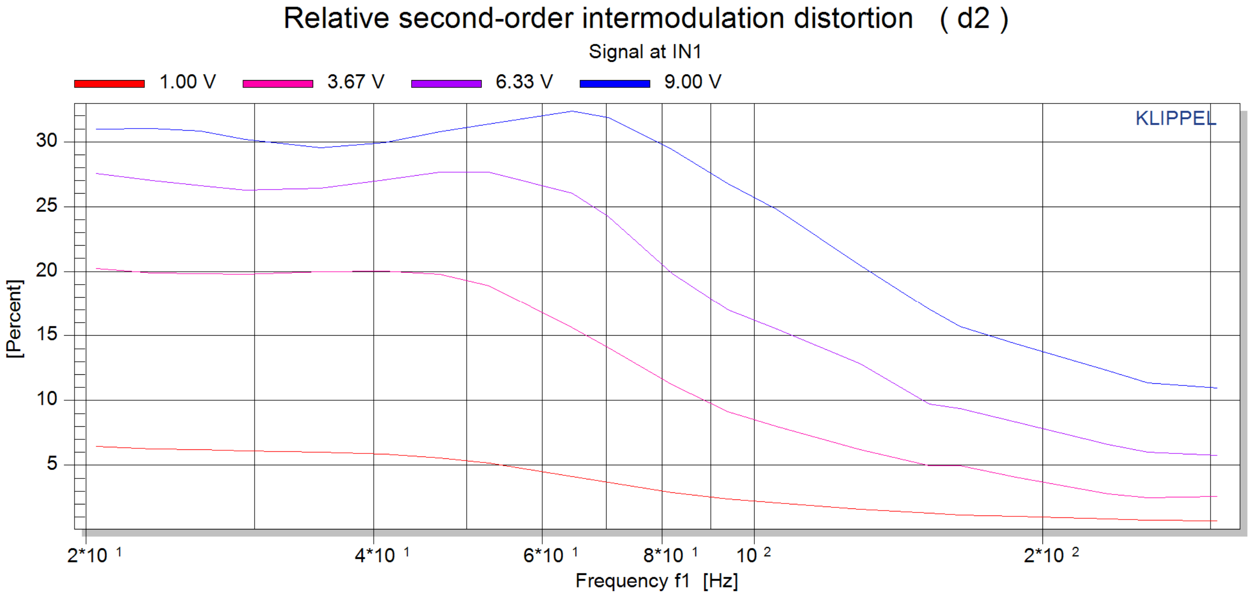
Intermodulation Distortion in Percent (relative)

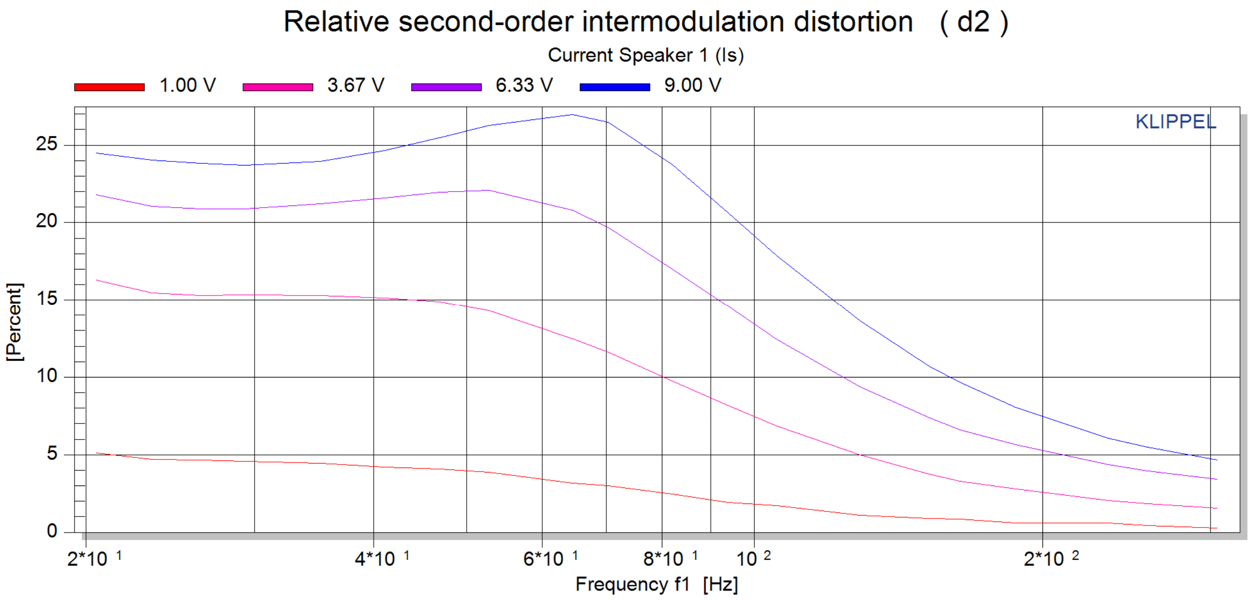
The figure below shows the magnitude response of the second-order intermodulation distortion of the radiated sound pressure for a sinusoidal excitation tone versus frequency f_1 at various input voltages $U_1 = U_2$ defined according to IEC 60268 as

$$d_2 = \frac{P(f_2 - f_1) + P(f_2 + f_1)}{P(f_2)} * 100\,\%$$

where p is the magnitude of the sound pressure component. During the measurement f_2 is set to a fixed frequency above f_s of the driver representing a voice tone in the audio signal. The frequency f_1 represents a lower-frequency component (bass) in the transfer band.

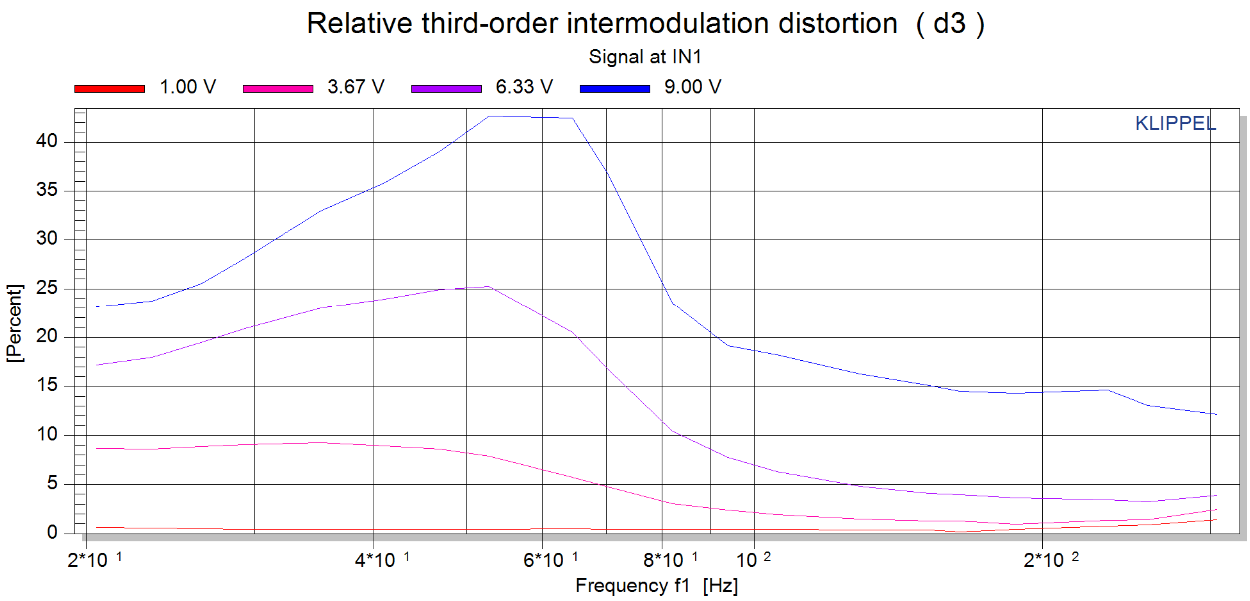
Second figure represents the intermodulation distortion in the current signal measured at the terminals.

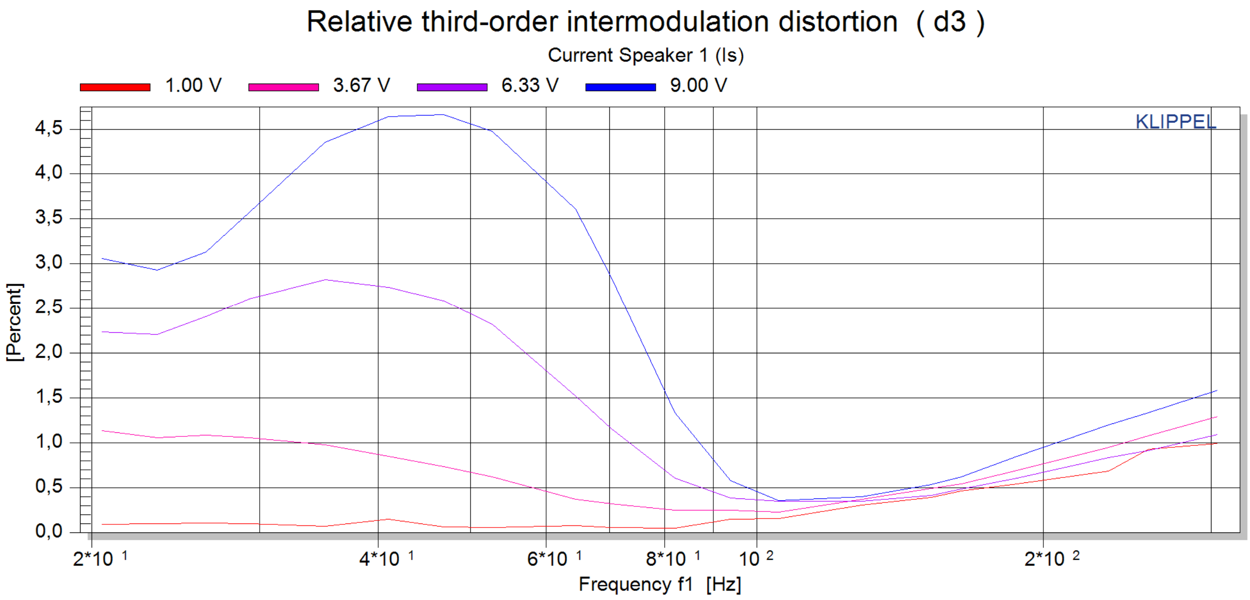




The figure below shows the magnitude response of the third-order harmonic distortion of the radiated sound pressure for a sinusoidal excitation tone versus frequency f at various input voltages $U_1 = U_2$ defined according to IEC 60268 as

$$d_3 = \frac{P(f_2 - 2f_1) + P(f_2 + 2f_1)}{P(f_2)} * 100\,\%$$





Number	Name	Date	Time	Comment
9	4c DIS IM Dist. (voice sweep) P	03/31/09	17:54:28	Target: Measure Intermodulation distortion in current and sound pressure by using a fixed bass tone f2 < fs and a variable voice tone f1>> fs Adjust Setup: - Maximal voltage in PP Stimulus (use Urms from the LSI or Umax) - Microphone sensitivity
10	4d DIS IM Dist. (voice sweep) I	03/31/09	17:54:28	Target: Change display mode to distortion versus current Adjust Setup: - copy and paste measured operation 4a - set State Signal to Current in PP Display - change operation name to "4d DIS IM Dist. (voice sweep) I"

3D-Distortion Measurement (DIS)

Intermodulation Distortion

in radiated SOUND PRESSURE and CURRENT
versus frequency and amplitude (linear 2D-plot)

Overview

The Introductory Report illustrates the powerful features of the 3D-Distortion Measurement Module which is a software module of the Klippel Analyzer System. Additional comments are added to the results of a practical measurement applied to the speaker specified above.

After presenting short information to the measurement technique the report comprises the following results

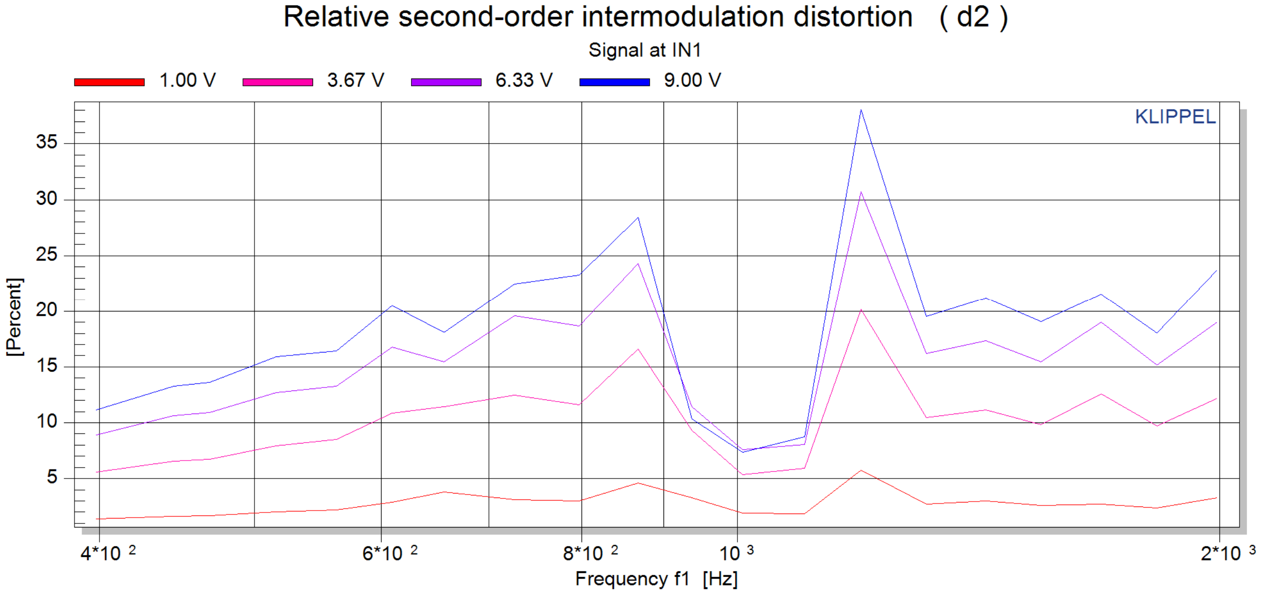
Second- and third-order intermodulation component versus amplitude and frequency in a 2D-plot

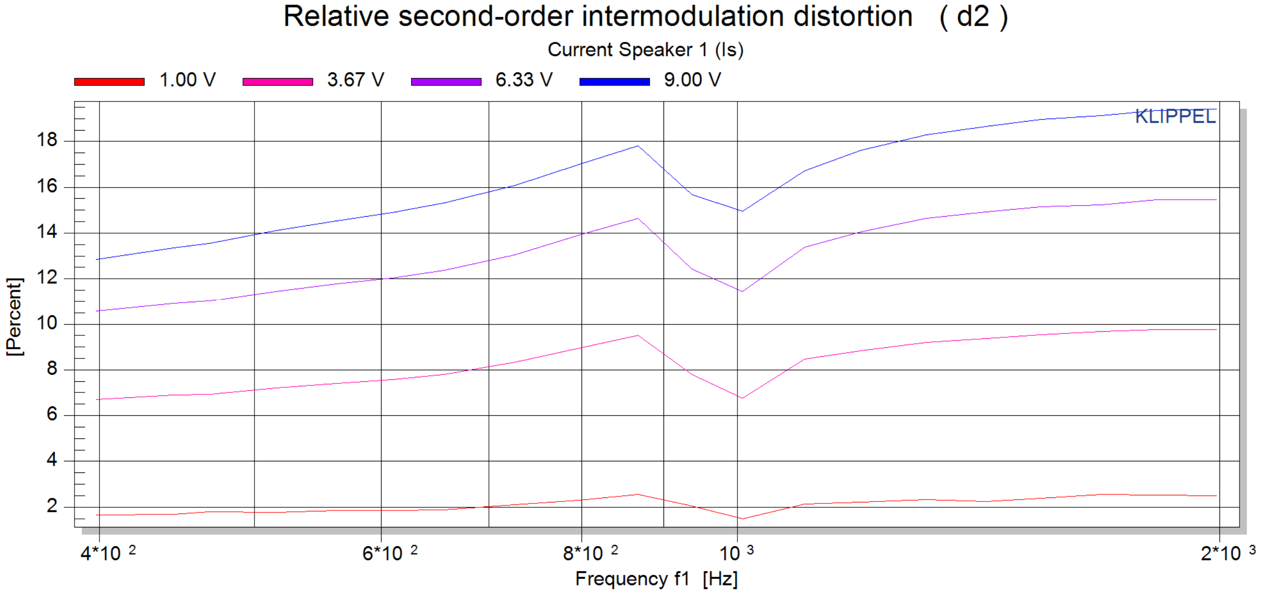
Intermodulation Distortion in Percent (relative)

The figure below shows the magnitude response of the second-order intermodulation distortion of the radiated sound pressure for a sinusoidal excitation tone versus frequency f_1 at various input voltages $U_1 = U_2$ defined according to IEC 60268 as

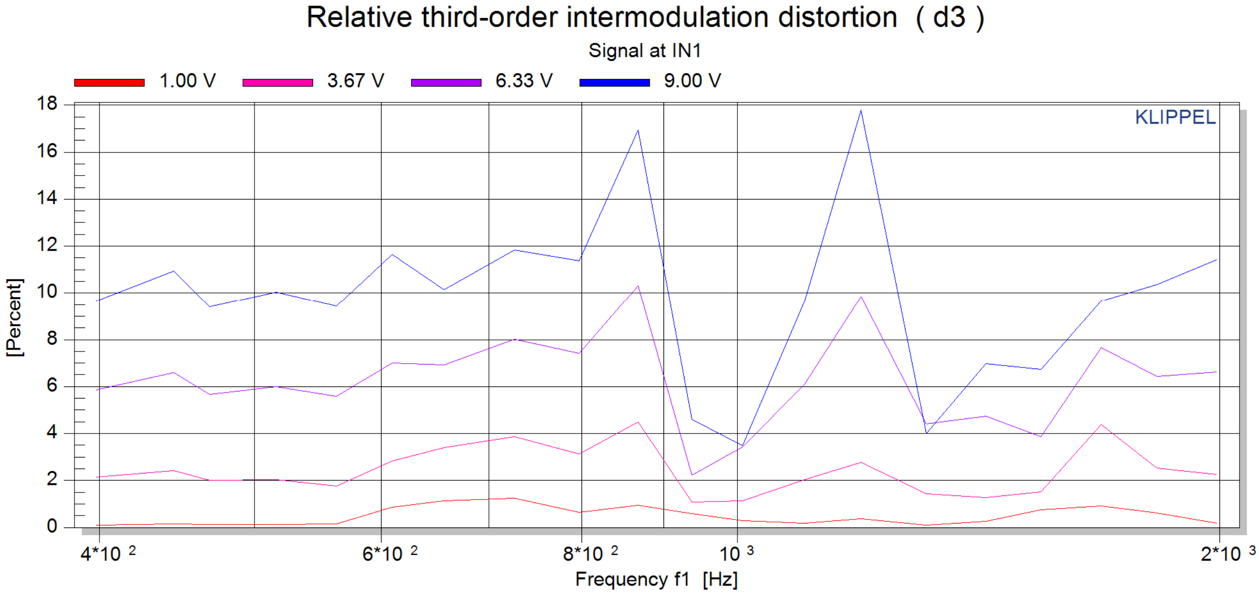
$$d_2 = \frac{P(f_2 - f_1) + P(f_2 + f_1)}{P(f_1)} * 100\,\%$$

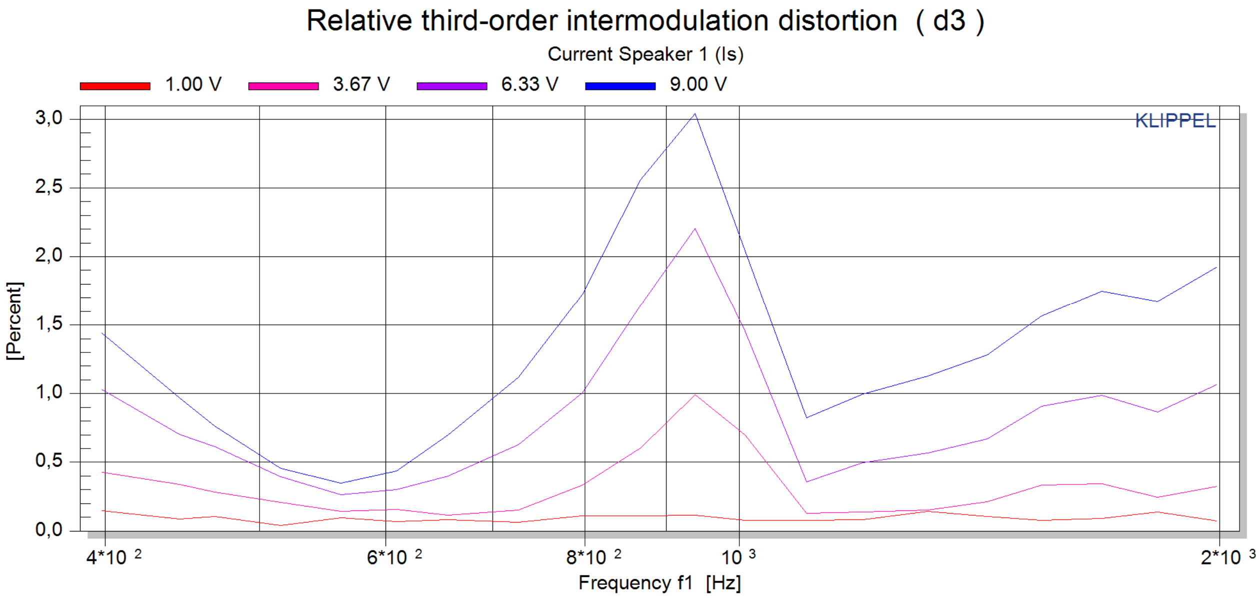
where p is the magnitude of the sound pressure component. During the measurement f_2 is set close to the resonance frequency f_s of the driver representing a bass tone in the audio signal. The frequency f_1 represents a higher-frequency component (voice) in the transfer band.





Second figure represents the intermodulation distortion in the current signal measured at the terminals.





Number	Name	Date	Time	Comment
11	5a TRF Crest Harm. CLEAN	04/06/09	12:56:21	Voltage applied to terminals: 3V Target: Search for maximal voltage without rub and buzz Adjust Setup: - Increase Voltage in PP Stimulus slowly - Microphone sensitivity - harmonics > 20th order in PP I-Dist
12	5b TRF Peak Harm. CLEAN	04/06/09	12:56:35	Voltage applied to terminals: 3V Target: Search for maximal voltage without rub and buzz Adjust Setup: - Increase Voltage in PP Stimulus slowly - Microphone sensitivity - harmonics > 20th order in PP I-Dist

Transfer Function Module (TRF)

Instantaneous Distortion

Introduction

The traditional distortion measurement transforms the time signal into the frequency domain to separate fundamental, harmonic and intermodulation components. This technique considers only the mean power in the analyzed interval and neglects the phase information. A new technique for the measurement of the signal distortion in time domain is presented that exploits both amplitude and phase information. It reveals the fine structure of the distortion and its dependency on frequency, displacement or other state variables. Besides the rms-value of the distortion the peak value and the crest factor are important characteristics for detection of rub and buzz phenomena. The practical application, the interpretation and the diagnostics of defects are discussed.

Introduction

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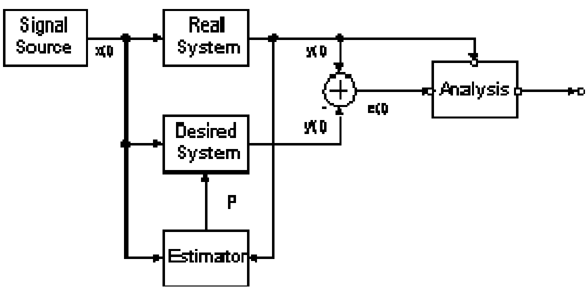
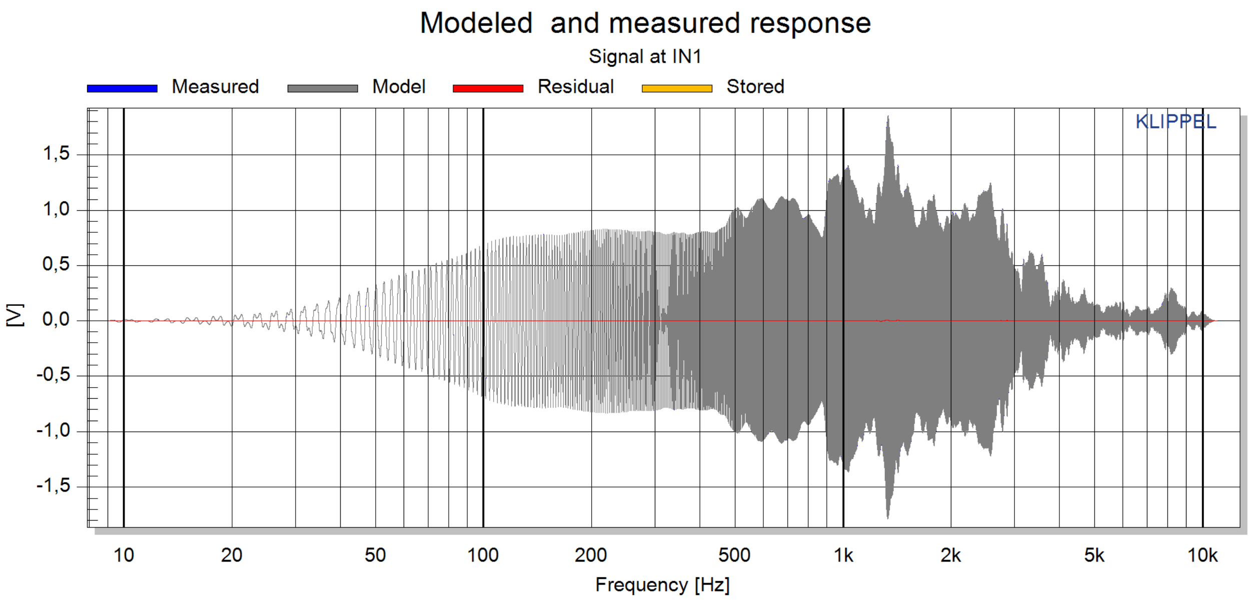
$$e(t) = y(t) - y'(t)$$

is a time signal depending on the properties of the stimulus and the system under test. The desired output *y'(t)* is generated by a model of the real system which is excited by the same stimulus *x(t).*



If the desired system is linear and models the linear output of the real system precisely we see only the nonlinear effects of the real system in the residual distortion e(t). To separate the triggered distortion from the regular distortion the desired system has to model the motor and suspension nonlinearities. In both cases the parameters of the desired system have to be adjusted to the real system by estimating the optimal parameter vector P.



Mean Value of Distortion (MD)

Neglecting the phase and fine structure of the residual distortion we may calculate the *mean distortion* (MD)

$$d_{MD}[k] = \frac{e_{RMS}}{y_{RMS}} \tag{2}$$

for each interval using the rms-value of the residual distortion signal

$$e_{RMS} = \sqrt{\frac{1}{t_{i+1} - t_i} \int_{t_i}^{t_{i+1}} e^2(t)dt}. \tag{3}$$

and the rms-value of the measured signal *y(t)*. This measure may also be expressed in dB or in percent. Representing each interval by one number the mean distortion reduces the amount of data significantly. For stationary distortion the loss of information is negligible.

For the sinusoidal sweep the time to frequency mapping gives the *mean harmonic distortion* (MHD)

$$d_{MHD}[f_i] = \frac{e_{RMS}}{y_{RMS}}. \tag{4}$$

If the desired system is linear and the residual distortion comprises the nonlinear distortion and the measurement noise the MHD coincides with the traditional THDN.

Peak Value of Distortion (PD)

Peaky, short time, low-energy distortion leads to large peaks in ID which are not represented by the mean distortion MD. In this case the peak value of ID in the interval is an valuable information that exploits both phase in amplitude information. The *peak distortion* (PD)

$$d_{PD}[k] = \frac{e_{peak}}{y_{RMS}} \tag{5}$$

is the ratio between the absolute peak value

$$e_{peak} = \max_{t_i \le t \le t_{i+1}} \left[\, | e(t) | \, \right] \tag{6}$$

and the rms-value of the measured signal .It is also a relative quantity and can be expressed in dB and percent.

For the sinusoidal sweep the time to frequency mapping yields the *peak harmonic distortion* (PHD)

$$d_{PHD}[f_k] = \frac{e_{peak}}{y_{RMS}}. \tag{7}$$

Crest Factor of Distortion (CD)

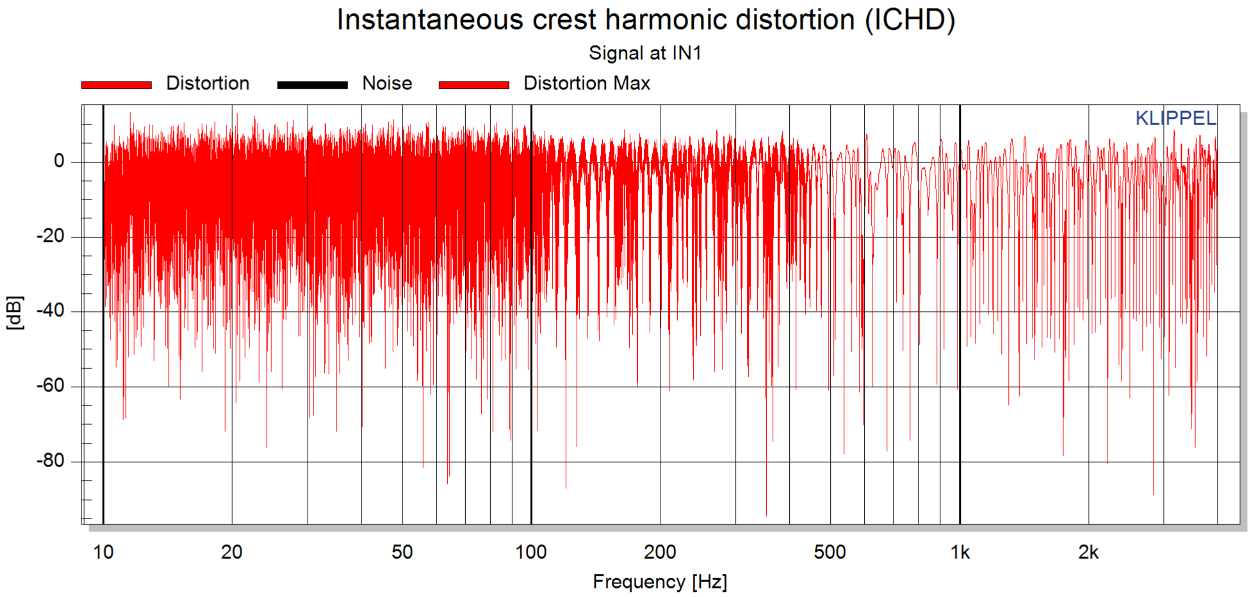
In traditional signal analysis the crest factor has proofed to be an excellent and universal measure for the transient properties of a signal. A constant signal has a crest factor of 0 dB, a sinusoidal signal corresponds to 3 dB while crest factors of more than 10 dB indicate large fluctuations. Therefore it seems promising to apply the crest factor to the residual distortion signal *e(t)*. The *crest factor of distortion* (CD)

$$d_{CD}[k] = \frac{e_{peak}}{e_{RMS}} \tag{8}$$

is defined as the ratio between the peak value and the rms-value of the *e(t)*, as defined in Eq. (6) and Eq. (3), respectively. This measures uses the phase information of the harmonics and is almost independent of the power of the distortion.

For the sinusoidal sweep the time to frequency mapping yields the *crest factor of harmonic distortion* (CHD)

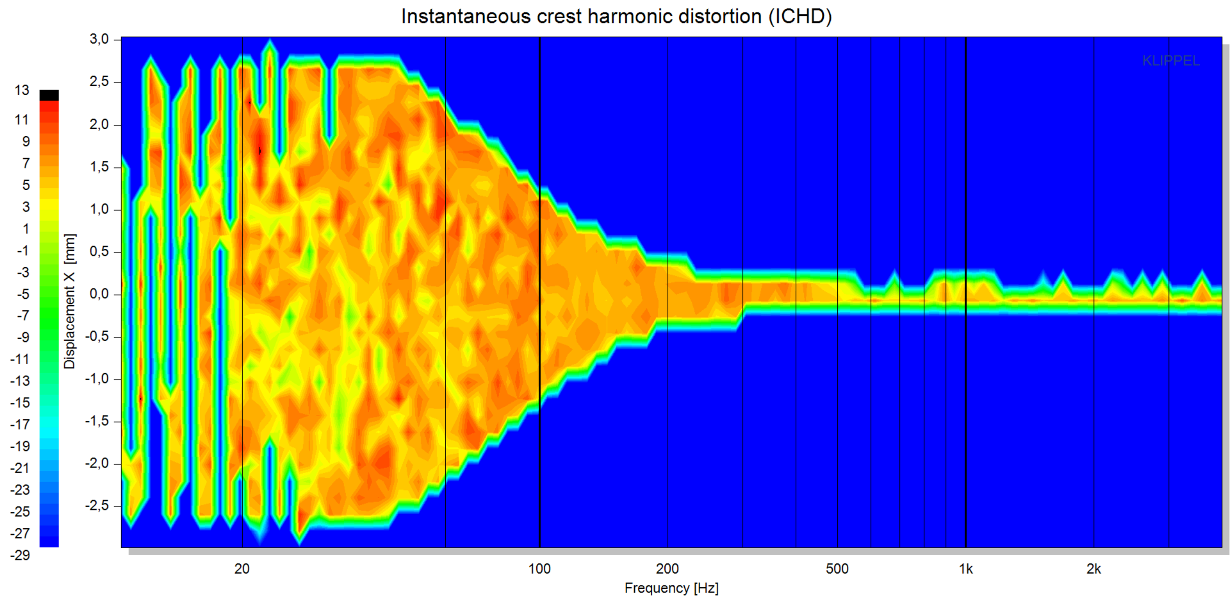
$$d_{CHD}[f_k] = \frac{e_{peak}}{e_{RMS}} \tag{9}$$



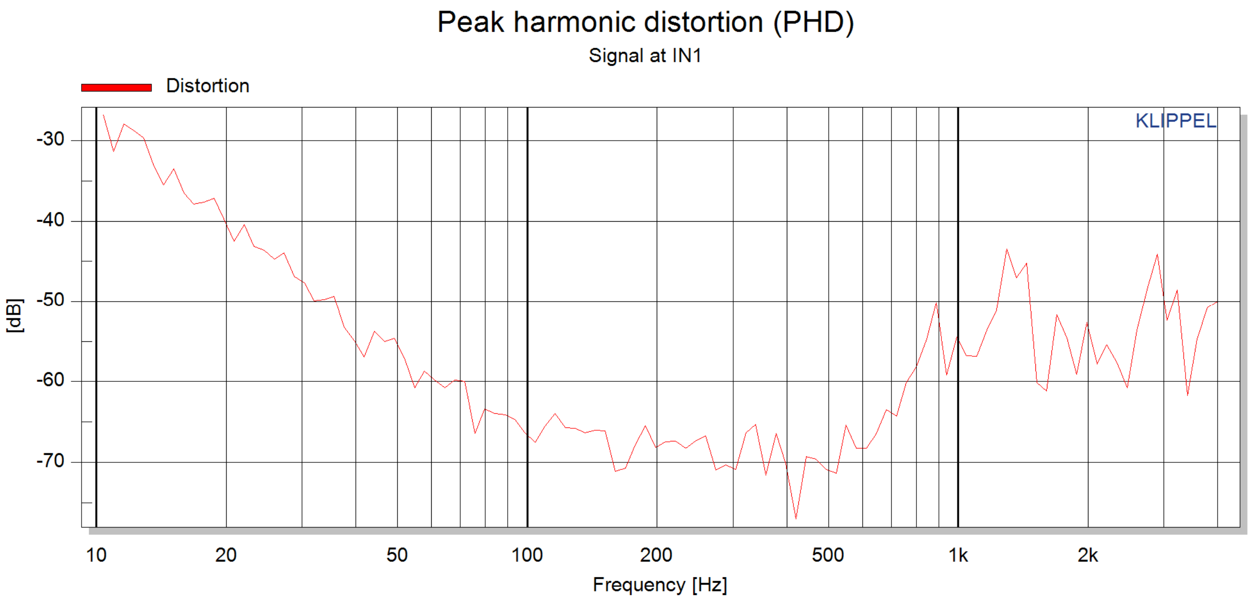
The crest factor is below 8-10 dB for the regular distortion. The triggered distortion have a significant higher crest factor.

Distortion versus frequency and sound pressure amplitude

Further insight into the particular distortion generation process can be gained by plotting the distortion signal versus the state variables of the system. The sound pressure signal and the voice coil displacement are two interesting candidates. Most of the triggered distortion are initiated by the movement of the cone. The displacement measurement requires a special sensor. Similar information can also be derived from the sound pressure measured in the near field as it is the second derivative of the displacement.



A black colour in the plot represents an instantaneous crest factor greater than 12 dB which is typical for a loudspeaker defect.



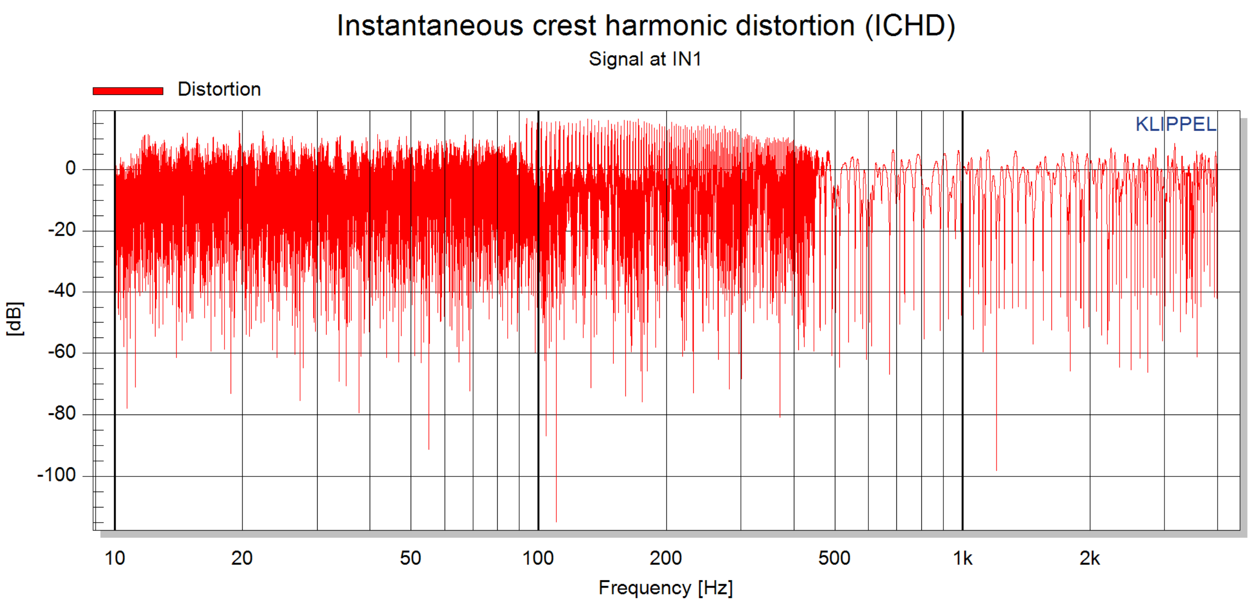
Please find more information in Application Note 22 (09/2008)

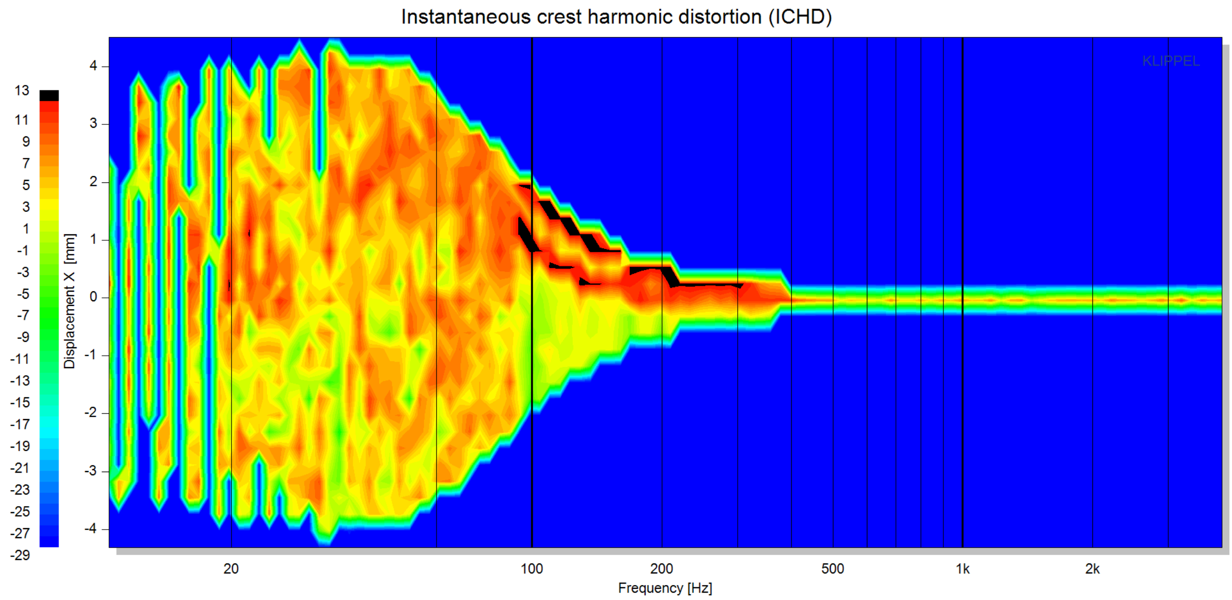
Number	Name	Date	Time	Comment
13	5c TRF Crest Harm. R&B	04/06/09	12:57:13	Voltage applied to terminals: 5V Target: Search for voltage where rub and buzz starts Adjust Setup: - Increase Voltage in PP Stimulus slowly - Microphone sensitivity - harmonics > 20th order in PP I-Dist
14	5d TRF Peak Harm. R&B	04/06/09	12:57:27	Voltage applied to terminals: 5V Target: Search for voltage where rub and buzz starts Adjust Setup: - Increase Voltage in PP Stimulus slowly

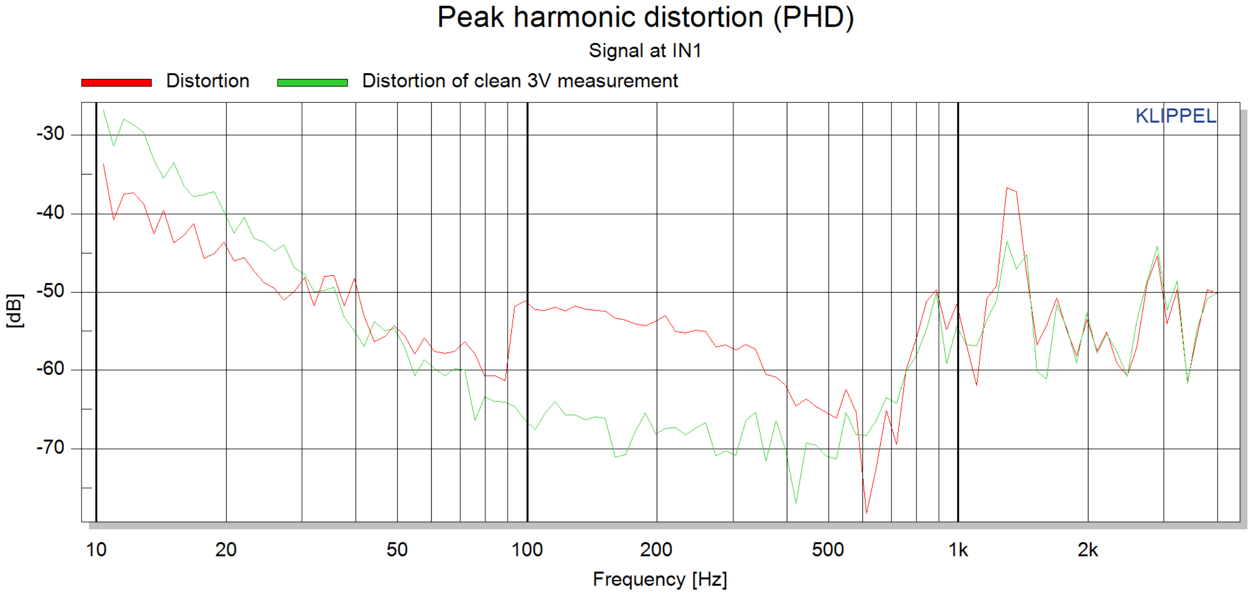
- Microphone sensitivity
- harmonics > 20th order in PP I-Dist

Transfer Function Module (TRF)

Instantaneous Distortion





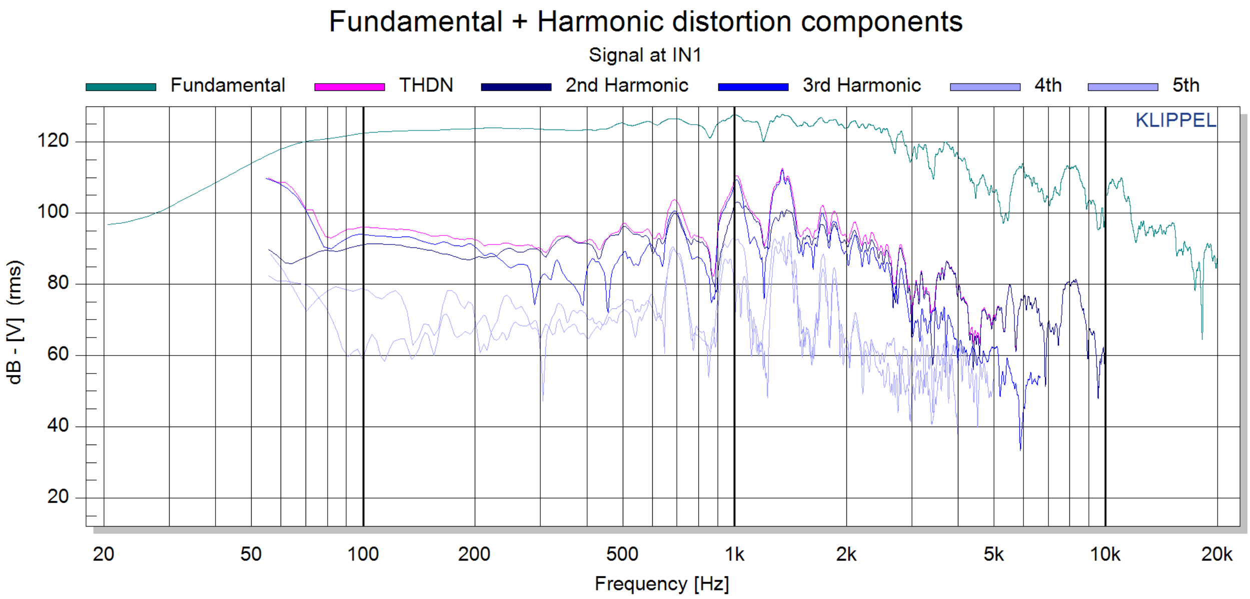


Number	Name	Date	Time	Comment
15	6 TRF SPL + Harm. (Umax)	03/31/09	17:55:24	Target: Measure SPL + harmonic distortion at high displacement Adjust Setup: - Maximal voltage in PP Stimulus (use Urms from LSI or Umax) - Microphone sensitivity

Transfer Function Module (TRF)

Fundamental + Harmonics (dB)

acoustical near field transfer function and harmonic distortion measurement at high level



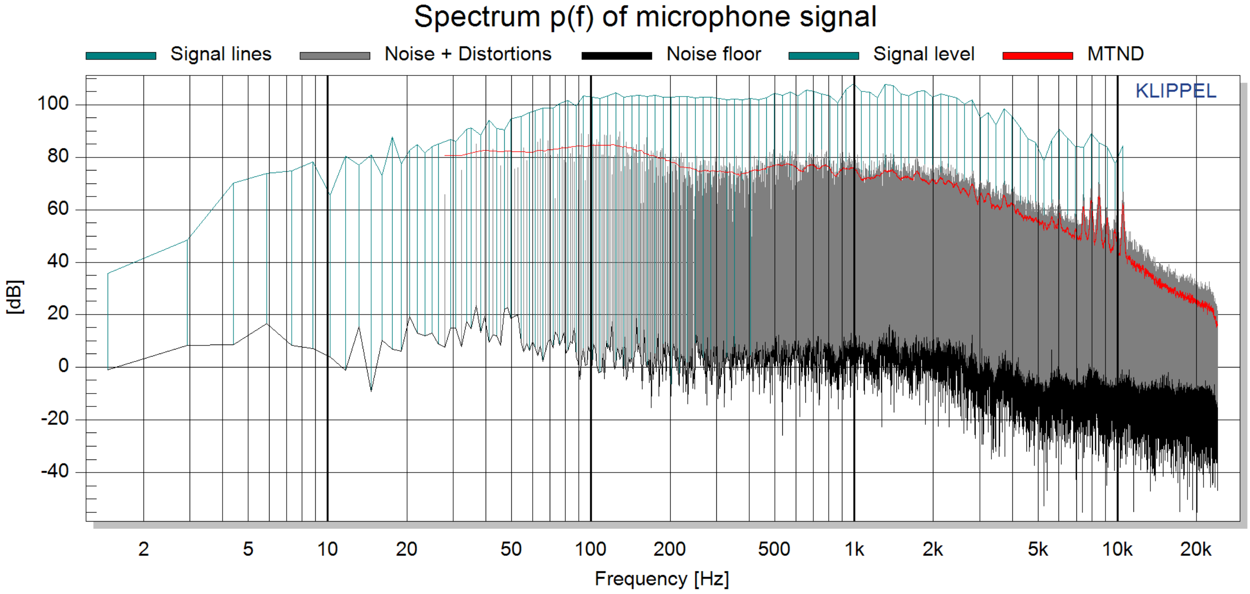
Number	Name	Date	Time	Comment
	7 LPM Multitone			Target: Measurement of Harmonic, Intermodulation Distortion at high amplitudes (using audiolike signal) Adjust Setup: Voltage in PP Stimulus to Urms from LSI

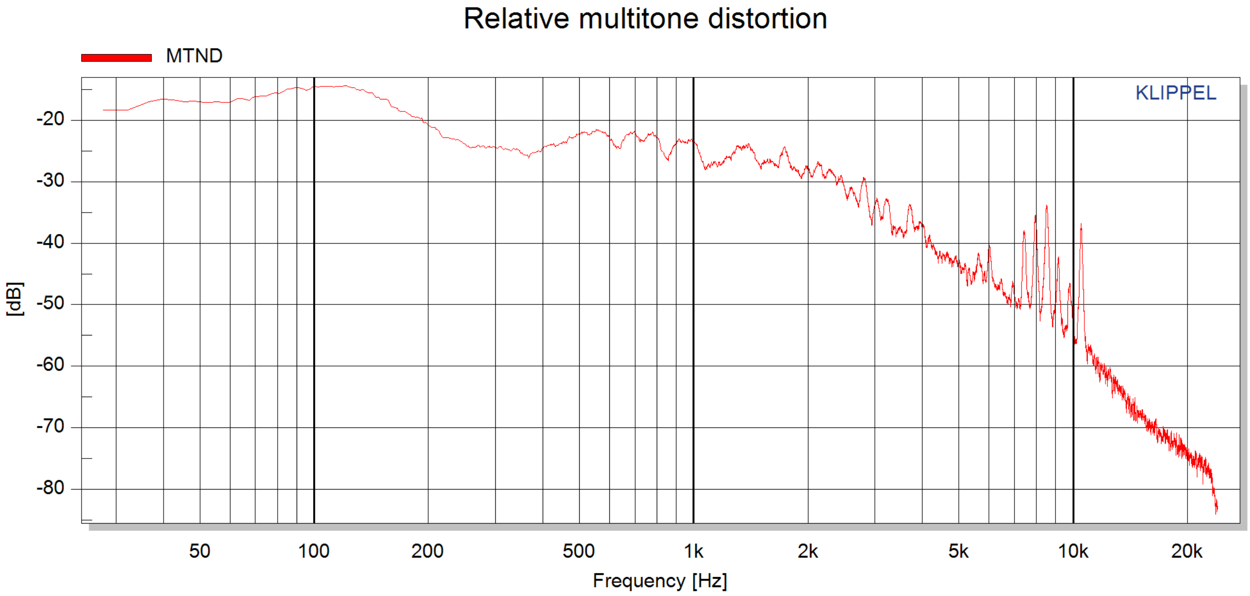
16	Dist. (Umax)	03/31/09	17:56:17	or Umax Reference: See further Information in application note AN16.

Linear Parameter Measurement (LPM)

Multi-tone Distortion Measurement

acoustical near field transfer function and harmonic distortion measurement at high level with audio like multi-tone stimulus





Please find more information in Application Note 16 (09/2008)

Number	Name	Date	Time	Comment
17	8 TRF Harmonics in current	03/31/09	17:56:54	Target: Equivalent harmonic input distortion based on current measurement (only if L(i) dominant) Adjust setup: Voltage in PP Stimulus Procedure: Copy fundamental form window

				FUNDAMENTAL+HARM. and paste into Reference curve on PP Processing.

Transfer Function Module (TRF)

Equivalent Input Distortion

What is equivalent input distortion ?

Traditional techniques for distortion measurement as defined in the IEC standard determine the contamination of the output signal by spectral components which are not in the input signal but are generated by transducer nonlinearities. The measured distortion depend not only on the nonlinear mechanisms but also on the linear properties of the transducer, radiation aids (enclosure, horn), the acoustical environment (room) and sensor (microphone, laser) used. The results are usually difficult to interpret and do not reveal the physical causes.

Assessing the distortion at the source result in more meaningful data describing the large signal behavior of loudspeakers independent of the linear transfer response. Since the distortion source is not accessible for direct measurements equivalent input distortion are calculated from sound pressure output. This technique makes it possible to predict the distortion in the sound field, to investigate the influence of the acoustical environment (room) and to separate noise and other disturbances not generated by the transducer.

The new concept of equivalent input distortion as presented in this paper exploits valuable information from loudspeaker modeling, because most of the dominant nonlinearities are located in the electrical and mechanical part and can be concentrated in a distortion source adding distortion to the loudspeaker input. If the following signal path to any point in the sound field is sufficiently linear then there is a direct relationship between the distortion measured in the sound field and the equivalent input distortion. Using a linear filter with the inverse linear transfer response $H(s,\mathbf{r})^{-1}$, this path can be equalized and the equivalent input distortion may be derived from distortion measured in the sound pressure signal. This technique has been applied to harmonic distortion and intermodulation distortion according to IEC 60268 but it is also useful for any kind of distortion measurement.

Advantages of measuring equivalent input distortion

The redundancy of the measured data can substantially be reduced and the results more easily be interpreted. A loudspeaker may be described by two sets of data: The linear set comprises an amplitude and phase response $H(s,\mathbf{r}_i)$ for each point $\mathbf{r}_i$ in the sound field. The nonlinear set comprises one frequency response U_n for each order n (with $n \geq 1$) of the equivalent input distortion. Thus, the equivalent input distortion describes the large signal performance of the loudspeaker by referring to the single-input of the loudspeaker. This techniques dispenses with measurements nonlinear distortion at all points r_i in the sound field.

The linear overall transfer behavior does not appear in the equivalent input distortion. Only the spectral properties of the state signals (displacement, velocity, current) which are directly related with the generation of the distortion u_d are considered. For the dominant nonlinearities the relationship between parameters, state signals and distortion components is derived in the appendix.

The measured equivalent input distortion are independent of the superposition of the direct sound and diffuse sound at the point r_i and of properties of the room.

The linear frequency response of the sensor used for the measurement of the loudspeaker output has no influence on the calculated equivalent input distortion. Thus, a precise or inexpensive microphone or even a laser measuring displacement, velocity or acceleration of the cone will result in the same equivalent distortion as long as the sensor behaves linearly, there is sufficient signal to noise ratio and the loudspeaker nonlinearities are only located in the one-dimensional signal path.

The equivalent fundamental U_1 compared with the input voltage U reveals the nonlinear amplitude compression C. This value shows the limits of the motor and suspension at low frequencies.

The equalization of the measured sound pressure signal prior to the spectral analysis speeds up the measurement of harmonic distortion because higher sweeping speeds of the sinusoidal stimulus may be used.

If the equivalent input distortion and the linear transfer response $H(s,\mathbf{r}_i)$ are known Eq. (21) allows to predict the distortion in the sound pressure at any point $\mathbf{r}_i$.

The comparison of predicted and measured distortion reveals noise and disturbances caused by parasitic vibrations (enclosure, cars) separated from loudspeaker distortion.

If the equivalent distortion curves derived from sound pressure measurements at different locations (at sufficient signal to noise ratio) does not agree then the deviation reveal dominant nonlinearities in the multi-dimensional signal path which can not be mapped to the same equivalent input distortion in the one-dimensional path. Thus, nonlinearities in the cone vibration, radiation and acoustical system can be recognized and distinguished from loudspeaker's motor and suspension nonlinearities.

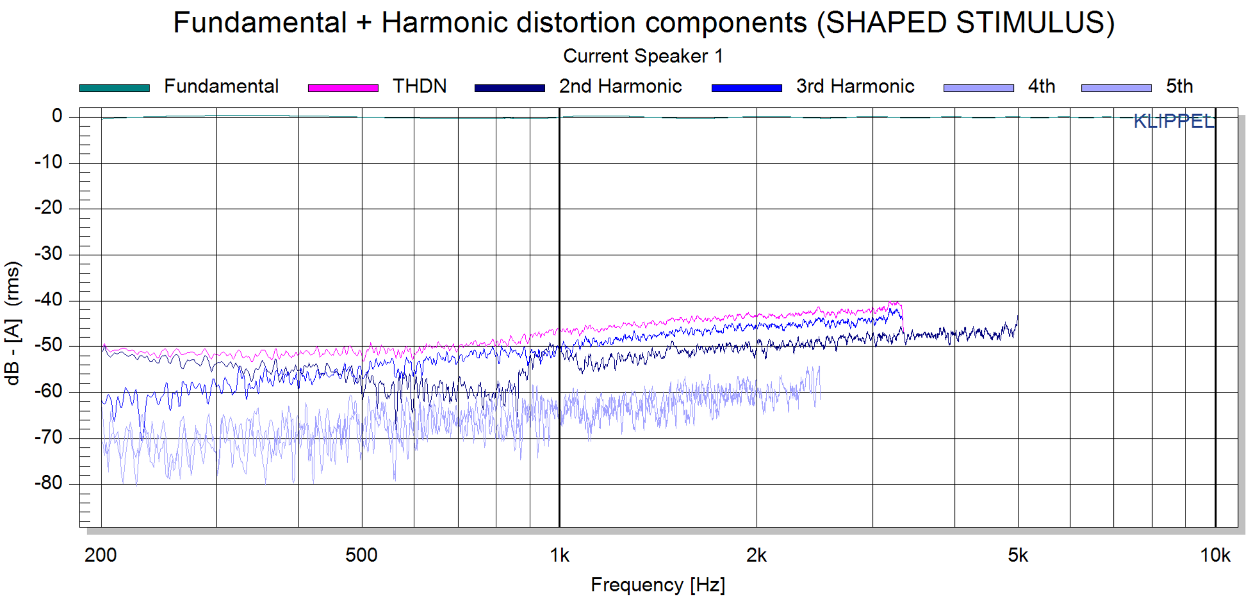
The concept of equivalent distortion is very convenient for assessing the large signal performance of active loudspeaker systems. For example, if the amplifier is integrated in the loudspeaker system the equivalent distortion may be calculated in the input signal of the amplifier.

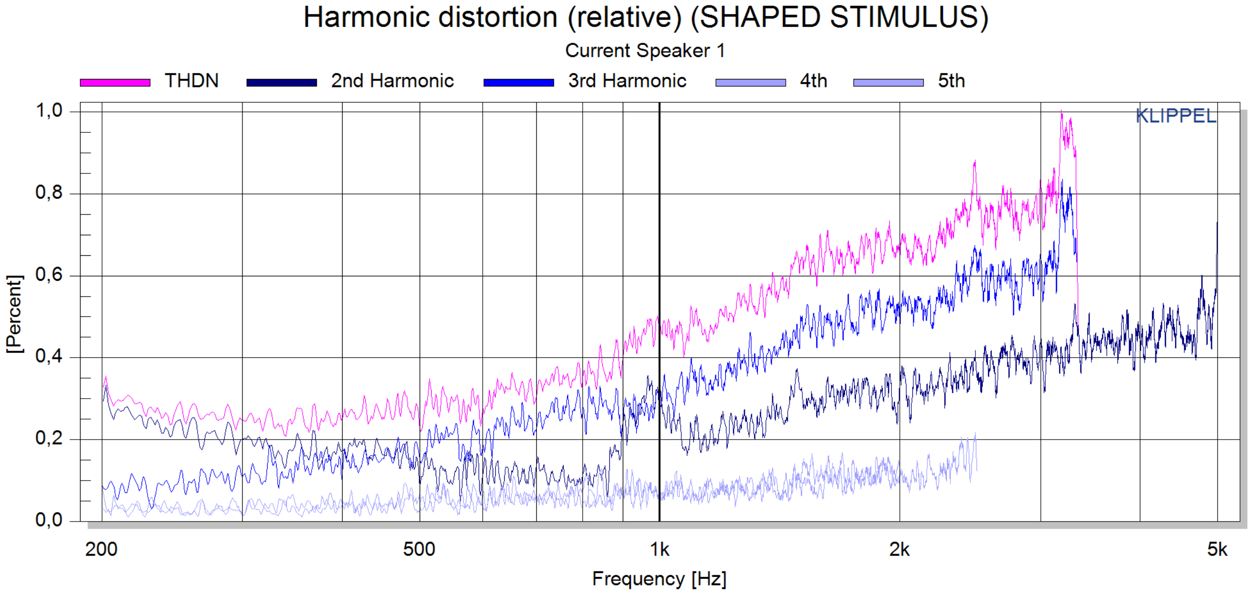
References

1. W. Klippel, "Equivalent Input Distortion," *J. Audio Eng. Society* **52**, No. 9 pp. 931-947 (2004 Sept.).

2. "Measurement of Equivalent Input Distortion," Application Note #20 to the KLIPPEL Analyzer System, www.klippel.de

RESULTS





Performing an inverse filtering of the loudspeaker output signal with the inverse linear transfer function $H(s)^{-1}$ gives an almost flat response of the fundamental response and the equivalent input distortion.

Please find more information in Application Note 20 (09/2008)

Number	Name	Date	Time	Comment
18	9 AUR auralization	02/12/09	16:15:48	Target: Make linear and nonliear distortions separately audible. Adjust setup: - Import paramter from LSI - use PP Auralization as mixer

Report generated:
Date: 06/20/11
Time: 16:20:40
Username: heuschmidt