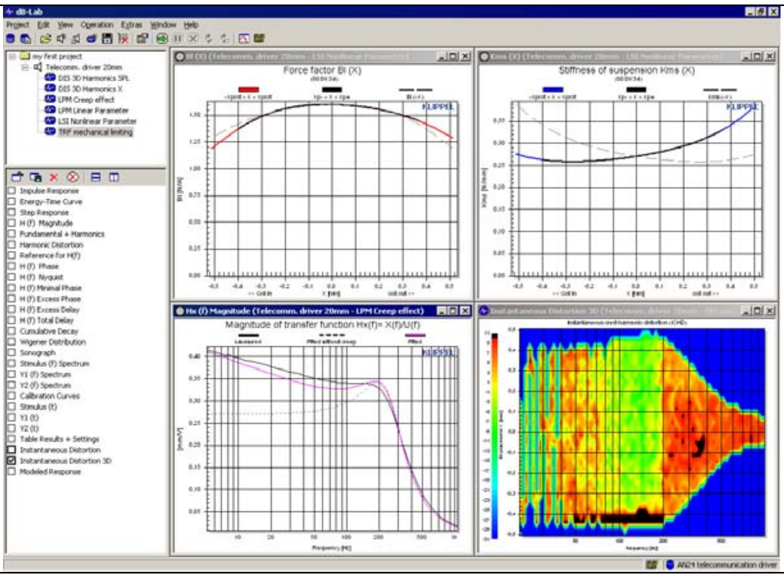
Measuring Telecommunication Drivers **AN 24**

Application Note to the KLIPPEL R&D SYSTEM

The Application Note is dedicated to particular properties of measuring micro-speakers, headphones and other actuators used in headsets, mobiles and other telecommunication devices. These drivers are designed to be operated over a wide frequency range with highest efficiency to save battery power. The small dimensions as well as low level state signals such as voltage or excursion require special measurement setups. Furthermore practical tips for mounting and recommended hardware components are given.



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Particularities of telecommunication drivers

Typical applications	Telecommunication drivers are designed for broadband reproduction of sound in mobiles, headsets, telephones and free-speech devices. Since battery power is crucial in mobile systems, the efficiency must be as high as possible. Therefore the moving mass is usually very small, causing a high resonance frequency f_s. Most micro-speakers, headphones and microphones use no additional spider as suspension.
Filtering	The sound pressure level SPL decreases with 12 dB/octave below f_s if the voltage at the terminals is constant. An equalizer or electrical filter may be used to extend the useable bandwidth to lower frequencies and to realize a desired damping of the system. Higher displacement usually generates higher nonlinear distortion due to nonlinear force factor Bl(x) and stiffness K_{ms}(x) varying with displacement x.
Resonance Frequency	The resonance frequency f_s of micro-speakers is usually between 200 Hz and 4 kHz, typically about 800 Hz. Transducers used in Headphones may have a lower resonance frequency 50 … 1000 Hz.
Q-factors	In some micro-speakers the electrical damping is much smaller than the mechanical damping. A relatively small Bl value gives a high value of the electrical loss factor Q_{es} and the high mechanical losses give a relatively low Q_{ms} value. To identify linear and nonlinear parameters properly the *Q*-factor ratio Q_{ms}/Q_{es} must be greater than about 0,1. For lower ratios than 0,1 the back-induced voltage around the resonance frequency is too small for identifying the driver parameter via the electrical input impedance.
Stiffness Nonlinearity	The suspension of the Microspeakers usually provide no reliable mechanical protection of the voice coil at high amplitudes as provided by a spider in a woofer. The geometry of the suspension may also cause a significant asymmetry in the stiffness characteristic K_{ms}(x) which may produce a dc displacement (rectification of the ac signal).
Thermal Parameters	The thermal dynamics of microspeakers can be modeled by an equivalent circuit as presented in [4]. Contrary to woofers a very short times is required for heating up and cooling down the tiny voice coil in microspeakers. The thermal time constants of the voice coil τ_{tv} usually less than 1 s and the time constant of the magnet and frame τ_{tv} is usually not much more than few minutes. If the microspeaker is operated in free air the thermal resistance R_{tv} between coil and magnet is usually smaller than the resistance R_{tm} between magnet and ambience. Thus the power handling of a microspeaker can significantly be improved if the magnet or frame of the microspeaker is mounted on metal or other material with good thermal conductivity. The forced air convection cooling is much less effective than in woofers. The direct heat transfer due to eddy currents which heat the pole tips directly is also negligible.
Creep effect	The suspension system exposed to a sustained constant force will show a slow increase of the displacement versus time (creep effect, see [1]). Due to visco-elastic effects in the material the stiffness of the suspension is not constant but becomes smaller at lower frequencies. Most telecommunication drivers show substantial creep. Loosing 50% of the stiffness at very low frequencies related to the stiffness at f_s is quite normal. High creep factor may affect the accuracy of the identified mechanical parameters because the measured impedance is higher for $f < f_s$ than predicted by conventional lumped parameter models. The creep model implemented in the LPM can partly compensate for this effect and gives a better agreement between model and real transducer. It is recommended to check the fitting of the transfer function H_x(f) between voltage and displacement and the fitting of the electrical impedance. While the visco-elastic effect in woofers is mainly caused by the fiber structure and the impregnation of the spider and surround the creep factor in microspeakers can also be caused by the air flow in the cavities below the diaphragm behaving like a vented system. The lack of DC stiffness result in an higher DC displacement produced by a given asymmetrical nonlinearity (*Bl(x), Cms(x)*) of the driver. The DC displacement shifts the coil away from the rest position and can increase harmonic and intermodulation distortion.

Air load	The air causes some additional losses, behaves as an additional air spring and increases the total mass. This effect depends on the measurement condition (vacuum, free air, coupler). A microspeaker measured in free air, mounted on a coupler or measured in vacuum gives different values of moving mass M_{ms}, mechanical resistance R_{ms} and compliance C_{ms}. Since the mechanical loss factor Q_{ms} dominates the total loss factor Q_{ts} the losses caused in the mechanical system and by the air have may have an significant influence on the total transfer behavior. The resistance $R_{ms}(x,v)$ also depend on displacement x and velocity v and may produce some nonlinear distortion. While the dependency of $R_m(x)$ versus x is mainly caused by the mechanical system the dependency $R_m(v)$ versus velocity v is mainly caused by turbulent air flow. Note the large identification module (LSI tweeter and LSI woofer) assume a constant resistance R_{ms}. The influence of the mass of the moving air is not negligible because the mass of the diaphragm and the coil is relatively small. The additional stiffness caused by an air volume in a test enclosure (coupler) or in cavities below the diaphragm depends on the effective volume and the effective radiation area S_D which may deviate from the projected area.
Measurement in Vacuum	 A measurement of the microspeakers in vacuum makes it possible to suppress the effect of the air load and to measure the mechanical properties of the microspeaker more precisely. Klippel provides a vacuum measurement kit [23] which provides the electrical stimulus to the speaker and gives access to the laser sensor.
Effective Radiation area	Most micro-speakers use the rim zone of the diaphragm as the suspension. This area is relatively large and there is a gradual transfer from fixed clamping at the outside rim to the point where the coil is attached. Thus the effective radiation area S_D does not correspond with the projected area of the diaphragm.
Inductance	Ideal Inductance (contributing to the imaginary part) and losses due to eddy currents (contributing to the real part) increase the electrical impedance at higher frequency above resonance frequency f_s. In microspeakers is the inductance usually very small and the para-inductance (the lossy part due to eddy currents) is negligible. The nonlinear variation of the voice coil inductance versus displacement x and current i is also negligible.
Optical properties of the diaphragm	The surface of the diaphragms used in micro-speakers are usually shiny or transparent. Laser measurements of the voice coil displacement require additional coating of the surface by applying ink (less recommended) or a light power sprayed on the surface (used for mechanical crack finding).
Break up modes	The first bending modes which break up on the diaphragm are usually circumferential modes followed by radial modes at very high frequencies. More details see [6]
Rocking modes	Micro-speakers use a single suspension (no spider) and are prone to rocking modes. It is recommended to measure the mechanical cone vibration at multiple points and average the results to get a meaningful average value for the voice coil displacement.

Preparations of the Measurements

Requirements	The following hardware and software is required: - Distortion Analyzer Rev. 1.2 or higher (sensitive Channel 2) - Laser stand or Laser Scanning Hardware - High resolution Laser Sensor (G32), Microphone - Software modules (LPM, LSI, DIS, TRF-Pro, dB-Lab, SCN, SIM, AUR)
General Setup	- Attach the driver in a fixture such a coupler, or a clamp. Plaster (easy to dismount) or double sided adhesive tape (difficult to dismount) are less recommended. Pay attention that you don't close any holes at the rear side of the microspeaker. - Use high sensitive *Speaker Channel 2* (for DA 1.2 and higher) for LPM, LSI, TRF, DIS, SCN measurements. - Solder short wires to the terminals since the clamps provided with the Analyzer are too big and too heavy for direct fastening. - Pay attention to the heating of the voice coil. Averaging (LPM) or additional excitation time (DIS) may heat the voice coil and increase the voice coil resistance considerably. - Place the microphone in the near-field of the driver for acoustical measurements. For small drivers it may be required to remove the laser sensor to adjust the microphone just on axis to the driver. - Create a new object using the object template setup *DIAGNOSTICS MCROSPEAKER SP1* as provided by KLIPPEL. - The excitation level should be adjusted according to the mechanical capabilities of the driver under test (for LSI and DIS). Find mechanical limits with the TRF-Pro rub& buzz detection. Read peak to peak displacement and ensure that in the large signal tests (LSI, DIS) this range is not exceeded.

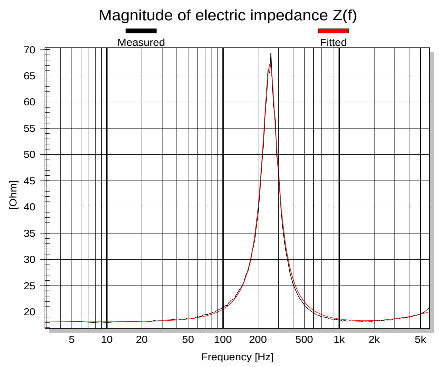
Measurement of Small Signal Parameters (T/S)

using Linear Parameter Measurement Module (LPM)

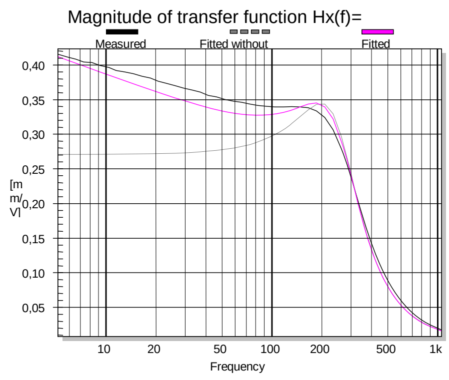
Diagnostic value for microspeakers	The small signal parameters describe the transfer behavior of the microspeaker at low amplitudes. Voice coil dc resistance R_e and force factor Bl(x=0) at the rest position x=0 give the driving force for a given voltage applied to the terminals. The stiffness K_{ms}(x=0) at the rest position x=0 determines the displacement for low frequencies ($f<f_s$) while the moving mass M_{ms} for high frequencies ($f>f_s$) . At the resonance frequency f_s the electrical loss factor Q_{es} and the mechanical loss factor Q_{ms} determine the total Q factor and the shape of the resonance curve.
Optimal Method	The Linear Parameter Measurement (LPM) provides alternative methods for measuring microspeakers: **<u>Single step measurement (Impedance + displacement by laser):</u>** This technique is recommended for microspeakers because it can be applied to microspeakers mounted on couplers, operated in free air and in vacuum. **<u>Two-step measurement with enclosure perturbation (impedance):</u>** This technique requires precise values of the effective radiation area S_D and of the air volume of the test enclosure. It is recommend to use a differential technique and to generate a defined variation of the air volume by using a medical injection pump. The accurate measurement of the effective radiation area S_D also requires a laser sensor (see application note [7]) **<u>Two-step measurement with added mass perturbation (Impedance):</u>** This technique is usually not applicable to microspeakers.

Optimal Setup	Use the object template “Diagnostics MICROSPEAKER Sp2” containing the operation “1 LPM Small signal parameter”. Ensure that the maximal frequency F_{max} is set to 6 kHz. Activate the noise floor measurement and adjust the voltage (50 mV). Activate on property page INPUT the routing to speaker 2. This selection of an optimal inductance model (on property page IMPORT/EXPORT of LPM) is not critical.
Laser usage	Optical measurements with triangulation lasers give very accurate results if the following points are considered: **<u>Use a high resolution, high bandwidth laser</u>** The recommended G32 laser has an upper frequency > 10 kHz and a good SNR ratio. **<u>Laser Calibration:</u>** A laser check (see manual) should be performed to ensure that the laser is properly calibrated. **<u>Remove the grill</u>** The grill, cloth or any kind of cover on the diaphragm has to be removed to give the laser access to the diaphragm. This “operation” may change the properties of the device under test if the mechanical system or the acoustical load is changed. It is recommended to measure the linear parameters before and after grill removal and to check for a significant parameter shift. **<u>Diffuse Target surface :</u>** Triangulation lasers might produce errors in the measured displacement if the target surface is transparent or highly reflective. Analogue laser sensors might produce a smaller output which results in a higher Bl value. Digital laser sensors might produce distortion in the laser output (drop outs) and large variation in the Bl value. White coating of the surface is strongly recommended to get a diffuse reflection. A simple and good way is to use white spray which is used for finding cracks in materials. White correction fluid is not recommended. Use a minimum of paint ! To prevent any error due to additional mass, it is recommended to measure the resonance frequency before and after painting. If no shift is measurable, the influence on the measurement due to the additional mass is negligible. If a shift of the resonance frequency is found, the additional mass may subtracted from the total mass. **<u>Average laser measurements</u>** Measure the vibration of multiple points on the diaphragm and average the results to compensate for rocking modes and optical errors. An automatic averaging can be realized by Laser scanning vibrometry [6]. **<u>Consider the glass plate in vacuum measurements</u>** A triangulation laser can be used to measure the voice coil displacement while the micro-speaker is operated in vacuum. Note that the calibration factor used for a measurement in free air can not be applied. A separate laser calibration is required if a glass plate is used between target and laser sensor.

Example Microspeaker



This driver has a high impedance peak, indicating sufficient back induced voltage for parameter identification both in LPM and in LSI.



The Creep effect has to be considered in the measurement of the linear parameters to reduce the fitting error in the displacement transfer function $H_x(f)$.

Check the LPM measurement

Klippel LPM measurement provides different ways[8] for checking the accuracy of the LPM measurements:

Check the fitting error of the impedance curve

- The deviation between measured and modeled impedance response is summarized in an fitting error rmse Z. Significant deviation can be caused by a noisy current and voltage measurement or by a discrepancies in the model used.
- High noise in the current measurement may cause a high failure in the estimated dc resistance R_e. Remedy: Increase voltage and averaging.
- Deviation below resonance frequency f_s are usually caused by visco-elastic effects and air load (cavities). Remedy: Perform measurement in vacuum and use creep model.
- Deviation above resonance may be caused by acoustical load. Remedy: Perform a measurement in vacuum.

Ensure 20dB SNR+D in Current(f).

The fitting of the impedance curve at the resonance frequency is important for estimating the loss factors. Note: 20 dB SNR+D (Signal to Noise + Distortion Ratio) correspond to 10% accuracy of parameter accuracy for Bl and 20% accuracy for M_{ms} and C_{ms}. Higher SNR+D improve the accuracy. Refer to AN25 for optimizing LPM accuracy [8].

Remedy: Increase average to obtain lower noise floor. Reduce amplitude of the excitation signal if nonlinear distortion cause low SNR

Check the fitting error of the displacement curve

The deviation between measured and modeled displacement response $H_x(f)$ is summarized in an fitting error rmse Hx. Significant deviation can be caused by insufficient reflection from the target surface, reflections of a shiny surface or glass plate of the vacuum chamber or visco-elastic behavior (creep).

Remedy: Check the proper usage of the laser. Perform a measurement in vacuum.

Measurement of Effective Radiation Area S_D

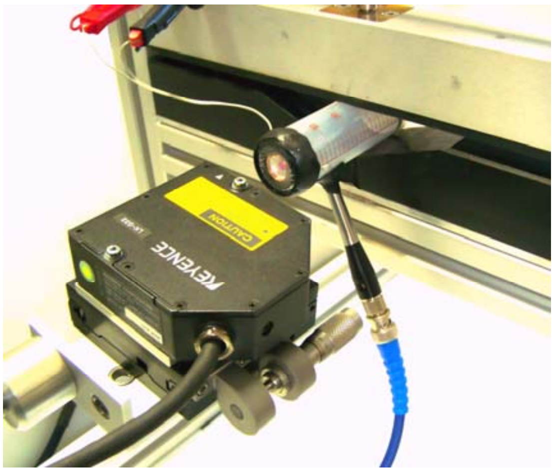
using application note [7]

Diagnostic value for microspeakers

The effective radiation area S_D is required for predicting the sound pressure output for a given voice coil displacement. Thus the sensitivity and the maximal output can only be calculated precisely if the parameter S_D is correct.

Optimal Method

The effective radiation area S_D can be determined by mounting the transducer to an enclosure with variable size (medical injection pump) and measuring the sound pressure in the enclosure as well as the displacement of the voice coil by using a laser triangulation sensor. A differential method as implemented in a special MAT module provided by KLIPPEL can be used to calculate the S_D value.



Measurement of Large Signal Parameters

using Large Signal Identification Module (LSI)

Diagnostic value for microspeakers

The force factor Bl(x) and stiffness K_{ms}(x) are important large signal parameters for microspeaker. They show the physical cause of the dominant harmonic and intermodulation distortion (more details in the tutorial [2])

BL(x) nonlinearity: The ratio between voice coil height and gap depth determines the symmetrical decay of the Bl(x) curve for positive and negative displacement. An asymmetry of the Bl(x) curve may be caused by an asymmetrical B field in the magnetic gap and an offset in the rest position of the coil.

K_{ms}(x) nonlinearity: High positive or negative excursion cause a deformation of the corrugation roll in the suspension (which is in microspeakers part of the diaphragm). This causes a symmetrical increase of the stiffness. An asymmetrical shape of the K_{ms}(x) is mainly caused by an asymmetrical geometry of the corrugation roll.

Thermal Parameters: The ratio between thermal resistance R_{tv} and R_{tm} reveals the bottleneck in the heat transfer.

Selection of the optimal module

The resonance frequency f_s must be lower than 400 Hz to perform a large signal identification using LSI Woofer. The LSI Tweeter can measure microspeakers with a higher resonance frequency (100 Hz < f_s < 2 kHz) but the measurement time is much longer.

NOTE: The LSI modules in the R&D system are relatively slow measurement of the nonlinear parameters. Single-valued large signal parameters such as

- voice coil position in mm
- stiffness asymmetry in percent
- maximal peak displacement X_{Bl} limited by force factor Bl(x)
- maximal peak displacement X_C limited by compliance
- and the maximal peak displacement limits X_{Bl}

can be measured in less than a seconds by the Motor and Suspension Check (MSC) of the QC System [22].

Preparing a measurement	Use the object template “Diagnostics MICROSPEAKER Sp2” containing the operation “2 LSI Nonlin. Parameter only”. **Reduce initial excitation level.** Set the external Power Amplifier gain to about 6-20 dB and the internal gain for small signal parameters G_{small} to –20 dB on the Property Page Protection within the LSI module. This ensures that the small drivers are operated in the safe, linear domain at the beginning of the Large signal Identification (LSI). In case the error “*Minimal output gain*” pops up, reduce the power amplifier gain. **Limit input power for protection**. Telecommunication driver have not a suspension and a motor structure which provides natural protection against mechanical overload and damage. The Bl_{min} and C_{min} parameters are not suited for input power limiting. Voice coil rubbing and hard limiting may damage the speaker before the regular nonlinearities (Bl(x) and C_{ms}(x) will limit the excursion. Set the Protection Parameter P_{lim} at the Property Page Protection to 0.05 W or lower. If the final drive level is too small, increase P_{lim} step by step. **Select the SPEAKER 2 terminal.** The Distortion Analyzer provides on SPEAKER 2 a more sensitive current sensor which is required for measuring micro-speakers. **Use Stimulus with sufficient bandwidth.** The identification of the nonlinear parameters require a noise signal exciting the microspeaker below and above resonance frequency f_s. Switch on the automatic noise button on Property Page “Generator” to use a stimulus with properties optimal for the speaker. **Operate the microspeaker in vacuum** to reduce the influence of the acoustical load. **Mount** the microspeaker on a small stand and adjust the **Laser to the diaphragm.** The laser is not required for the identification of the linear and nonlinear parameters but is very useful to check the polarity of the microspeaker. Note the polarity of the speaker and the way how the SPEAKER cable is connected to the terminals of the microspeaker because the orientation will determine the orientation of the nonlinear curves. If the cables are connected correctly then a label COIL IN will be plotted at negative displacement.
Performing the measurement	**Start** the LSI measurement. **Check** the identified resonance frequency after finishing the LINEAR MODE. IF the resonance frequency is significantly different from the resonance frequency found in LPM then repeat the measurement with increased initial excitation level. **Wait** until the adaptive algorithm is converged to the optimal parameters. Select at least 5 min in the NONLINEAR MODE on the property Page CONDITION for LSI Woofer (20 min for LSI TWEETER). **Listen to rub and buzz**. If you hear impulsive distortion during LSI measurement then reduce the limit value MAXIMAL INPUT POWER P_{lim} on the property page PROTECTION. The system will go back to the enlargement mode and will reduce the amplitude of the stimulus automatically. **Import the value Bl(x=0) at rest position x=0**. Export all parameters of the Linear Parameter Measurement LPM to the clipboard on Property Page IMPORT/EXPORT and import those values on property page IM/EXPORT of the LSI. The imported Bl value is much more precise than the Bl value determined by the laser in the LSI.
Accuracy of the measurement	Open the result window Bl(x) and check that the COIL IN position is printed at negative displacement. Open the result window STATE and check the fitting error ei(t) which should be below 15 %. If the error is higher the assumed model does not fit with the real microspeaker. Remedy: Perform the measurement in vacuum to reduce the influence of the acoustical load. Use the current sensor at Speaker 2 to ensure high SNR.

Results

Open the result window BL(x)

- symmetrical decay of the force factor Bl(x) correspond with the voice coil height and gap depth and the fringe field
- asymmetry of the Bl curve indicates a voice coil offset and asymmetrical B field in the gap

Open the result window $K_{ms}(x)$

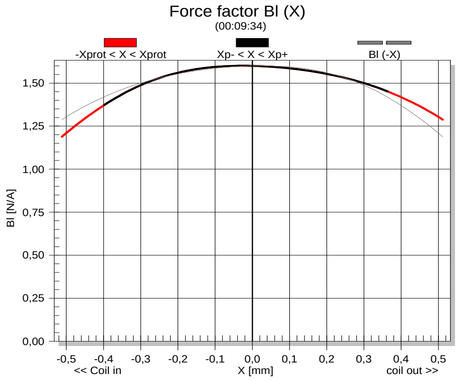
- symmetrical increase of the stiffness $K_{ms}(x)$ related with the size of the corrugation
- asymmetry of the stiffness $K_{ms}(x)$ related with the asymmetry in the geometry of the corrugation causing 2nd order distortion and a dc component
- loss of the stiffness $K_{ms}(x=0)$ at the rest position at high amplitudes due to visco-elastic properties of the suspension

Open the result window DISTORTION and see the separated distortion components generated by force factor and compliance in the sound pressure output.

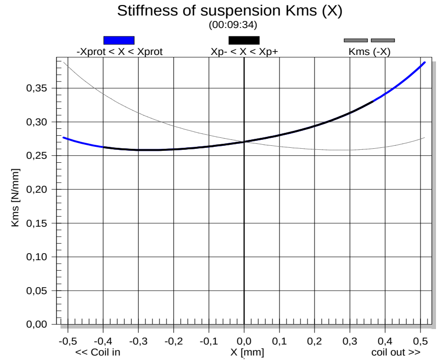
The following information measured in LSI are less important for microspeakers

- The nonlinearities of the **Inductance** L(x) and L(i) of microspeakers is usually very small and hardly measurable in the LSI analysis range up to 1.5 kHz.
- The parameters at the rest position x=0 such as the resonance frequency f_s, total loss factor Q_{ts}, electrical loss factor Q_{es}, and mechanical loss factor Q_{ms} measured at low amplitudes are less accurate than the parameters measured by the LPM which is dedicated for the small signal domain. At high amplitudes the parameters at rest position x=0 show the influence of visco-elastic effects, dc components and voice coil temperature.

Example Microspeaker



This driver has a good symmetrical Bl-curve. Thus the 2nd-order motor symmetry.



The stiffness (inverse of compliance) has a significant asymmetry. This causes not only a 2nd order harmonic distortion in sound pressure ($f \leq f_s$) but generates also a dc component X_{dc} in displacement directed to negative displacement (where the stiffness is lower).

The maximal voice coil excursion [-0.5mm … 0.5mm] was limited by maximal input power P_{max} = 0.05 W. The regular nonlinearities $Bl(x)$ and $C_{ms}(x)$ *stayed above the allowed* limit values of Bl_{min} and C_{min} (set to 50%). However, at a peak displacement of 0.5 mm impulsive distortion (see TRF) have been generated which indicate the limits of the usable working range.

Measurement of Large Signal Performance

using 3D Distortion Module (DIS)

Diagnostic value for microspeakers

The 3D Distortion Measurement module (DIS) is a perfect tool for assessing the large signal performance of microspeakers by performing a steady-state measurement with a single tone or two-tone stimulus. This measurement reveals problems caused by motor structure and suspension. The following characteristics are useful:

- Peak displacement versus frequency and voltage [17]
- Nonlinear and thermal compression of the fundamental amplitude in SPL and Displacement versus varying input voltage [16]
- dc displacement due to nonlinear rectification processes [17]
- intermodulation distortion in sound pressure output [13]
- harmonic distortion versus voltage and frequency [14]

Temperature protection. All DIS measurements should be performed with automatic temperature protection. Enable the Temperature Monitoring at Property Page Protection and set the temperature limit according to your driver. Select the monitor tone at least two octaves away from the resonance frequency f_s. The default value is set to 2250 Hz which is good for f_s<900 Hz. For f_s >900 Hz use a test tone of 375 Hz

Displacement

Use the template operation 5 DIS X FUNDAMENTAL, Xdc in the template Diagnostics MICROSPEAKER Sp2 . Set the starting frequency f_{start}= 0.1 f_s and end frequency f_{end}= 10 f_s of the single-tone stimulus and the maximal voltage U_{end}. Adjust the laser sensor to the voice coil.

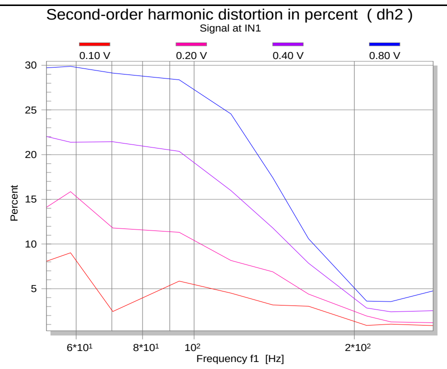
Open result window "**Fundamental**" to see the rms value of the displacement versus frequency and voltage. The change of the resonance peak reveals the changing total loss factor Q_{ts} caused by nonlinearities and visco-elastic effects.

Open the result window "**Peak – Bottom**" to see the positive and negative peak displacement versus frequency and voltage. This window also shows a dc in the displacement which may be produced by rectification of the ac signal by asymmetries in Bl(x) and K_{ms}(x).

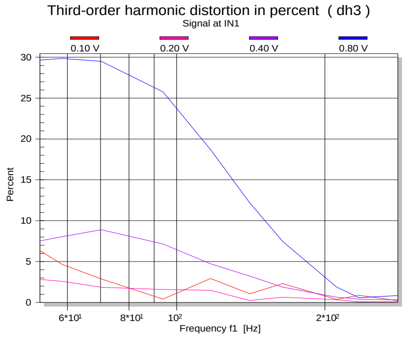
Open the result window **"DC displacement"** to see the dc component separated from the fundamental component. A high dc component causes a dynamic shift of the voice coil which increases the intermodulation distortion and may cause hard limiting of the voice coil displacement (See below TRF measurement of impulsive distortion) .

Open the result window **"Compression"** to see the compression of the fundamental component due to nonlinear and thermal effects. All curves measured at high voltages are referred to the measurement at the lowest voltage U_{start}.. If all curve are identical there is no amplitude compression in the loudspeaker.

Example microspeaker



The asymmetry in the stiffness curve K_{ms}(x) generates cause most of second order distortion below the resonance frequency.



3rd –order harmonic distortion are generated by the symmetrical decay of Bl(x) and symmetrical increase of K_{ms}(x). Suddenly at 0.8V the 3rd order distortion comes up due a hard limiting effect of the K_{ms}(x).

Harmonics

Harmonic distortion are measured versus voltage and frequency in the sound pressure output by using the results of 6 DIS IMD bass sweep and 6 DIS IMD voice sweep.

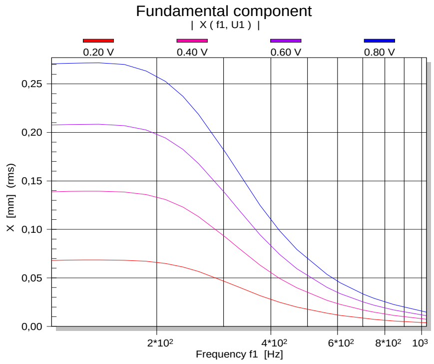
Open the result window **“Harmonics, %”** to see the 2nd harmonic and 3rd harmonic of frequency f_1 varied versus frequency $f_{start} < f_1 < f_{end}$ at maximal voltage U_{end}

Open the result window **“Total Harmonics, %”** to see the THD of frequency f_1 varied versus frequency $f_{start} < f_1 < f_{end}$ and voltage.

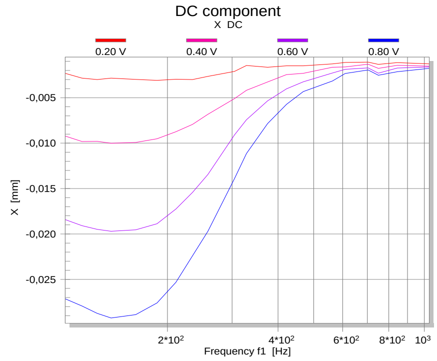
Open the result window **“2nd Harmonic, %”** to see the 2nd order distortion related to the total output of frequency f_1 varied versus frequency $f_{start} < f_1 < f_{end}$ and voltage.

Open the result window **“3rd Harmonic, %”** to see the 3rd order distortion related to the total output of frequency f_1 varied versus frequency $f_{start} < f_1 < f_{end}$ and voltage.

Example Microspeaker



The fundamental of the excursion shows the typical roll off of 12 dB / octave above f_s. Compression effects are hardly detectable. See also the Compression window in DIS.



The measured DC part is generated by the asymmetry in the stiffness curve $K_{ms}(x)$. This DC part shifts the working point dynamically to negative displacement where the stiffness is lower.

Intermodulation

Intermodulation distortion are measured versus voltage and frequency in the sound pressure output by using the template 6 DIS IMD bass sweep and 6 DIS IMD voice sweep. Select the mode “harmonic + intermodulations (f2) to evaluate the intermodulation about f_2. The 6 DIS IMD bass sweep keeps the higher frequency f_2 constant at f_2= 5 f_s while the lower frequency component (bass tone) is varied versus frequency $f_s/10 < f_1 < f_s$. Select

The alternative measurement 6 DIS IMD voice sweep keeps the lower frequency f_2 constant at f_2= 0.1 f_s while the higher frequency component (bass tone) is varied versus frequency $f_s/10 < f_1 < f_s$. Select the mode “harmonic + intermodulations (f1) to evaluate the intermodulation about f_1.

Open the result window **“2nd Intermod, %”** to see the 2nd order intermodulation versus frequency $_{start} < f_1 < f_{end}$ at various amplitudes.

Open the result window **“3rd Intermod, %”** to see the 3rd order intermodulation versus frequency $_{start} < f_1 < f_{end}$ at various amplitudes.

Intermodulation measurements provides unique symptoms for identifying Bl nonlinearity.

Please find more to loudspeaker diagnostics in the in the paper “Loudspeaker Nonlinearities – Cause, Parameter, Symptoms” [2]

Sound Pressure Frequency Response

using Transfer Function Module (TRF)

Diagnostic value for microspeakers	The TRF module is a convenient tool for measuring the linear transfer function, applying windowing to the impulse response and providing amplitude and phase response of the fundamental. Nonlinear distortion and ambient noise may corrupt the measurement of the linear transfer function. The TRF module provides means for checking the operation in small signal domain where the nonlinearities can be neglected and a sufficient signal to noise ratio (SNR) can be realized.
Performing a measurement	Use the operation template 3 TRF SPL + harmonics in object “Diagnost. Rub & Buzz Sp2” which uses the SPEAKER 2 connector. Enter the maximal voltage generated at the terminals of the speaker. Use shaping of the stimulus to avoid high peak displacement at low frequencies or to realize an optimal SNR. Select on property page “Processing” and select Half Hanning for windowing the impulse response. Use the time cursor in result window “Impulse Response” to gate the meaningful impulse response without room reflection. Open the result window “H(f) Magnitude” to see the amplitude response. Select on property page “Display” the checkbox “Show phase without time delay” and open result window “H(f) Phase”.
Low Noise ?	Noise from ambience or generated by the microphone may corrupt the measurement. ▪ Activate “Noise floor + dc monitoring” in property page Stimulus” before doing the measurement ▪ Open the result window “Y(f) Spectrum” to compare the noise floor with the fundamental components and to see the SNR for each frequency
Low Distortion ?	The TRF module provides always the 2^{nd}-,3^{rd}- and higher-order distortion in the measured signals by exploiting properties of the sinusoidal sweep with logarithmic sweep speed. To see the harmonics open the property page “Processing” and select under “Transfer function” stimulus “STIM” as denominator. Ensure that the harmonic distortion are sufficiently low when measuring the linear amplitude and phase response. Open the result window “Fundamental + Harmonics” to find the low-order harmonic distortion and the THD on an absolute scale in dB. Open the result window “Harmonic Distortion” to see relative distortion measures referred to the fundamental response in percent and dB depending on the setup selected on property page “DISPLAY”.

Equivalent Harmonic Input Distortion

using the Transfer Function Module (TRF)

Diagnostic value for microspeakers	Harmonic distortion measured in the sound pressure output signal reveal not only the nonlinear properties of the loudspeaker but also the acoustical properties of the acoustical environment. Those acoustical properties cause a linear filtering (post-shaping) of the nonlinear distortion generated in the motor structure and diaphragm of the microspeaker. This effect can be removed by a applying an inverse filtering of the harmonic distortion with the fundamental response and provides the equivalent input distortion (see AES paper [3]). The equivalent input distortion are independent on the linear properties of the microphone, the position of the microphone, and diffraction and reflection of the sound wave at the baffle.

How to do it ?	After windowing the linear impulse response properly (see above) open the result window Fundamental + Harmonics and copy the curve "Fundamental" into the clipboard. Open the property page Processing and export the curve into "Reference". The reference curve is applied as an inverse filter prior to the spectral analysis giving a almost flat fundamental response and the equivalent harmonic input distortion. Find more practical tips in application note [19]

Impulsive Distortion (rub & buzz)

using the Transfer Function Module (TRF)

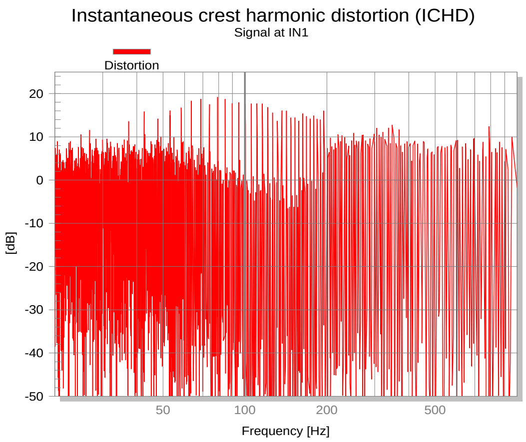
Diagnostic value for microspeakers	The regular nonlinearities such as force factor Bl(X) and nonlinear stiffness $K_{ms}(x)$ of the suspension produce low order harmonic and intermodulation distortion. Other defects in the microspeaker such as hard limiting of the coil at the grill, voice coil rubbing, buzzing mechanical components, loose particles produce impulsive distortion which are audible and have a high impact on perceived sound quality. Performing a spectral analysis we find many high order components (20 …200) which have very low energy. Conventional measurements of THD or assessing the amplitude of the higher-order components gives a low signal to noise ratio. It is more advantages to measure the impulsive distortion in the time domain. Here both amplitude and phase information are exploited and all higher-components generate an impulse in the time domain. Impulsive distortion have a higher crest factor than distortion generated from regular nonlinearities. More details in the paper [2[15].
Measurement	Create a new measurement object based on the template "Diagnost. Rub & Buzz Sp 2". This object contains a series of TRF operations for impulsive distortion measurements where the voltage is increased by 0.1 V steps from 0.1 V to 1 V **automatically**. Ensure that the maximal voltage is permissible for the microspeaker under test. Select the object and press the start button. All operations are processed sequentially. If you hear significant rub and buzz distortion you may cancel the measurement at any time. Alternatively you may start the operation "0 TRF Crest factor harmonics" in the object Diagnost. RUB & BUZZ Sp2. This is a measurement repeated **manually**. Starting at low levels the excitation is increased (not more than 20% per step) until suddenly peaky distortion show up, characterizing the mechanical limiting of the suspension (such as flapping, hitting). Usually these disturbance are also audible quite well. Select on property page "I-DIST" the mode "Rub &Buzz (Order 20) "to consider the higher-order distortion components (order > 20).
Peak Value	The **peak value** of the impulsive distortion is a powerful measure for quantitative assessment on an absolute scale. A limit curve may be applied to derive a PASS/FAIL decision. Select any operation "PHD (peak)" in the automatic batch of object "Diagnost. Rub & Buzz Sp2" and open the result window "Instantaneous Distortion". You see peak value measured in the time domain of the higher order distortion (order > 20). To display the peak value in the manual measurement "0 TRF Crest factor harmonics" select the property page "I-DIST" and select the measure "PHD".
Crest Factor	The **crest factor** of the impulsive distortion is a good measure for finding impulsive distortion. It is the ratio of peak value to rms value of the impulsive distortion. Regular distortion from Bl(x) and $K_{ms}(x)$ have a low crest factor below 12 dB but rub and buzz defects exceed this limit. Select any operation "IHCD" in the automatic batch of object "Diagnost. Rub & Buzz Sp2" or select the manual measurement "0 TRF Crest factor harmonics". Select the measure "CHD" on property page "I-DIST" and open the result window "Instantaneous Distortion". If the crest factor is higher than 12 dB impulsive distortion are generated at the particular frequency.

Instantaneous Crest Factor

The **instantaneous crest factor** is defined as the ratio of the instantaneous distortion to the rms value of the instantaneous distortion. This measure gives shows the fine structure of the impulsive distortion at full temporal resolution. The envelope is identical with the crest factor defined above. However, the instantaneous crest factor plotted versus frequency and voice coil displacement is a very powerful measure for finding the physical causes of the rub and buzz problem.

Select any operation “IHCD” in the automatic batch of object “Diagnost. Rub & Buzz Sp2” or select the manual measurement “0 TRF Crest factor harmonics”. Select the measure “ICHD” on property page “I-DIST” and open the result window “Instantaneous Distortion”.

Example Microspeaker



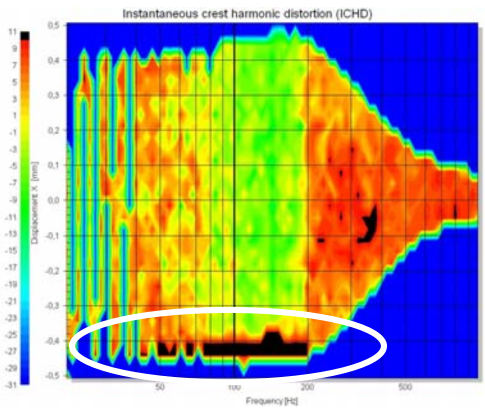
Exciting the driver with a sine sweep of 0.9V suddenly impulsive distortion are generated between 50 Hz and 200 Hz which have a crest factor which is much higher than the crest factor (about 12 dB) found at other frequencies.

Crest factor versus displacement and frequency

To understand the causes of impulsive distortion it is crucial to understand when it happens and where is the coil is located at this particular incident. The TRF module uses the displacement (or acceleration) derived from additional laser sensor used during the distortion measurement. If no laser is available the microphone signal may also be used to plot the crest factor versus acceleration.

Open the result window “Instantaneous distortion 3D” to find the crest factor plotted versus frequency and voice coil displacement. Search for “black dots” which represent impulsive distortion with a crest factor larger 12 dB.

Example Microspeaker

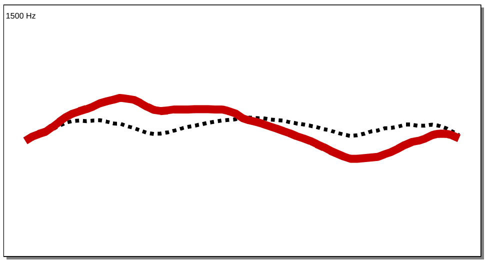
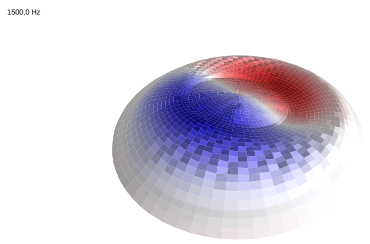


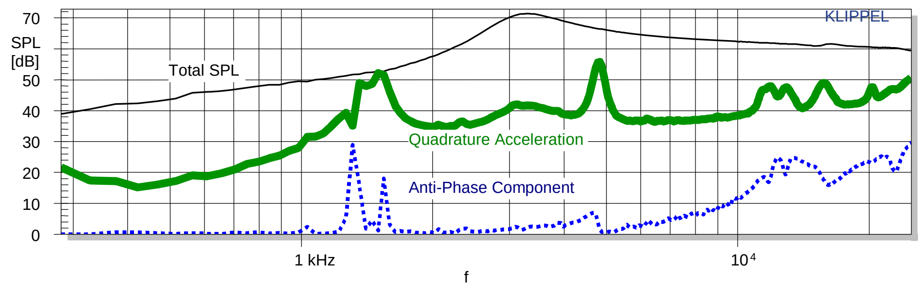
The instantaneous crest factor of the higher-order distortion represented as colour in left figure can be plotted versus frequency (x-axis) and voice coil displacement (y-axis). If the colour becomes black the crest factor is larger than 12 dB indicating impulses with high peaks.

In the microspeaker under test impulsive distortion are generated between 50 Hz and 200 Hz at the negative peak displacement of 0.5 mm. This is caused by a hard limiting of the coil excursion. The impulses above resonance (300 Hz) may be caused by a rocking mode which initiates voice coil rubbing in the gap.

Mechanical Vibration and Radiation Analysis

using the KLIPPEL Scanning Vibrometry (SCN)

Diagnostic value for microspeakers	Scanning the diaphragm and measuring the displacement directly shows modes of the mechanical vibration. Axial-symmetrical modes in radial direction are responsible for generating the sound pressure output on-axis and to the total sound power radiated by the diaphragm. The contribution of the circumferential modes is less due to acoustical cancellation effects. However, low-order circumferential modes may cause a rocking movement and a rubbing of the coil in the gap producing impulsive distortion. This is much more critical in microspeakers because the single suspension system is relative flat and an unbalanced mass or additional force caused by the wires may initiate this mode. Prediction of the circumferential modes by FEA are difficult because the input parameters are not known.
Measurement Procedure	The complete scanning of the diaphragm requires the same preparation of the target surface as discussed in the context of linear and nonlinear parameter measurement. Scanning of sufficient points (50) on the diaphragm can be accomplished in few minutes [6]. It is recommended to perform a scan in free air and in vacuum to separate the influence of the mechanics and the air load.
Example Microspeaker	 Cross-sectional view of the geometry (dotted line) and superimposed vibration (solid line) of a rocking mode  3D view of the geometry and superimposed vibration (blue and red indicates opposite directions) of a rocking mode



The geometry and vibration data measured by laser scanning are valuable parameters of the microspeakers in addition to linear and nonlinear parameters. They are the basis for predicting the sound pressure output at any point in the far field as shown as black solid line (**Total SPL**) in the figure above.

An vibration and radiation analysis [6] may reveal additional components which are the basis for deeper diagnostic:

Total Acceleration level describes the sound pressure potential without any acoustical cancellation effects. This measure is the basis for a modal analysis and reveals Natural Frequencies and modal loss factors which may be used to estimate the material parameters required by Finite Element Analysis.

Peaks in the **Quadrature Acceleration** (green curve) may indicate high circumferential modes. The peak at 1.5 kHz is caused a rocking mode.

The **Anti-phase Component** (blue curve) evaluates the acoustical cancellation effect. IF the Total SPL is more than 15 dB higher than the anti-phase component the in-phase component is always dominant and the acoustical cancellation can be neglected.

Prediction of the Transfer Behavior

using the KLIPPEL Simulation Module (SIM)

Diagnostic value for microspeakers

The small signal and large signal behavior of the micro-speaker can be predicted on the basis of linear, nonlinear and thermal parameters of electro-acoustical model. The SIM-module is a twin of the DIS module supporting a direct comparison of measured and predicted performance [18]. A good agreement verifies the lumped or distributed parameters measured or predicted by FEA. In addition to this the influence of each nonlinearity can be investigated in detail and new design choices can be evaluated. The SIM modules considers an electrical pre-filter or and acoustical post filter and the influence various types of enclosures and acoustical loads and is a powerful tool for system design.

How to do a prediction ?

Create a new SIM operation based on a SIM template which corresponds with the DIS module (for example “SIM X Fundamental, DC”)

Import the large signal parameters on property page IM/Export from the LSI large signal identification.

Import the setup parameter from the corresponding DIS module (for example “DIS X Fundamental, DC “)

Select “Driver in Baffle” on property page “System” or specify the enclosure used.

Start the SIM module.

Select the state signal of interest (displacement, sound pressure, temperature) on property page DISPLAY.

Auralization of Nonlinear Distortion

using the KLIPPEL Simulation Module (SIM)

Diagnostic value for microspeakers	The auralization module makes it possible to separate the nonlinear distortion from the linear signal for any test or audio signal provided to the analogue input of the Distortion Analyzer. A mixing console may be used to synthesize a defined ratio between distortion and linear signal as a stimulus for systematic listening tests. At the same time the internal state information in the microspeaker (displacement, distortion values, temperature) are recorded during the session. Thus this technique combines objective and subjective evaluation of microspeakers and is the basis for checking the audibility of each distortion component and their impact on perceived sound quality.
How to do an auralization ?	Create a new AUR operation and import the large signal parameters (in digital format) from the LSI operation. Select the crossover frequency. Start the AUR and apply an external stimulus. Adjust the level of the input signal to use the working range where the microspeaker should be evaluated. Adjust the level of the output level to reproduce the signal at a loudness is comfortable for the listener. Use the A/B switch to realize a blind comparison between a distorted and a non-distorted reproduction.

More Information

Papers	[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 [2] W. Klippel, Tutorial: Loudspeaker Nonlinearities - Causes, Parameters, Symptoms *J. Audio Eng. Society* **54**, No. 10 pp. 907-939 (2006 Oct.). [3] W. Klippel, "Equivalent Input Distortion," *J. Audio Eng. Society* **52**, No. 9 pp. 931-947 (2004 Sept.). [4] W. Klippel, "Nonlinear Modeling of the Heat Transfer in Loudspeakers," *J. Audio Eng. Society* **52**, erscheint Heft 1, 2004 January. [5] W. Klippel, J. Schlechter, "Measurement and Vizualisation of Loudspeaker Cone Vibration" preprint #6882, presented at the 125th Convention of the Audio Engineering Society, 2006 October 7 – 10, New York, USA. [6] W. Klippel, J. Schlechter, Distributed Mechanical Parameters Describing Vibration and Sound Radiation of Loudspeaker Drive Units, preprint presented at the 125th Convention of the Audio Engineering Society, 2008 October 2–5San Francisco, CA, USA
Application Notes	Find under www.klippel.de further application notes related to this subject [7] AN32 - Effective Radiation Area S_d [8] AN25 - Maximizing LPM Accuracy, [9] AN1 – Optical voice coil Posiition [10] AN4 – Measurement of Peak Displacement X_{max} [11] AN5 – Displacement Limits due to Driver Nonlinearities [12] AN19 – Measurement of Nonlinear Thermal Parameters [13] AN08 - 3D Intermodulation of speakers [14] AN09 - 3D Harmonic Distortion Measurement [15] AN22 - R&B Detection without Golden Unit [16] AN12 - Causes for Amplitude Compression [17] AN15 - Dynamic Generation of DC Displacement [18] AN17 - Credibility of Nonlinear Parameter Measurement [19] AN20 – Measurement of Equivalent Input Distortion [20] AN21 – Reduce distortion by shifting Voice Coil [21] AN31 – Cone Vibration and Radiation Diagnostics
Specification	[22] Motor und Suspension Check (MSC) S13,, software module of the KLIPPEL QC System [23] Vacuum Measurement Kit A7, hardware accessory of the KLIPPEL R&D System,
Software	User Manual for the KLIPPEL R&D SYSTEM.

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