Hands-On Training 7

Microspeaker

1 Objectives of the Hands-on Training

- Understanding the particularities of microspeakers (headphones, tweeters)
- Identifying the physical cause of the distortion
- Influence of air load (air mass, air in enclosed cavities)
- Creep and visco-elastic effects
- Performing measurements on microspeakers (free-air and vacuum)
- Understanding the measurement process
- Avoiding common mistakes in measurements
- Achieving practical skills for executing and interpreting the measurement
- Basic knowledge for development and evaluation of microspeakers

2 Requirements

2.1 Previous Knowledge of the Participants

It is recommended to do the previous *Klippel Trainings* before starting this training.

2.2 Minimal Requirements

Participants will need the results of the measurement provided in a Klippel database *Microspaker.kdbx* dispensing with a complete setup of the KLIPPEL measurement hardware. The data may be viewed by downloading *dB-Lab* from www.klippel.de/training and installing the software on a Windows PC.

2.3 Optional Requirements

If participants have access to a KLIPPEL R&D Measurement System, we recommend performing additional measurements on transducers provided by the instructor or other participants. In order to perform these measurements, you will also need the following software and hardware components:

- LSI, TRF, DIS and LPM Module
- Distortion Analyzer DA2
- Laser Sensor + Controller
- SCN Laser scanning system
- Vacuum chamber
- Amplifier
- Driver Stand

3 The training process

1. Read the theory that follows to refresh your knowledge required for the training.
2. Watch the demo video to learn about the practical aspects of the measurement.
3. Answer the preparatory questions to check your understanding.
4. Follow the instructions to interpret the results in the database and answer the multiple-choice questions off-line.
5. Check your knowledge by submitting your responses to the anonymous evaluation system at www.klippel.de/training.
6. Receive an email containing a **Certificate of Mastery, Knowledge or Participation** (depending on your performance).
7. Perform some optional measurements on transducers if the hardware is available.

4 Introduction

Microspeakers and transducers for headphones are small drivers used in headsets, mobiles and other telecommunication device. They normally must be operated over a wide frequency range with highest efficiency to save battery power. These characteristics cause some particularities, which will be explained in this training.

4.1 Particularities of Microspeakers

Most of the microspeakers use the electro-dynamical principle shown in Figure 1, with the rim zone of the diaphragm as suspension. Thus the diaphragm area is relatively large, the effective radiation area S_d does not correspond with the projected area of the diaphragm and direct mechanical measurements of the mechanical vibration by optical sensors (e.g. laser scanning techniques) are required to measure the S_d reliability as well as the mechanical parameters (T/S).

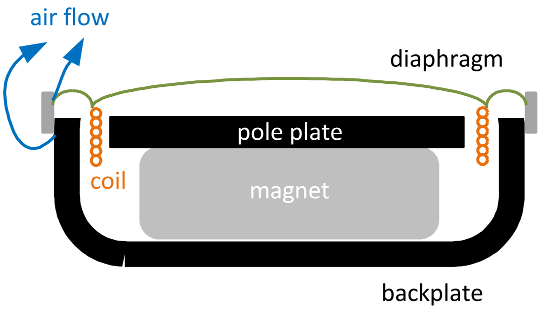


Figure 1: Sectional view of a micro-speaker

The flat and relatively small radiating source (1 cm diameter, approximately) compared with the wavelength at higher frequencies result in a low directivity factor.
Mechanical resistance $R_{ms}(v)$ is a dominant nonlinearity. The mainly reason for that are the low force factor *Bl* (lower than 1N/A), a high resonance frequency f_s (usually between 200Hz – 4kHz) and a quality factor Q_{ts} dominated by the mechanical losses.
The suspension without spider is prone to rocking modes, irregular vibration modes of the wire, the rim zone and hard limiting at maximal excursion. In the last case, Rub & Buzz and other impulsive distortion generating low-order circumferential modes may occur. Furthermore, the suspension system exposed to a sustained constant force increases the displacement versus time (creep effect). Due to visco elastic effects in the material the stiffness becomes smaller at lower frequencies and the air flow in the cavities below the diaphragm can also behave like a vented system.

4.2 Linear Lumped Parameter Modeling

The electrical, mechanical and thermal behavior of the drive unit at fundamental resonance frequencies can be described by the equivalent network comprising lumped elements with linear and nonlinear parameters. Distributed parameters are required to describe the vibration and radiation behavior at higher frequencies where the radiator (diaphragm) does not vibrate as a rigid body and the longitudinal waves propagate in circumferential and radial direction.
The linear parameters comprise the Thiele-Small parameters and visco-elastic parameters (creep factor). The inductance and the losses due to eddy currents are negligible. The dominant nonlinearities are the force factor $Bl(x)$, stiffness $Kms(x)$ and the nonlinear damping $Rms(v)$ versus velocity v as illustrated below.

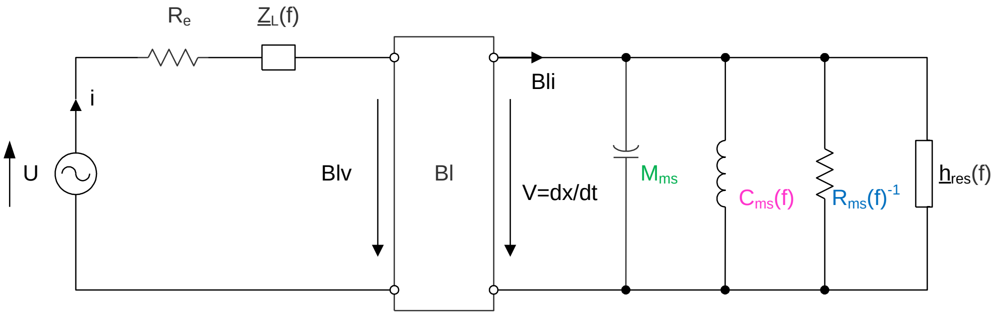


Figure 2: Simplified Lumped parameter model of a transducer operated in free air

Most of microspeakers show substantial creep which increases the compliance at low frequencies according to Figure 3. As consequence the high creep factor generates more displacement than predicted by conventional lumped parameter models, affecting the accuracy of the measured mechanical parameters.

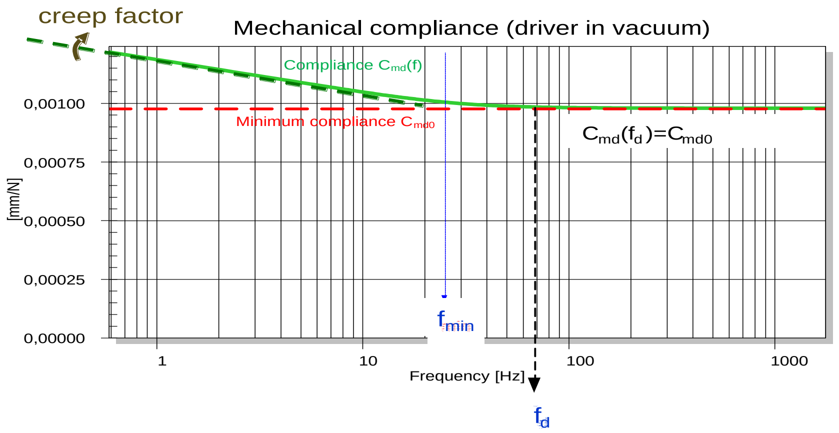


Figure 3: Frequency dependency of the mechanical compliance due to visco-elasticity (creep)

The visco-elasticity also affects the mechanical resistance, increasing the losses at low frequencies as shown in Figure 4.

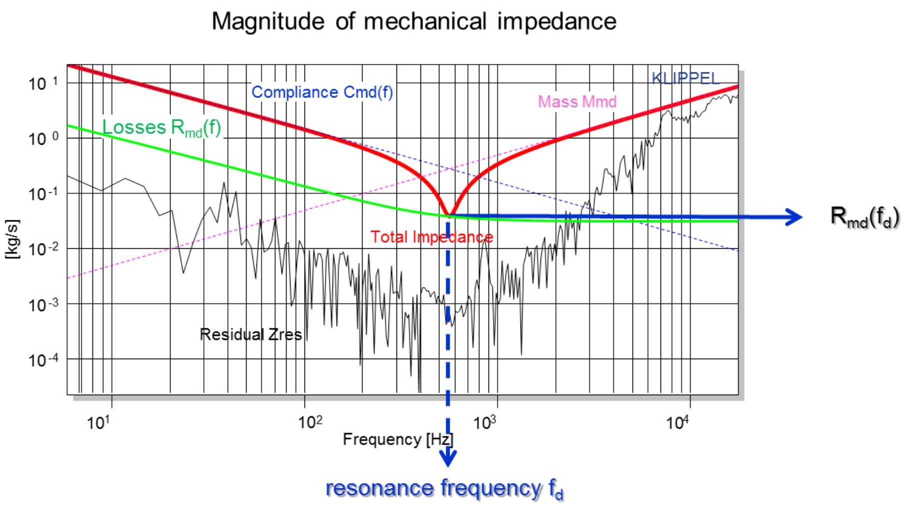


Figure 4: Magnitude of the mechanical impedance and the contribution of the lumped parameters

Nevertheless the air load causes additional losses, behaves as an additional air spring and increase the total mass, whose influence is not negligible. This effect depends on the measurement condition (vacuum, free air, enclosure…) which gives different values of moving mass M_{ms}, mechanical resistance R_{ms} and compliance C_{ms}.

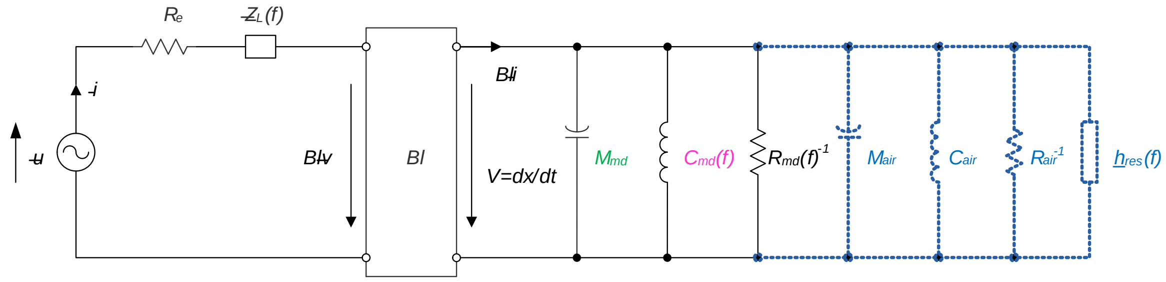
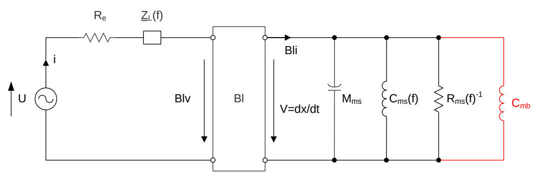
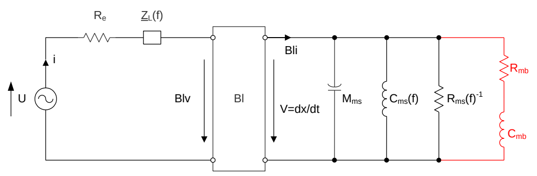
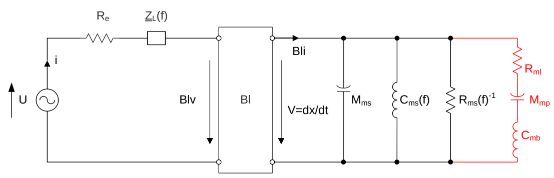
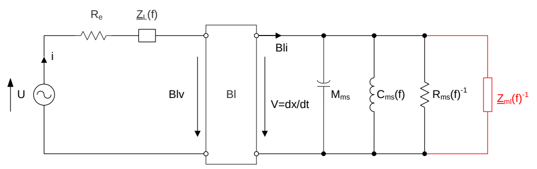


Figure 5: Modeling the transducer by pure mechanical parameters and additional parameters representing the air load

Table 1 shows different models for each measurement condition.

Table 1: Modeling and identification of enclosure properties

 Microspeaker in perfectly sealed enclosure	 Microspeaker in enclosure with air leak
 Microspeaker in vented enclosure	 Microspeaker with complex mechanical or acoustical load

5 Large Signal Modeling

Figure 6 shows the equivalent circuit used for transducer with f_s > 100 Hz like headphones, tweeter and microspeakers to identify nonlinearities. This model assumes constant R_{ms}. If $Q_{es} > Q_{ms}$ then it is recommended to measure in vacuum.

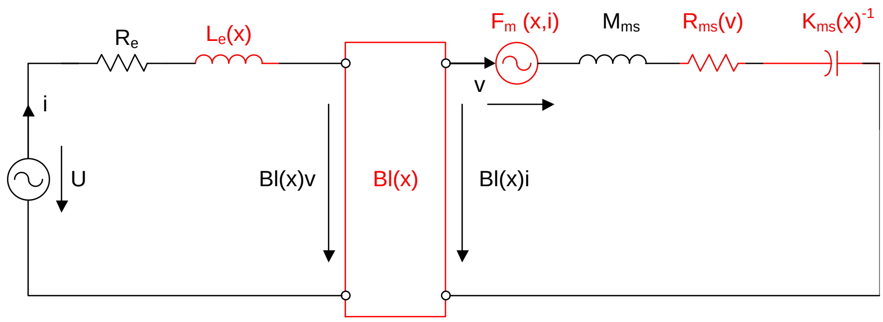


Figure 6: Equivalent circuit of the micro-speaker using lumped elements with nonlinear parameters

The measurement in vacuum reveals properties of the mechanical suspension without the influence of the air cavities in the transducer which causes, for instance, asymmetry in the stiffness curve (Figure 7).

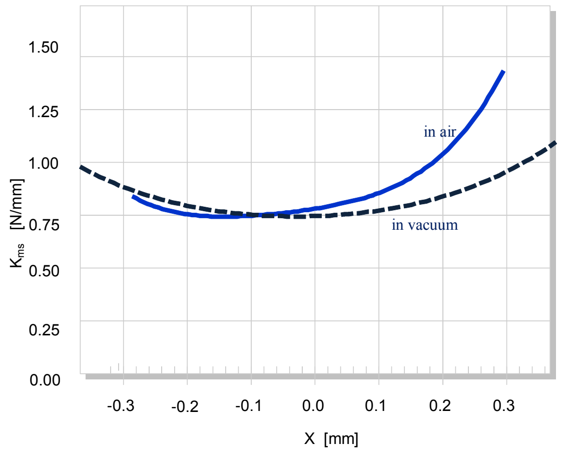


Figure 7: Nonlinear stiffness $K_{ms}(x)$ versus displacement x of a micro-speaker measured in air (solid curve) and in vacuum (dashed curve)

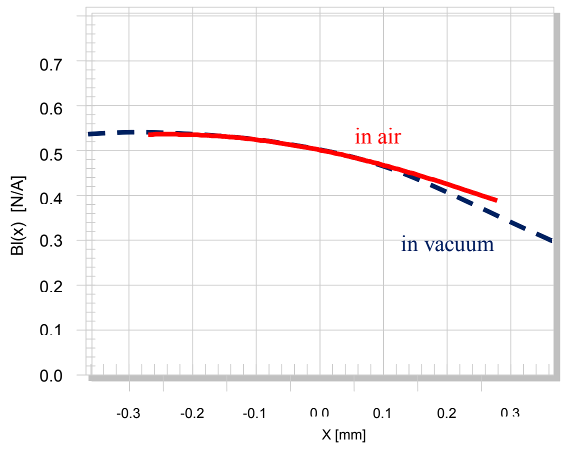


Figure 8: Nonlinear force factor $Bl(x)$ versus displacement x of a micro-speaker measured in air (solid curve) and in vacuum (dashed curve)

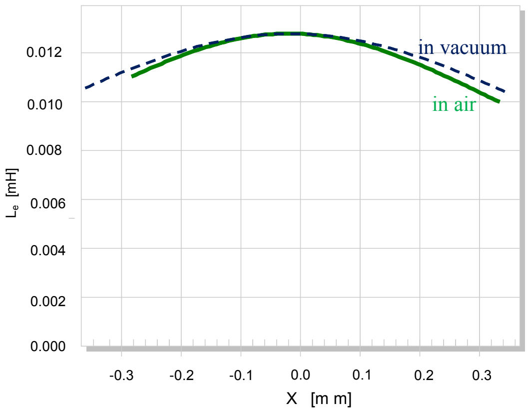


Figure 9: Nonlinear inductance $L_e(x)$ versus displacement x of a micro-speaker measured in air (solid curve) and in vacuum (dashed curve)

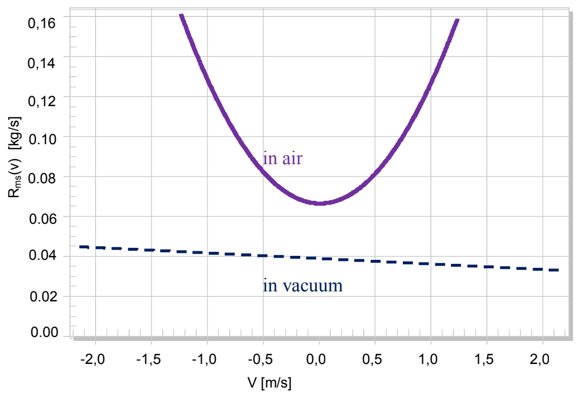


Figure 10: Nonlinear resistance $R_{ms}(v)$ versus voice coil velocity v of a micro-speaker measured in air (solid curve) and in vacuum (dashed curve)

Both measurements show an offset in the rest position of the voice coil, as illustrated on Figure 8. Figure 9 reveals that the air has no significant influence on the $L_e(x)$ nonlinearity. Contrary to these parameters the air caused a large diference in the $R_{ms}(v)$ curve, as shown in Figure 10.

6 Large Signal Performance

A transducer operated at high amplitudes generates nonlinear effects which are not found at small amplitudes. Comparing the predicted symptoms with the measured results it is possible to verify the model and identify the parameters besides investigating the effect of each transducer non-linearity separetely.

The most fundamental nonlinear symptom is the compression of the voice coil peak diplacement at high amplitudes. Figure 11 shows a good agreement between measured and predicted peak displacement while a linear model neglecting inherent nonlinearities would produce a much higher peak displacement at 700 Hz.

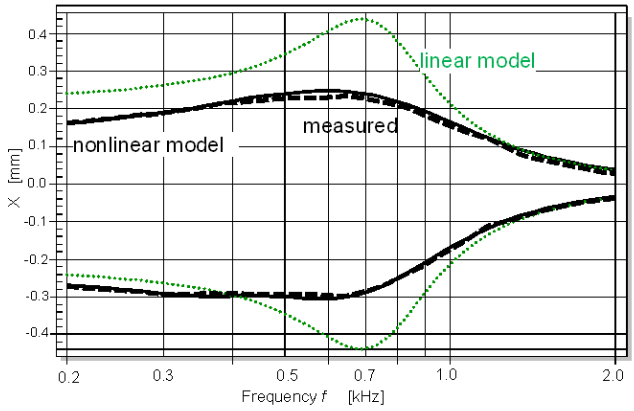


Figure 11: Peak and bottom displacement measured (solid line) and predicted by using a linear model (dotted curve) and a nonlinear model (dashed curve)

Figure 12 shows the peak displacement predicted by considering each nonlinearity separately. Thus it is possible to conclude that, in this case, the nonlinear resistance $R_{ms}(v)$ generates the amplitude compression at 700 Hz.

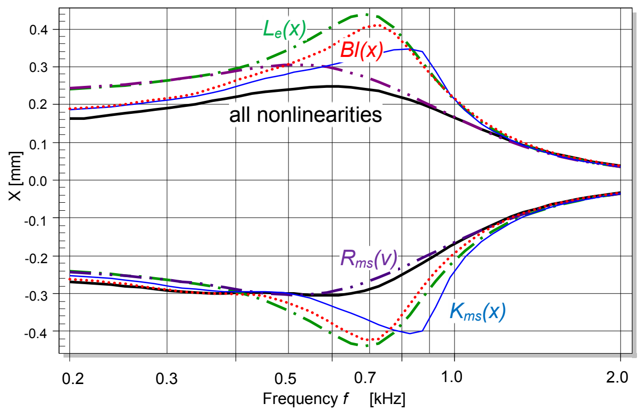


Figure 12: Peak and bottom displacement predicted by using all nonlinear parameters (solid line) and by considering each separated nonlinear cause

The negative DC displacement at the resonance frequency f_s= 700 Hz in Figure 13 is caused by $K_{ms}(x)$. At lower frequencies $Bl(x)$ and $K_{ms}(x)$ contribute the same amount to the total DC displacement. The $Bl(x)$-nonlinearity generates a positive DC displacement above 900 Hz increasing the voice coil offset, which is almost compensated by the stiffness nonlinearity. Since $L_e(x)$ and $R_{ms}(v)$ nonlinearities have almost a symmetrical curve shape, they do not significantly contribute to the total DC displacement.

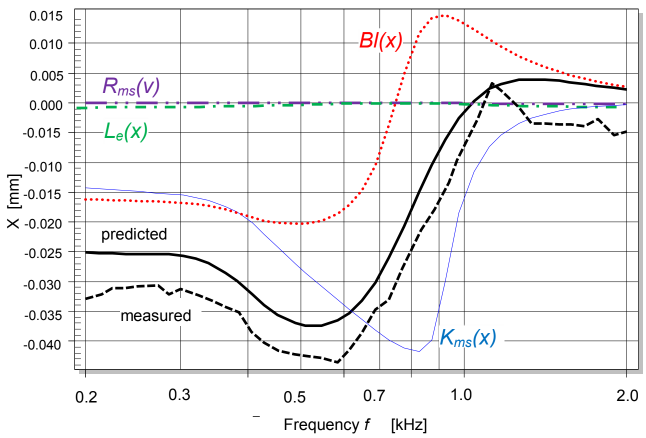


Figure 13: Measured (dashed line) and predicted (solid line) dc displacement versus frequency f and the contribution of each nonlinearity

Figure 14 reveals a compression of 6 dB of the fundamental component at 700 Hz by increasing the voltage at the terminals. This corresponds to a reduction of the total quality factor.

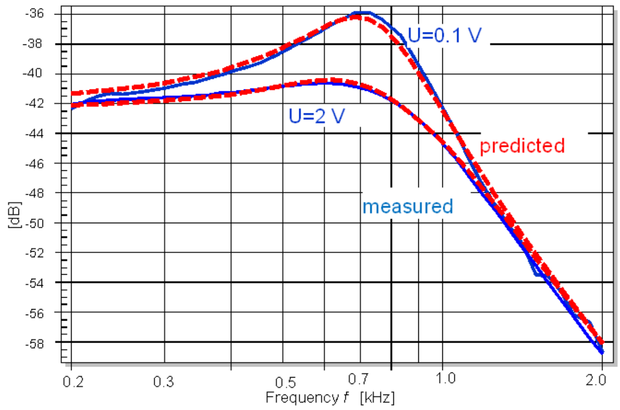


Figure 14: Measured (solid line) and predicted (dashed line) amplitude ratio of the fundamental component of voice coil displacement x and input voltage *U* in the small and large signal domain

7 Harmonic Distortion

The total harmonic distortion measured with a single-tone stimulus with varying frequency *f* is the most popular measurement of nonlinear symptoms. Figure 15 shows the measured distortion with the predicted response using all nonlinearities and the contribution of each nonlinearity. At frequencies below resonance the force factor *Bl(x)* and stiffness $K_{ms}(x)$ are the dominant causes of the distortion.

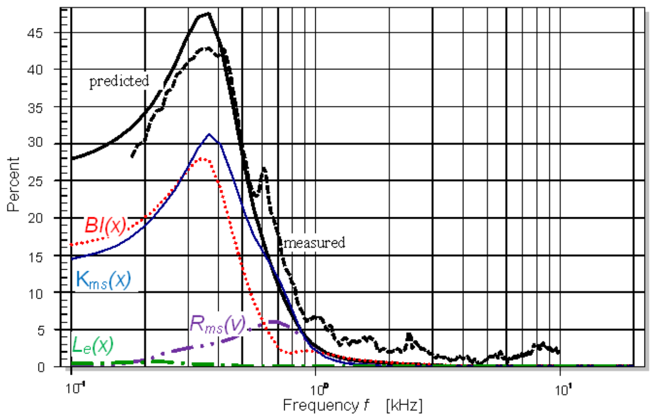


Figure 15: Total harmonic distortion in percent measured in the sound pressure output (dashed line) and simulated by nonlinear modeling using all nonlinearities (solid line) and each separated nonlinearity

There is also a good agreement between the measured and predicted 2nd-order equivalent harmonic input distortion at low frequencies in Figure 16. The discrepancy above 6 kHz is caused by measurement noise transferred into equivalent input distortion erroneously. Clearly the asymmetrical curve shape of *Bl(x)* and $K_{ms}(x)$ account for 2nd-order distortion below f_s.

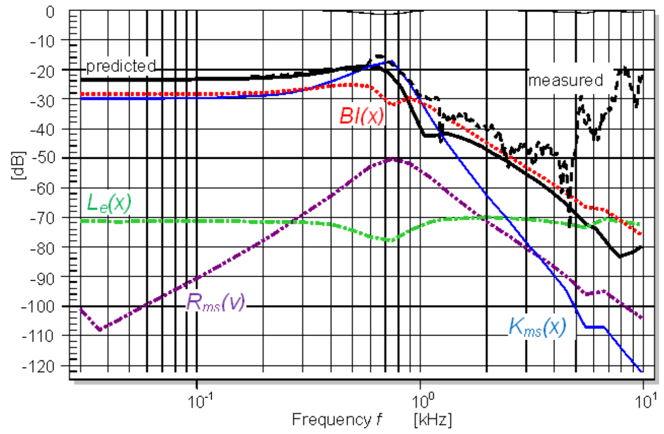


Figure 16: Measured (dashed line) and predicted (solid line) 2nd-order equivalent harmonic input distortion versus frequency *f* and the contribution of each nonlinearity

Figure 17 reveals the nonlinear resistance $R_{ms}(v)$ as the dominant source of the 3rd-order distortion at resonance frequency and above. Contrary to the $Bl(x)$ and $K_{ms}(x)$ distortion, the distortion generated by $R_{ms}(v)$ depends on the velocity and falls to lower and higher frequencies with the distance to the resonance frequency f_s=700 Hz.

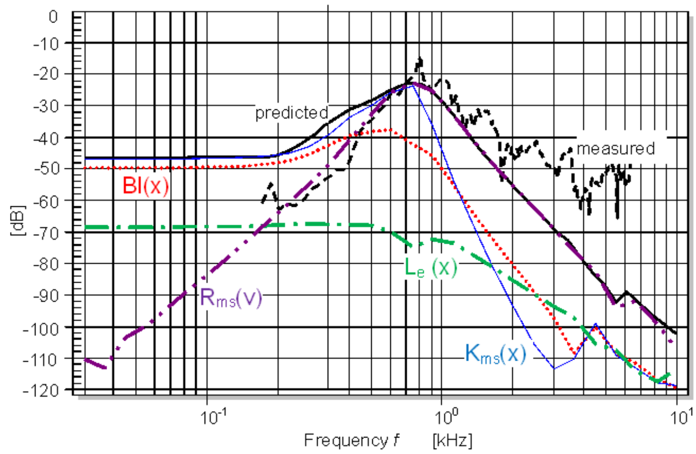


Figure 17: Measured (dashed line) and predicted (solid line) 3rd-order equivalent harmonic input distortion versus frequency *f* and the contribution of each nonlinearity

8 Intermodulation Distortion

The measurement of the intermodulation distortion (IMD) reveals the interaction with the two state variables multiplied in the static nonlinearity. The interpretation of the IMD frequency responses can be simplified by using a two-tone stimulus. The frequency of one tone is varied and the frequency of the other tone is constant.

For the particular micro-speaker used as example in Figure 18 the 2nd-order IMD is almost independent of the frequency f_1 of the first tone above 15 %. Only the IMD generated by the force factor $Bl(x)$ has a similar curve shape and is the dominant source of this distortion component. Although the $K_{ms}(x)$ generates significant 2nd-order harmonics of f_1, as shown in Figure 17, the IMD falls with the displacement generated by the second tone at a slope of 12 dB per octave to higher frequencies f_2.

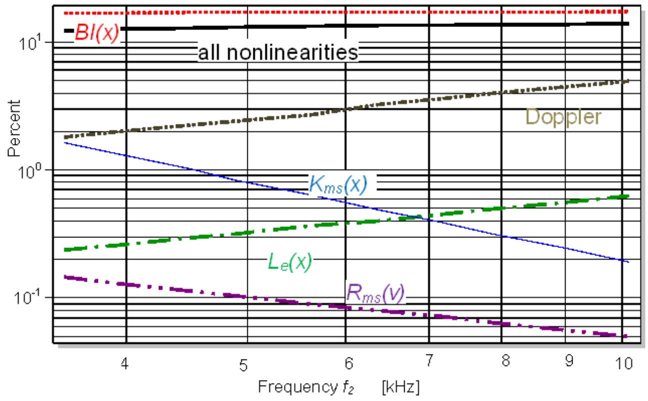


Figure 18: 2nd-order intermodulation distortion versus frequency f_2 generated by a two-tone stimulus (with f_2 > f_1=400 Hz) considering all nonlinearities (thick solid line) and only one selected nonlinearity in the modeling

The 3rd-order intermodulation distortion in Figure 19 is dominated by the *Bl(x)* and $R_{ms}(v)$ nonlinearities. While the IMD generated by *Bl(x)* only is almost constant versus frequency, the falling slope of the $R_{ms}(v)$ causes a destructive interference in the total IMD at lower frequencies f_1.

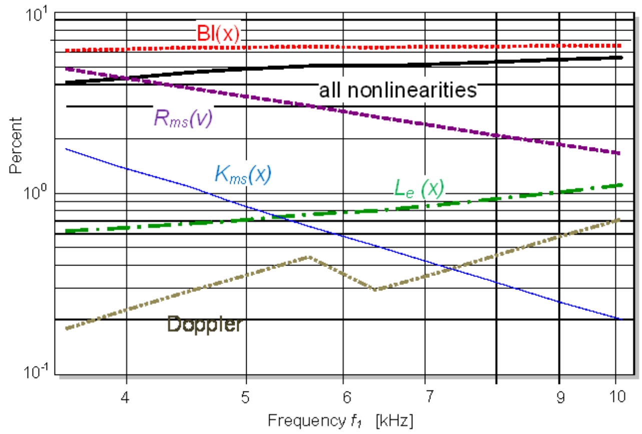


Figure 19: 3rd-order intermodulation distortion versus frequency f_1 generated by a two-tone stimulus (with $f_1 > f_2$= 400 Hz) considering all nonlinearities (thick solid line) and only one selected nonlinearity in the modeling

An alternative measurement of IMD distortion performs a variation of the low frequency tone f_1 while keeping the high-frequency tone f_2 at a fixed frequency. Figure 20 shows the 2nd-order IMD which is dominated by the *Bl(x)*-nonlinearity. For frequencies $f_1 > f_s$=700 Hz the IMD distortion falls with the voice coil displacement generated by the first tone.

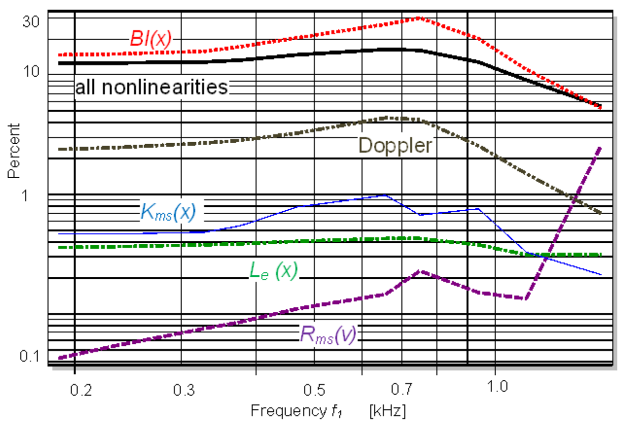


Figure 20: 2nd-order intermodulation distortion versus f_1 generated by a two-tone stimulus (with $f_1 < f_2$= 6 kHz) considering all nonlinearities (thick solid line) and only one selected nonlinearity in the modeling

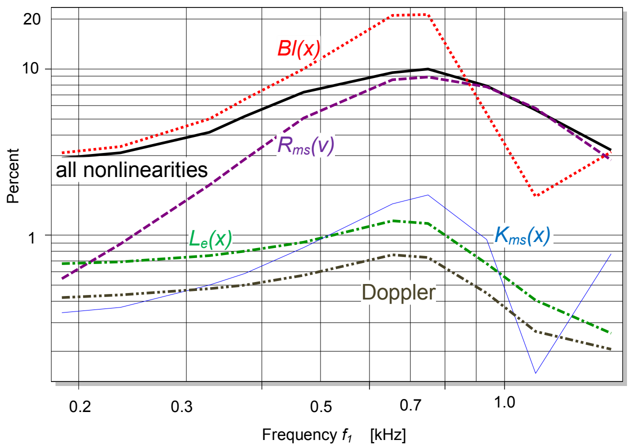


Figure 21 reveals the *Bl(x)* and $R_{ms}(v)$ as the dominant cause of the 3rd-order intermodulation distortion at resonance frequency. Again the contribution of the force factor *Bl(x)* is overestimated because the amplitude compression caused by $K_{ms}(x)$ is not considered.

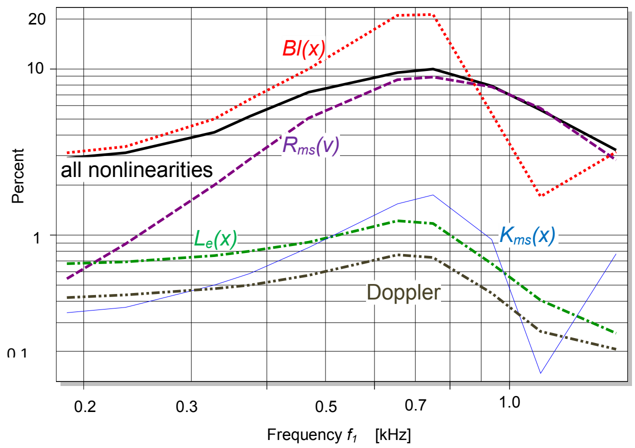


Figure 21: 3rd-order intermodulation distortion versus f_1 generated by a two-tone stimulus (with $f_1 < f_2$= 6 kHz) considering all nonlinearities (thick solid line) and only one selected nonlinearity in the modeling

9 Preparatory Questions

Check your theoretical knowledge before you start the regular training. Answer the questions by selecting all correct responses (sometimes, there will be more than one).

QUESTION 1: Is the resonance frequency f_s of a microspeaker measured in free air without any enclosure always higher than the resonance frequency of the same unit measured in vacuum?

□ **MC a:** No, the moving air particles increase the total mass which reduces the resonance frequency of all microspeakers when performing the measurement in free air.

□ **MC b:** Yes, the enclose air in cavities below the diaphragm contributes to the total stiffness and increases the resonance frequency of all microspeakers when performing the measurement in free air.

□ **MC c:** No, it depends on the properties of the particular microspeaker. If the influence of the added air mass is larger than the influence of the additional stiffness then the resonance frequency will be lower for operating the microspeaker in air.

□ **MC d:** No, the resonance frequency is independent of the air load and measurements in air and in vacuum give the same results.

QUESTION 2: Is the displacement *x* constant and independent of the frequency *f* (with $f << f_s$) for a constant excitation voltage?

□ **MC a:** Yes, at very low frequencies the displacement *x* is proportional to the compliance of the mechanical suspension (assuming R_e and *Bl* is constant) which is constant at frequencies $f << f_s$.

□ **MC b:** No, the enclosed air below the diaphragm contributes with additional air stiffness to the total stiffness which will vanish at very low frequencies due to a small leakage. This effect increases the displacement x to lower frequencies.

□ **MC c:** No, the visco-elastic behaviour of the mechanical suspension causes a decrease of the stiffness to lower frequencies. This effect increases the displacement x to lower frequencies.

□ **MC d:** Yes, the displacement x is independent of the frequency f (for $f << f_s$) if the microspeaker is operated in vacuum.

QUESTION 3: What role plays the mechanical resistance R_{ms} in micro-speakers?

□ **MC a:** The mechanical resistance R_{ms} plays a minor role in most micro-speakers because the quality factor Q_{ts} is dominated by the electrical quality factor Q_{es} because $Q_{es} << Q_{ms}$. Thus the total damping can be realized by a low electrical dc resistance R_e and a high Bl factor.

□ **MC b:** The mechanical resistance R_{ms} is an important parameter in most micro-speakers because mechanical (and acoustical) losses are required to provide sufficient damping to the resonance and to realize a desired quality factor Q_{ts}. Contrary to woofers the electrical damping is not dominant in microspeakers because the value of the force factor Bl is much smaller.

□ **MC c:** The mechanical resistance R_{ms} plays an important role at very low frequencies $f << f_s$ because this value increases to lower frequencies due to the visco-elastic properties of the material (creep effect).

□ **MC d:** The effect of the mechanical resistance R_{ms} is negligible at very low frequencies $f << f_s$ and at very high frequencies $f >> f_s$ because this stiffness and the moving mass, respectively, determine the total impedance at those frequencies.

QUESTION 4: What limits the reproduction of low frequencies in microspeakers?

□ **MC a:** The effective radiation area S_d is relatively small.

□ **MC b:** The maximal peak displacement of the voice coil is limited.

□ **MC c:** The total moving mass is too small giving a high resonance frequency f_s.

QUESTION 5: Why are shorting rings and other means for linearizing the inductance nonlinearity not required in microspeakers?

□ **MC a:** The inductance value (L_e << 0.2mH) is relatively small.

□ **MC b:** The dc resistance R_e is relatively large (R_e > 4 Ohm) and dominates the total electrical impedance at higher frequencies.

□ **MC c:** The inductance is constant and does not vary with displacement and current.

QUESTION 6: Why is the nonlinear variation of the mechanical resistance $R_{ms}(v)$ versus velocity v one of the dominant source of signal distortion in some microspeakers but not relevant in woofers?

□ **MC a:** The electrical damping dominates the total quality factor Q_{ts} in woofers and it does matter whether $R_{ms}(v)$ and the corresponding mechanical quality factor varies with velocity.

□ **MC b:** The mechanical resistance $R_{ms}(v)$ is constant and does not vary versus velocity v in woofers.

□ **MC c:** The moving mass in woofers is much higher than in microspeakers.

□ **MC d:** The force factor Bl is much higher in woofers than in micro-speakers generating a dominant electrical damping ($Q_{es} < Q_{ms}$).

QUESTION 7: Most woofers use a spider but microspeakers do not. What are the pro and contra of using a spider?

□ **MC a:** The spider and a surround assembled with a cone would suppress rocking modes and provide robustness against voice coil rubbing in micro-speakers.

□ **MC b:** The spider would increase the height of micro-speaker which is not acceptable in many applications.

□ **MC c:** A spider may be used to generate a desired nonlinear stiffness characteristic which increases progressively for positive and negative displacement and provides a natural protection of the voice coil against bottoming.

□ **MC d:** A spider supports centring of the coil in the gap and makes it possible to use a smaller clearance between coil and pole tips in the gap. A smaller air gap reduces the magnetic resistance and leads to maximal magnetic flux density B and force factor *Bl*.

10 Interpretation of Measurements (no hardware required)

Step 1: View the demo movie *Microspeaker* provided at www.klippel.de/training to view how a practical measurement of the linear lumped parameters is performed.

Step 2: Click on the database *Microspeaker.kdbx* to view the results of the measurement. Open the object ' ***LPM Measurement*** and view the results of the LPM measurement in free air.

Advice: It is recommended to do the following exercises offline and to note the answers of the multiple choice questions on a paper!

QUESTION 8: Are the measurement of the impedance and the displacement transfer function performed by using a useful stimulus and sufficient number of averaging?

□ **MC a:** Yes, the measurements are useful because the noise and distortion components found in the current signal and displacement spectrum are more than 30 dB below the fundamental component.

□ **MC b:** No, the measurements are not useful because the noise and distortion components (grey lines) measured by applying the stimulus are almost identical with the noise floor (black lines) measured without stimulus. Thus the averaging should be increased.

Step 3: Open the operation ***CAL LPM Summary*** and view the results windows 1 and 2 showing the "***Magnitude response of the transfer function*** $H_x(f,r_c)$***"*** measured at point r_c and the ***"Magnitude of the electrical impedance*** $Z_e(f)$***",*** respectively. Compare the variation (difference Air Measured Maximum and Air Measured Minimum) of the electrical impedance $Z_e(f)$ and mechanical transfer function $H_x(f)$ of the microspeaker measured in free air.

QUESTION 9: Why is the variation of the mechanical transfer function much higher than the variation of the electrical impedance?

□ **MC a:** A diaphragm of a microspeaker does not vibrate at low frequency like a piston. Irregularities in the thickness and geometry of the diaphragm (mainly the outer surround area) may cause significant variation of the magnitude of the displacements versus various measurement points.

□ **MC b:** Mechanical measurements of displacement or velocity are always more corrupted by measurement noise than measurements of the electrical signals (voltage and current).

□ **MC c:** The laser sensor was not correctly calibrated.

Step 4: Open the result window "***Magnitude of transfer function*** $H_x(f,r_c)$***"*** in the operation ***CAL LPM Summary*** and copy the curve *Air Measured Average* into the clipboard. Paste it into the result window "$H_x(f)$ ***Magnitude"*** of the operation ***LPM AIR Center***. Compare the averaged curve with the center curve.

QUESTION 10: Is the center curve measured at the center of the diaphragm identical with the averaged curve? If not which curve is a more accurate basis for calculating the force factor *Bl*?

□ **MC a:** The two curves are always identical and they give the same *Bl* value.

□ **MC b:** The two curves are not identical if there are irregular circumferential modes on the diaphragm. The curve averaged on multiple points equally distributed over the circumference of the voice coil gives a more accurate value of the *Bl* than the vibration in the center because it describes the mean displacement x_{coil} of the voice coil which corresponds with the generated back EMF.

□ **MC c:** The vibration measured in the center of the diaphragm describes the mean movement of the coil and is the most accurate basis for calculating the force factor. The mean displacement calculated by averaging the vibration of equally distributed points is corrupted by rocking modes and other irregular vibrations. This produces a high error in the estimated *Bl*.

QUESTION 11: Which measurement (transducer operated in air or in vacuum) gives the most accurate value of the force factor *Bl*?

□ **MC a:** The force factor should be determined by the results of the air measurement because the force factor is not affected by the air load and optical problems caused by laser sensor are minimal. If the transducer is operated in vacuum a transparent glass plate is required which may affect the accuracy of the optical measurement.

□ **MC b:** The force factor is a mechanical parameter and should be measured without the influence of the air.

□ **MC c:** The force factor *Bl* is calculated only by using the electrical impedance curve and the laser measurement will not affect the accuracy of the force factor measurement.

Step 5: Open the window "***Result Variables"*** in the operation ***CAL LPM Summary*** and compare the resonance frequency f_d of the microspeaker in vacuum with the resonance frequency f_s measured in air.

QUESTION 12: Which parameter and which physical effect causes the difference?

□ **MC a:** The change of the resonance frequency is caused by the stiffness of the enclosed air below the diaphragm which is large compared with the stiffness of the mechanical suspension. The mass of the air load is negligible to the mass of the moving mechanical parts.

□ **MC b:** The change of the resonance frequency is caused by the air load increasing the total moving mass. The stiffness of the enclosed air below the diaphragm is small compared with the stiffness of the suspension.

□ **MC c:** The change of the resonance frequency is caused by mechanical resistance which is higher in air then in vacuum.

Step 6: Open the result window ***"Curve 4 Directivity Index"*** in the operation ***0a CAL Scanner Results*** of the object ***Result microspeaker***.

QUESTION 13: Why is the directivity factor of this microspeaker significantly lower than the directivity factor of an ordinary full-band loudspeaker using a paper cone?

□ **MC a:** The diaphragm in the microspeaker has a much smaller diameter than the full-band speaker. The radiating surface is relatively small compared with the wavelength at higher frequencies.

□ **MC b:** The diaphragm in the microspeaker uses a flat diaphragm whereas the full-band loudspeaker uses a cone.

□ **MC c:** The diaphragm in the microspeaker has a lower mass than the cone of the full band speaker.

□ **MC d:** The diaphragm in the microspeaker has a smaller thickness than the paper cone giving a lower bending stiffness.

Step 7: View the curve *Quadrature Acceleration* in the result window "***Curve 1 SPL Decomposition"*** in the operation ***0a CAL Scanner Results*** in the object ***Result microspeaker***. Search for the frequencies where the quadrature acceleration shows significant peaks. Start the scanner analysis software by clicking on the file ***Microspeaker.ksp*** and animate the quadrature component at these frequencies.

QUESTION 14: The dominant rocking mode occurs at about

□ **MC a:** 14 Hz

□ **MC b:** 590 Hz

□ **MC c:** 5.8 kHz

□ **MC d:** 8.7 kHz

- □ **MC e:** 16.4 kHz

QUESTION 15: Which constructional components are involved in the rocking mode?

- □ **MC a:** The rocking mode causes deformation of all areas of the diaphragm.
- □ **MC b:** The rocking mode causes a deformation of the outer part of the diaphragm (surround zone) between voice coil and frame causing a tilting of the inner part of the diaphragm vibrates which still moves (rocks) as a rigid body.
- □ **MC c:** The rocking mode affects all parts of the diaphragm but not the area where the voice coil gives the diaphragm sufficient stability.

Step 8: Open the operation ***2a LSI Nonlinear Parameters*** in the object ***Result microspeaker***. View the result window "***State***" and compare the parameters and state variables (force factor $Bl_{min,}$ compliance C_{min}, increase of voice coil temperature delta T_v and input power P) at the end of the measurement with corresponding protection parameter (Bl_{lim}, C_{lim}, T_{lim} and P_{lim}).

QUESTION 16: Which protection parameter limits the permissible working range?

- □ **MC a:** The force factor nonlinearity limits the working range because the minimal force factor $Bl_{min}=Min(Bl(x)/Bl(x=0))$ is below the permissible limit Bl_{lim}= 50 %.
- □ **MC b:** The force compliance nonlinearity limits the working range because the minimal value of the compliance $C_{min}=Min(C_{ms}(x)/C_{ms}(x=0))$ is below the permissible limit C_{lim}= 50 %.
- □ **MC c:** The voice coil temperature limits the working range because the increase of the voice coil temperature ΔT_v exceeds the permissible limit ΔT_{lim}.
- □ **MC d:** The maximal input power P limits the working range because this value exceeds the permissible limit P_{lim}.

QUESTION 17: Why should the permissible working range of microspeakers, headphones, microphones and other transducer using no spider be limited by the input power P_{lim}?

- □ **MC a:** Microspeakers may be damaged if the mechanical system performs irregular vibration (coil rubbing at the pole tips, bottom at the pole plate, wire beat…). Those irregular vibrations may already occur at small displacement where the force factor and the compliance shows only minor nonlinearities. Therefore the user should limit the input power during the LSI measurement if impulsive distortion (rub & buzz) are generated by the transducer during testing.
- □ **MC b:** Microspeakers operated in air will be damaged by a high value of voice coil velocity because the mechanical resistances $R_{ms}(v)$ rises with velocity v.
- □ **MC c:** Microspeakers will be damaged by a high level of acceleration.
- □ **MC d:** Microspeakers will be damaged by a high peak value of the input current.

Step 9: Open the result window ***"Distortion"*** and compare the peak value of the distortion D_b, D_c, D_l and $D_{rms(v)}$ generated by $Bl(x)$, $C_{ms}(x)$, $L(x)$ and $R_{ms}(v)$.

QUESTION 18: Why are the distortion generated by $L(x)$ negligible in microspeakers ?

- □ **MC a:** The curve shape of the $L(x)$ nonlinearity is almost constant over displacement x.
- □ **MC b:** The stimulus of the signal used in the LSI is low-pass filtered and there is not much spectral energy at higher frequencies where the inductance $L(x)$ causes a variation of the electrical impedance.
- □ **MC c:** The total electrical impedance at higher frequencies ($f > f_s$) is dominated by the electrical resistance R_e because the value of the inductance $L(x)$ is relatively small (L_e< 0.1 mH).
- □ **MC d:** The stimulus used in the LSI measurement generates only small displacement which cause no variation of the inductance $L(x)$.

Step 10: View the shape of the nonlinear mechanical resistance in the result window "***$R_{ms}(v)$***" in the operation ***2a LSI Nonlinear Parameters***.

QUESTION 19: What kind of distortion is generated by $R_{ms}(v)$?

- □ **MC a:** This nonlinearity generates a significant compression of the fundamental component at the resonance frequency f_s because $R_{ms}(v)$ has almost a symmetrical shape generating all kinds of odd-order nonlinear components (including a 1st-order contribution reducing the fundamental) and the velocity is maximal at the resonance.

□ **MC b:** This nonlinearity generates significant 2nd-order distortion (harmonics and intermodulation distortion) because the curve shape of $R_{ms}(v)$ is very asymmetrical.

□ **MC c:** This nonlinearity generates significant 2nd-order distortion (harmonics and intermodulation distortion) because $R_{ms}(v)$ has an almost symmetrical shape.

□ **MC d:** This nonlinearity generates 3rd–order distortion for excitation frequencies close to the resonance frequency f_s because $R_{ms}(v)$ has a symmetrical shape generating all kinds of odd-order nonlinear components (harmonics and intermodulations) and the velocity is maximal at the resonance.

Step 11: View the shape of the nonlinear force factor in the result window “***Bl(x)***” in the operation ***2a LSI Nonlinear Parameters.***

QUESTION 20: Does the force factor *Bl(x)* of this microspeaker generate a DC component in the displacement?

□ **MC a:** Force factor nonlinearity cannot produce any dc displacement at the resonance frequency f_s because current and displacement multiplied by each other at this frequency are 90 degree out of phase at this frequency.

□ **MC b:** Force factor nonlinearity cannot produce any DC displacement at any frequency.

□ **MC c:** Force factor nonlinearity generates a negative DC displacement moving the coil towards the *Bl* maximum at low frequencies ($f < f_s$).

□ **MC d:** Force factor nonlinearity generates a positive DC displacement moving the coil away from the *Bl* maximum at high frequencies ($f > f_s$).

Step 12: View the shape of the nonlinear stiffness in the result window “$\boldsymbol{K_{ms}(x)}$” in the operation ***2a LSI Nonlinear Parameters*****.**

QUESTION 21: Does the stiffness nonlinearity $K_{ms}(x)$ of this microspeaker generate a DC component in the displacement?

□ **MC a:** Stiffness nonlinearity produces no DC displacement at any frequency.

□ **MC b:** Stiffness nonlinearity generates a positive DC displacement moving the coil towards the softer side of the suspension for any frequency (the dc component becomes smaller at higher frequencies when the total displacement decreases).

□ **MC c:** Stiffness nonlinearity generates a negative DC displacement moving the coil towards the softer side of the suspension at low frequencies ($f < f_s$) but this nonlinearity generates a positive dc displacement moving the coil towards the harder side of the suspension at high frequencies ($f > f_s$).

□ **MC d:** Stiffness nonlinearity produces no dc displacement at the resonance frequency f_s.

Step 13: View the result window “***Compression***” in the operation ***3a DIS X Fundamental, DC.***

QUESTION 22: What causes the compression of the fundamental component at the resonance frequency f_s?

□ **MC a:** The compression of the fundamental is caused by the force factor nonlinearity *Bl(x)* because a decreasing *Bl(x)* reduces the electrical damping and increases the electrical quality factor Q_{es}.

□ **MC b:** The compression of the fundamental at the resonance frequency f_s is caused by the force factor nonlinearity *Bl(x)* because a decreasing *Bl(x)* reduces the electro-dynamical driving force.

□ **MC c:** The compression of the fundamental at the resonance frequency f_s is caused by the stiffness nonlinearity $K_{ms}(x)$ because the $K_{ms}(x)$ rises at higher excursion cause a higher restoring force.

□ **MC d:** The compression of the fundamental at the resonance frequency f_s is caused by the mechanical resistance $R_{ms}(v)$ because velocity is maximal at the resonance and the $R_{ms}(v)$ increases the total damping significantly.

Step 14: View the result window "***DC Component***" in the operation ***3a DIS X Fundamental, DC.***

QUESTION 23: Why does the dc displacement fall with rising voltage at low frequencies ($f < f_s$)?

- □ **MC a:** The stiffness characteristic $K_{ms}(x)$ has a distinct asymmetry at low amplitudes but becomes more symmetrical at higher amplitudes.
- □ **MC b:** The asymmetry of the force factor generates a negative dc component (moving the coil to the *Bl* maximum) which rises with voltage and compensates the positive dc component generated by the asymmetrical stiffness $K_{ms}(x)$ at higher amplitudes.
- □ **MC c:** The mechanical resistance $R_{ms}(v)$ has a very symmetrical curve shape which keeps the voice coil in the gap at low frequencies.

Step 15: View the result window "Fundamental + Harmonic Distortion" in the operation 4a TRF SPL + Harm. (Usine).

QUESTION 24: Why is the absolute value of total harmonic distortion maximal at the resonance frequency?

- □ **MC a:** The amplitude of the displacement *x* is maximal at f_s which generates the largest variation of the displacement depending parameters *Bl(x)* and $K_{ms}(x)$.
- □ **MC b:** The amplitude of the velocity v is maximal at f_s which generates the largest variation in the velocity depending resistance $R_{ms}(v)$.
- □ **MC c:** The total harmonic distortion are falling by 12dB per octave for $f < f_s$ (like the fundamental component) because the nonlinear distortion generated by the nonlinearities *Bl(x)*, $K_{ms}(x)$ and $R_{ms}(v)$ have to pass the linear loudspeaker part behaving as a post-filter with a high-pass characteristic.

Step 16: Copy the *Fundamental* response in the result window "***Fundamental + Harmonic Distortion***" in the operation ***4a TRF SPL + Harm. (Usine)*** into the clipboard. Open the property page *Processing* and paste the curve into EDIT under reference curve. Open the window "***Impulse response***" and set the cursor properly to window the impulse response between 0 ms and 200 ms. This imported Fundamental curve is used as an inverse filter to the measured sound pressure signal prior to the spectral analysis. View result window "***Fundamental + Harmonic Distortion***"**.** where the fundamental component should be approximately flat curve and equivalent input distortion are presented.

QUESTION 25: Why are the equivalent input distortions (2nd-order, 3rd-order, THD) approximately constant between 100 -400 Hz?

- □ **MC a:** The distortion generated in this frequency range are generated by the displacement varying nonlinearities *Bl(x)* and $K_{ms}(x)$ and the amplitude of the displacement *x* is almost constant (see result window **"*Fundamental Component*"** in operation ***3a DIS X Fundamental, DC***).
- □ **MC b:** The equivalent input distortion describes the distortion at the loudspeaker terminals close to the source. Therefor the high-pass characteristic of the loudspeaker between terminal voltage and sound pressure output will not cause an additional shaping of the nonlinear distortion in this frequency range by 12dB per octave.
- □ **MC c:** The equivalent input distortion are always constant for excitation frequencies $f < f_s/2$.

Step 17: Inspect the *2nd-order* and *3rd-order harmonic* distortion measured in the input current in the results window "***Harmonic Distortion (relative)***" in the operation ***4b TRF Harmonics in current.*** Compare the values and the shape of the distortion response in the current with the distortion found in the sound pressure output in corresponding window in the operation ***4a TRF SPL + Harm. (Usine).***

QUESTION 26: What does the comparison show?

- □ **MC a:** The nonlinear inductance *L(x)* is the dominant source of the nonlinear distortion in the microspeaker.
- □ **MC b:** The relative harmonic distortion found in the input current is much smaller than the distortion found in the sound pressure. Therefore the dominant distortion is generated in

mechanical and acoustical domain.

- □ **MC c:** The frequency response of the harmonics distortion found in the input current is almost flat above the resonance frequency f_s. This reveals the effect of the displacement varying inductance *L(x)*.

Step 18: Inspect the 2^{nd}*-order* and 3^{rd}*-order intermodulation* distortion in the sound pressure output measured by using a two-tone signal with f_1 varied between 2 – 10 kHz and f_2 = 200Hz shown in the result windows "***2nd Intermod,%"*** and "***3rd Intermod, %"*** in the operation ***4c DIS IM Dist. (voice sweep) P***. Compare the distortion found in the sound pressure with the corresponding distortion found in the input current in operation ***4d DIS IM Dist. (voice sweep) I***.

QUESTION 27: What does the comparison show?

- □ **MC a:** The relative intermodulation distortion found in the input current is much smaller than the intermodulation distortion found in the sound pressure. Since the curve shape of the distortion in current is very random it is very likely that this measurement is corrupted by noise. The inductance *L(x)* does not cause neither 2^{nd}–order nor 3^{rd}-order intermodulation distortion found in the sound pressure output.
- □ **MC b:** The frequency response of 2^{nd}-order intermodulation distortion in the sound pressure is almost constant and independent of the frequency f_1 for 1 kHz < f_1 < 6kHz. This distortion is generated by the asymmetrical variation of the force factor *Bl(x)* because the tone f_2 generates almost constant displacement *x* causing the same intermodulation of the first tone for any frequency f_1. The intermodulation distortion at higher frequencies f_1 > 6 kHz are generated by a different nonlinearity caused by the modal vibration at higher frequencies.
- □ **MC c:** The frequency response of 3^{rd}-order intermodulation distortion in the sound pressure is almost constant and independent of the frequency f_1 for 1 kHz < f_1 < 6 kHz. This distortion is generated by the symmetrical part of the force factor variation *Bl(x)* because the tone f_2 generates almost constant displacement *x* causing the same intermodulation of the first tone for any frequency f_1. The intermodulation distortion at higher frequencies f_1 > 6 kHz are generated by a different nonlinearity caused by the modal vibration at higher frequencies.

Step 19: Inspect the sound pressure spectrum in the result window "***P(f) spectrum"*** in the operation ***4e LPM Multitone Dist. (Umt)*** measured by using a multi-tone stimulus. Determine the difference in dB between fundamental components and distortion at higher frequencies 3 – 10 kHz. Compare this difference with the difference of the corresponding distortion in the input current in result window "***Current I(f) Spectrum"***.

QUESTION 28: What does the comparison show?

- □ **MC a:** The sound pressure spectrum reveals no nonlinear distortion. The signal component which are between the fundamental components are measurement noise (for example generated by an insensitive microphone).
- □ **MC b:** The inductance *L(x)* is not a dominant nonlinearity in this microspeaker because the difference between fundamental and distortion in the sound pressure spectrum is about 40 dB while the difference in the current spectrum is much higher (about 60 dB).
- □ **MC c:** The force factor nonlinearity *Bl(x)* is the dominant cause of nonlinear distortion at higher frequencies because the distortion are almost constant for $f > f_s$ and much lower than the corresponding distortion in the input current.
- □ **MC d:** The stiffness $K_{ms}(x)$ is the cause of the intermodulation distortion at higher frequencies $f > f_s$.
- □ **MC e:** The nonlinear force factor *Bl(x),* stiffness $K_{ms}(x)$ and nonlinear resistance $R_{ms}(v)$ contributes to the distortion maximum at the resonance frequency f_s.

Step 20: Open the object '***Systematic R&B Test Tweeter*** showing a series of TRF measurements where the voltage of the stimulus is varied from 0.1 V up to 2 V. Inspect the peak value of the higher-order distortion (*Absolute PHD*) in the result window ***"Fundamental + Harmonics"*** and the crest factor of the higher-order distortion *(ICHD*) in the result window ***"Instantaneous Crest Factor".*** Search for the lowest voltage when PHD > PHD_{limit} (with limit threshold PHD_{limit} which

40dB below the mean fundamental response) and ICHD > 12 dB which is a useful criterion of a significant defect.

QUESTION 29: The microspeaker generates the first significant irregular distortion for a sinusoidal sweep at a terminal voltage:

- □ **MC a:** U = 0.1 V corresponding with a peak displacement of 20 µm.
- □ **MC b:** U = 0.2 V corresponding with a peak displacement of 35 µm.
- □ **MC c:** U = 2.0 V corresponding with a peak displacement of 350 µm.

Step 21: Open the result window ***"Instantaneous Distortion (3D)"*** which shows the crest factor versus frequency *f* of the sinusoidal stimulus and voice coil displacement *x*. Search for the voice coil position where the first impulsive distortions are generated.

QUESTION 30: What does the Instantaneous Crest factor reveal?

- □ **MC a:** The impulsive distortion is generated by hard limiting at negative peak displacement which may be caused by bottoming of the voice coil at the lower backplate.
- □ **MC b:** The first significant impulsive distortion is generated at maximal positive displacement (turning point) at 750 Hz sinusoidal excitation. The distortion are not generated by hard limiting of the moving capabilities of the diaphragm because the diaphragm can be moved to much higher values of the peak displacement at lower frequencies in other measurements using a higher terminal voltage.
- □ **MC c:** It very likely that the distortion are generated by voice coil rubbing because the instantaneous impulsive distortion are generated at the turning point of the displacement where the instantaneous acceleration signal becomes maximal within one period. Furthermore the distortions are generated at 750 Hz which is close to resonance frequency where the amplitude is maximal and close to the natural frequency (600 Hz) of the rocking mode.

11 Performing Measurements (Hardware required)

If the hardware of the KLIPPEL Analyzer System is available it is recommended to perform some of the measurements on a microspeaker provided from your side. You may use the measurements provided in the database microspeaker as a template but you should adjust the settings to the particular type to avoid a damage of the microspeaker and ensure optimal measurement results. Here are some tips how to customize the mesurements:

11.1 Creating a new Database

Step 22: At first it is recommended to copy the database provided by KLIPPEL and to rename the new database. Open the database and duplicate all objects in the database by clicking with the right mouse button on the each object and selecting ***"duplicate"***. In this way all the data are deleted but the settings are preserved. Select the old objects provided by Klippel and delete them.

11.2 Object LPM Measurement

Step 23: Open the object '***LPM Measurement*** and select the Operation ***LPM air center***. Perform the first small signal measurement to find the resonance frequency f_s. Adjust the bandwidth of the multi-complex in property page *Stimulus* according to the resonance frequency f_s to ensure an optimal identification of the small signal parameters.

Step 24: Adjust the voltage of the multi-tone stimulus in the property page *Stimulus*. Start with a small excitation voltage. After the measurement check the Spectrum *I(f)*, if no distortion or too low distortion are visible then increase the excitation level to improve the accuracy of the measurement.

Step 25: Check the result window ***"X(f)"*** spectrum. The signal lines should be constant below the resonance and decrease 12 dB per octave above the resonance and the SNR+D should be at least 20 dB. Check also the agreement between measured and fitted curves in the result window ***"Magnitude of Transfer Function"***. If the displacement is too low or the fitting does not agree, increase the averaging and/or the excitation level.

Step 26: After finding a reliable stimulus, select the operation ***LPM air north***, adjust the laser to point to the extreme north of the diaphragm and run the measurement. Do the same with other directions (south, west and east). After the measurement inspect the result window ***"X(t)"*** to analyse the rocking of the diaphragm.

11.3 Object Results: Microspeakers

Step 27: Open the Object '***Results: microspeakers*** and select the Operation ***2a LSI Nonlinear Parameters*** and on the property page *Im/Export* check the ***Force Factor*** button and enter the ***Bl*** value acquired from the LPM Measurement.

Step 28: In property page *Protection* set the internal gain G_{small} to -20dB and the P_{lim} very low to ensure that the driver will operate on safe. During the measurement check the polarity and increase P_{lim} step by step to find its working range. After the measurement, ensure that the fitting error is below 15%, thus the assumed model fits with the real microspeaker.

Step 29: All DIS measurement should be performed with an automatic temperature protection. Before running the measurement select the property page *Protection* and select in *Monitoring* the ***Frequency for test measurement*** at least two octaves away from the resonance frequency f_s. Choose ***increase of voice coil temperature (speaker 1) exceeds*** and enter 60 K for the *thermal protection.*

Step 30: Select the Operation ***3a DIS X Fundamental, DC*** and in the property page *Stimulus* adjust the frequency f_{start} << *fs* << f_{stop} and U_{end} to generate the displacement of X_{prot} from LSI ***"state"*** window. Use the voltage from LSI ***"state"*** window U_{rms} as a starting point. After the measurement, adjust the voltage in the case U_{end} exceeds or does not reach the X_{prot}.

Step 31: Select the Operation ***3b DIS Motor stability*** and set the frequency f_1 to 1.5**fs* and the maximal voltage to U_{end}.

Step 32: Before running a TRF Operation check always the ***Calibration*** of the microphone in the property page *Input*.

Step 33: In the Operation ***4a TRF SPL + Harm. (Usine)*** enter the voltage U_{end} and in the property page *Processing* check windowing: Impulse response must include all relevant signal energy.

Step 34: Select the Operation ***4b TRF Harmonics in current***. Set the voltage 2*U_{end} and make sure that f_{start} >> f_s. After the measurement copy the fundamental curve and import it as *Reference* curve (Processing tab).

Step 35: In the Operation ***4c DIS IM Dist. (voice sweep) P*** enter the voltage U_{end} and adjust bass tone f_2 < f_s (check the operation **3a** for a frequency which creates displacement of X_{prot}) and the voice tone f_1 >> f_s. Pay attention to the frequency settings: It is important that f_1 is at least 4.5 * f_2 to avoid 2nd/3rd order harmonics of f_2 being misinterpreted as intermodulation, f_{1end} should be at least 10*f_{1start}.

Step 36: Use the same setup for the Operation ***4d DIS IM Dist. (voice sweep) I*** but switch in the property page *Display* the *State Signal* for ***Current Speaker 2***.

Step 37: Check the microphone calibration before running the Operation ***4e LPM Multitone Dist. (Umt)*** and enter the voltage U_{rms}. Note that the warning of too low SNR in current can be ignored.

11.4 Object Systematic R&B Test Tweeter

Step 38: Open the Object '***Systematic R&B Test Tweeter*** and ensure that the maximal voltage is permissible for the microspeaker under test. Select the object and press start button . Pay attention to significant rub & buzz distortion, if it occurs, cancel the measurement at any time.

12 Further Literature

User Manual for the KLIPPEL R&D SYSTEM – *Linear Parameter Measurement*

User Manual for the KLIPPEL R&D SYSTEM – *Large Signal Identification*

User Manual for the KLIPPEL R&D SYSTEM – *3D Distortion Measurement*

User Manual for the KLIPPEL R&D SYSTEM – *Transfer Function*

Application Note AN 24 Measuring Telecommunication Drivers:
http://www.klippel.de/fileadmin/klippel/Files/Know_How/Application_Notes/AN_24_Telecomm_Driver.pdf
Application Note AN 25 Maximizing LPM Accuracy:
http://www.klippel.de/fileadmin/klippel/Files/Know_How/Application_Notes/AN_25_Maximizing_LPM_Accuracy.pdf
Application Note AN 32 Effective Radiation Area S_d:
http://www.klippel.de/fileadmin/klippel/Files/Know_How/Application_Notes/AN_32_Effective_Radiation_Area.pdf
Paper *Large Signal Performance of Tweeters*, *Microspeakers and Horn Drivers*:
http://www.klippel.de/uploads/media/Large_signal_performance_of_tweeters_05.pdf