Hands-On Training 4

Loudspeaker Distortion Measurements

1 Objective of the Hands-on Training

- Getting an overview of the important distortion measurement techniques
- Understanding the conditions required to generate dominant nonlinear distortions
- Investigating the influence of the stimulus and the role of the state variables
- Interpreting measurement results (spectral and temporal properties)
- Developing practical skills in performing an accurate measurement
- Optimizing the measurement setup

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 measurement results provided in the Klippel database *Training 4_Loudspeaker Distortion Measurements.kdbx*. For the purpose of this training, there is no requirement to have access to a Klippel R&D measurement system. The database 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:

- Transfer Function Measurement Module (TRF)
- 3D Distortion Measurement Module (DIS)
- Distortion Analyzer DA2
- Laser Sensor + Controller
- Amplifier
- Driver Stand

3 The Training Process

1. Read the introduction.
2. Watch the demo video.
3. Answer the preparatory questions.
4. Follow the instructions to interpret the results in the database and answer the multiple-choice questions off-line.
5. Submit your responses to the anonymous evaluation system at www.klippel.de/training.
6. Receive an email containing either a **Certificate of Mastery, a Certificate of Knowledge** or **a Certificate of Participation** (depending on your performance).
7. Perform some optional measurements on transducers if the hardware is available.

4 Introduction

Electro-dynamic loudspeakers have inherent nonlinearities which limit the acoustical output at higher input amplitudes and generate distortions in the reproduced sound. The generation of signal distortion can be modeled by a flow chart as shown in Figure 1 below.

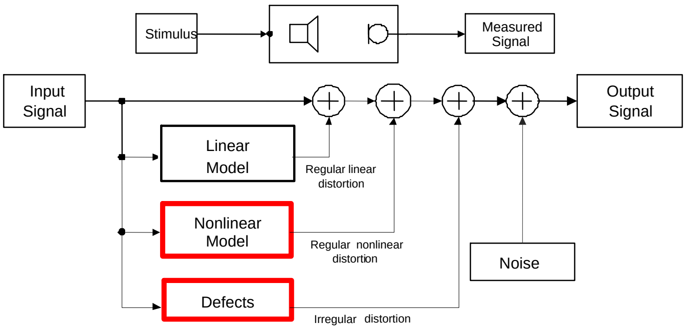

Figure 1: Flow chart modeling generation of signal distortion in audio system

The dominant nonlinearities, which are the principal causes of the nonlinear distortions, are located in the motor and in the suspension part of the electro-dynamic transducer. For example, nonlinearities exist when the voice coil displacement is relatively large compared to the dimensions of the coil-gap configuration and when the voice coil displacement is relatively large compared to the size of the corrugation rolls in the suspension (spider, surround).

Irregular distortions are mainly generated by defects caused during the manufacturing process The effects from ageing and other external factors, such as overload and climate, become evident during the later life cycle of the product.

Interpreting distortion to assess the quality of loudspeakers requires a variety of measurements, which will be explained in this training. The 3D-Distortion (DIS) and the Transfer Function (TRF) modules provide special features for making transducer measurements and detecting critical distortions.

The purpose of these measurements is to identify typical symptoms which can be correlated to individual nonlinear characteristics of the loudspeaker.

4.1 Test Stimulus

A nonlinear system excited by a stimulus generates an output signal which exhibits symptoms of the nonlinearities. Since the nonlinearities of the motor and suspension are only activated at higher excitation levels, the loudspeaker behaves almost linearly for sufficiently small amplitudes of the input stimulus. There is very little voice coil displacement and nearly all nonlinear symptoms are not evident when the loudspeaker is working in the small signal domain. The dependency on the input signal amplitude and the resulting voice coil displacement is a reliable indicator of nonlinearities.

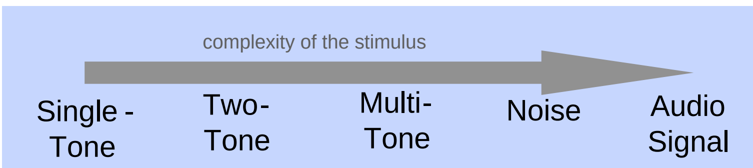

Figure 2: Stimulus for nonlinear distortion measurements

The generation of nonlinear distortion is a complex multidimensional process. A single tone excitation will not necessarily activate all nonlinearities and, as a result, only a subset of the nonlinear symptoms will be detected.

On the other hand, a more complex stimulus signal will require a more complex evaluation of the measurement results. Therefore, it is sensible to use different stimuli for specific measurements.

As shown in Figure 3, the simplest test stimulus is a sinusoidal tone of a defined frequency f_1 that can be discretely stepped or continuously swept through the audio band. The acoustic output signal of the device under test is captured by a sensor (e.g. microphone) and a spectral analysis is performed.

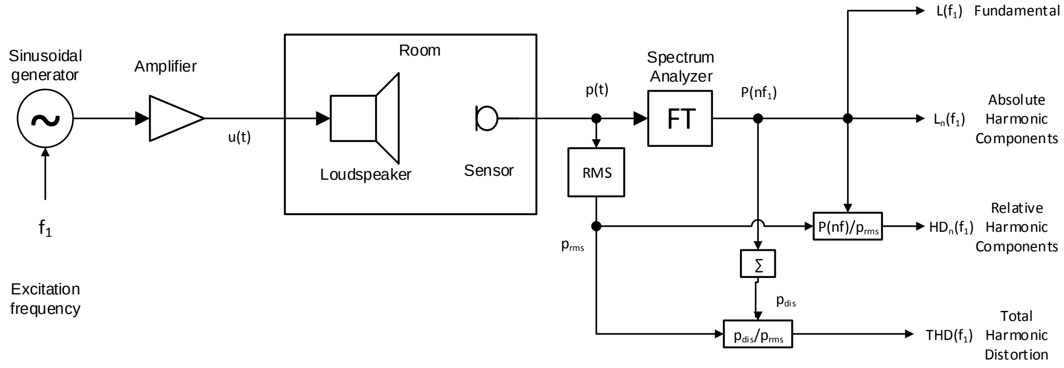

Figure 3: Measurement of harmonic distortion using a sinusoidal input at excitation frequency f_1

Figure 4 shows the spectrum $P(f)$ in the acoustic output signal $p(t)$ generated by input signal $u(t)$. The fundamental component $P(f_1)$ is located at the excitation frequency f_1 of the sinusoidal input. The harmonic tones are additional unwanted frequency components $P(nf_1)$ with $n > 1$ that are multiples of the excitation frequency f_1. The amplitude of each harmonic distortion component can be expressed in absolute or relative values. Absolute values are usually expressed in dB's of sound pressure level $L_n(f_1)$ compared to the reference sound pressure 20 uPa. Relative values are expressed in dB's relative to the rms value p_{rms} of the total acoustic output signal p(t). The Total Harmonic Distortion THD is defined as the ratio between the rms value $P(nf)$ of all harmonic distortion components ($n \geq 2$) referred to the rms value p_{rms} of the output signal.

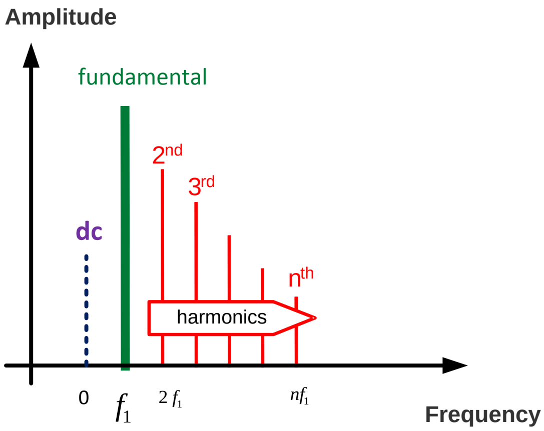

Figure 4: Spectrum of an acoustic output signal showing harmonics produced from nonlinearities excited by a single tone stimulus at frequency f_1

Harmonic distortion components found in the output spectrum indicate that nonlinearities are inherent in the device under test. However, the measured output harmonics using a single tone input stimulus are not a comprehensive description of the nonlinear system.

4.2 Compression of the Fundamental

In the large signal domain, the thermal and nonlinear effects limit the acoustic output of the transducer. Thus, the amplitude of the fundamental displacement component in the laser signal does not change proportionally with the amplitude of the sinusoidal input voltage at the speaker terminals. This is shown in Figure 5.

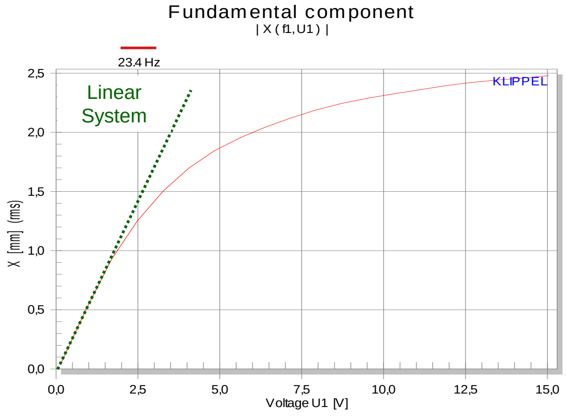

Figure 5: Fundamental compression

At low input voltages, there is almost a linear relationship between input voltage and voice coil displacement amplitudes as required for linear parameter modelling. At higher input voltage amplitudes, the nonlinearities in the motor and suspension cause a limiting of the voice coil displacement amplitude which limits the maximal acoustic output.

4.3 Generation of a DC-Component

As shown in Figure 6, asymmetries in the motor and suspension nonlinearities generate a DC offset component in the voice coil displacement, which can be detected by a laser sensor. For example, an asymmetrical stiffness $K_{ms}(x)$ characteristic generates a DC component, which always shifts the coil towards the softer side of the suspension. For excitation frequencies above resonance, an asymmetrical force factor $Bl(x)$ characteristic generates a significant DC component, which shifts the coil away from the $Bl(x)$ maximum. Designs that combine a steep flanked nonlinear force factor characteristic $Bl(x)$ with a low suspension stiffness can produce instable driver behaviour, which increases the risk of a substantial DC generation.

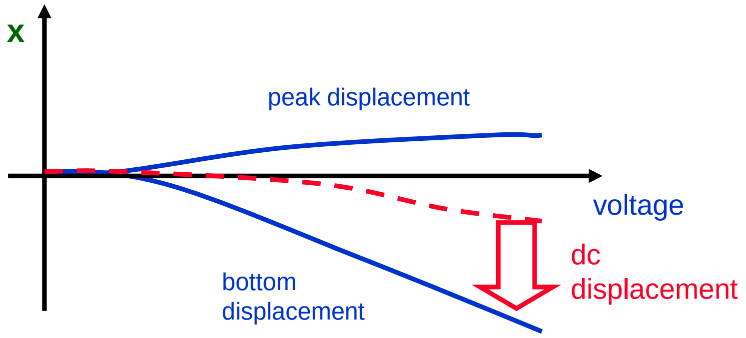

Figure 6: DC displacement generated by an asymmetrical force factor characteristic

This DC component is very critical because it changes the operating point for all displacement dependent nonlinearities. Thus, the generation of a DC displacement by one nonlinearity can activate the generation of more nonlinear symptoms from other nonlinearities.

4.4 Harmonic Distortion Interpretation

The harmonic distortion is usually plotted versus the excitation frequency f_1 and it may be displayed in absolute or relative values. For example, the amplitude of a third order harmonic measured at $3f_1$, which has been generated by a sinusoidal excitation at f_1, will be plotted at location f_1 in the 3^{rd} order harmonics graph. However, the interpretation of the harmonic responses are not always this simple.

Figure 7 shows the absolute distortion of a typical loudspeaker displaying the fundamental, 2nd-order and 3rd-order harmonics versus excitation frequency measured under two different acoustical environments.

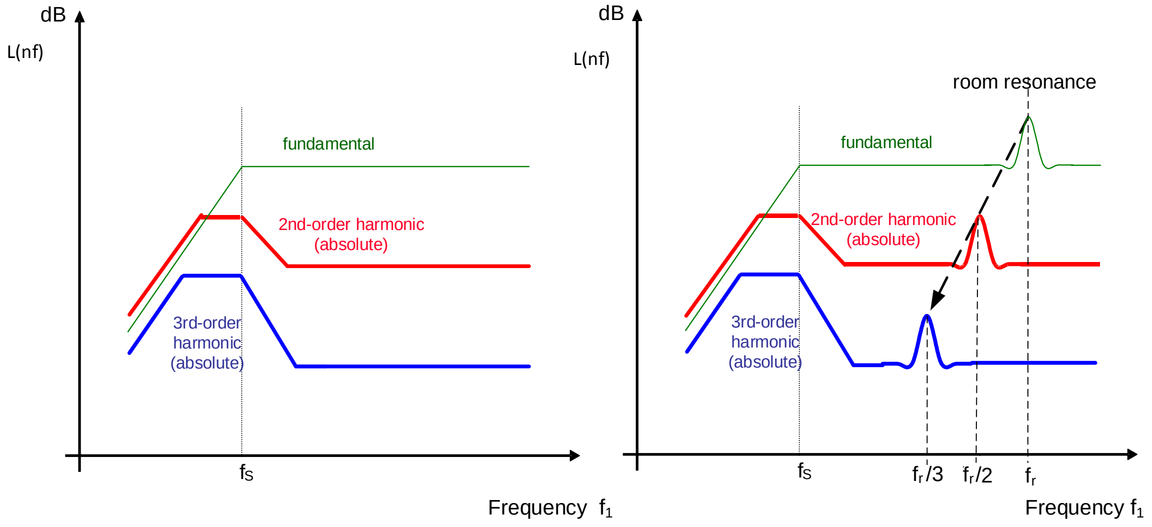

Figure 7: Absolute harmonic distortion versus excitation frequency f_1 of the same loudspeaker measured under free field conditions (left) and in a listening room (right)

The left picture shows the response of the fundamental and harmonic distortion of a loudspeaker operated in free field (anechoic room) while the right picture shows the responses of the same loudspeaker measured in a reverberant environment with a room resonance at frequency f_r.

The room behaves like a linear post-filter which enhances all signal component at the room resonance frequency f_r. This creates a peak in the SPL response of the fundamental component at the excitation frequency $f_1 = f_r$. The responses of the nth-order harmonic distortion components also have peaks but they occur at lower excitation frequencies $f_1 = f_r/n$. This is because the nth-order harmonic distortion components, as found in the free field output spectrum, pass via the room's linear post-filter at multiples of the excitation frequency. This example shows how the linear transfer behavior of the acoustical environment increases the complexity of the interpretation of the harmonic responses. It is recommended to place the microphone in the near field of the loudspeaker where the room influence is negligible and the measurement can be performed at a high SNR ratio.

Figure 8 shows the same loudspeaker and acoustic conditions as found in Figure 7. However, the nth-order harmonic distortion components $P(nf_1)$ are plotted relative to the rms value $p_{rms}(f_1)$ of the output signal (including fundamental component) for an excitation frequency f_1.

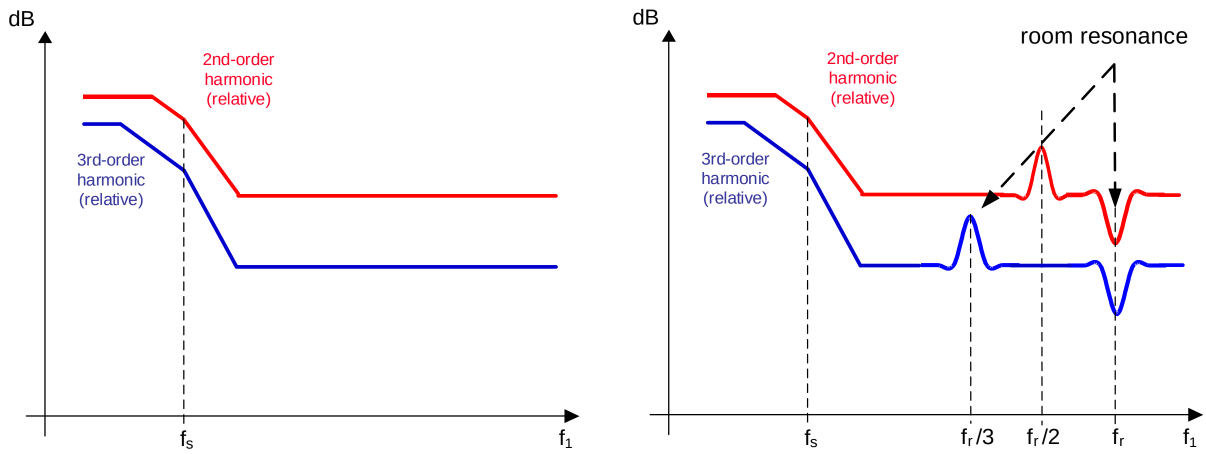

Figure 8: Relative harmonic distortion versus excitation frequency

The room resonance causes an additional dip in the nth-order harmonic distortion components at frequency f_r because the level of the fundamental component is increased at this frequency while the absolute harmonic distortion is constant. This example shows how the calculation of relative distortion increases the complexity of the harmonic responses and makes the interpretation more difficult.

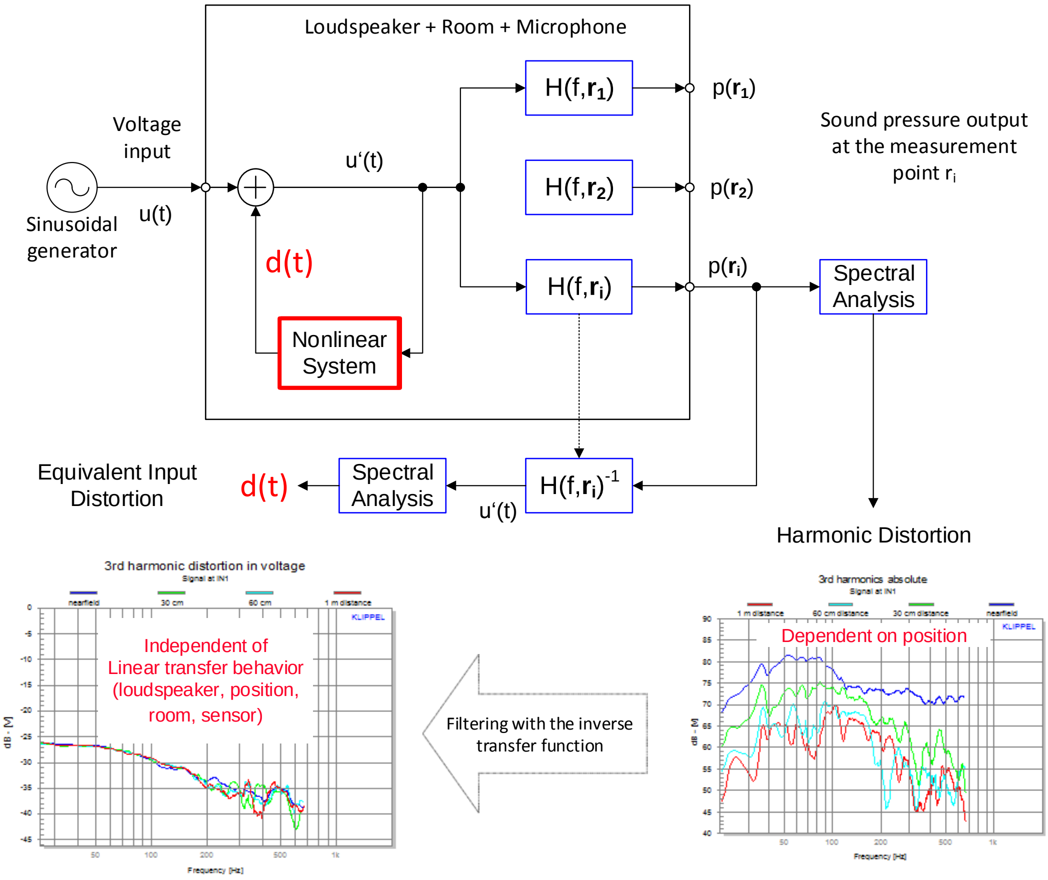

Figure 9: Measurement of equivalent harmonic input distortion

4.4.1 Equivalent Harmonic Input Distortion

The effects shown in Figure 7 make it very hard to separate the pure harmonic distortion characteristics of the loudspeaker from the distortion characteristics of the linear post-shaping in the room. The Equivalent Input Distortion (EHID) measurement simplifies the interpretation of the harmonic distortion by removing the linear post-shaping from the measurement.

The dominant nonlinearities such as force factor $Bl(x)$, inductance $L(x)$ and mechanical suspension stiffness $K_{ms}(x)$ are located in the electrical and mechanical domains of the transducer. As illustrated in Figure 9, these nonlinearities can be modelled by a nonlinear system adding distortion $d(t)$ to the input signal $u(t)$ and generating a distorted input signal $u'(t)$. The distorted signal $u'(t)$ is transferred via a linear system with the transfer function $H(f,r_i)$ to the sound pressure output $p(r_i)$ at the measurement point r_i. This linear system represents modal cone vibrations, sound radiation/propagation, the acoustical environment (e.g. the room) and the properties of the sensor (e.g. microphone and/or laser). Although the nonlinear distortions are generated by a single source, the measured harmonic distortions depend on the position of the measurement point r_i as illustrated in the lower right corner of Figure 9. The high complexity of the distortion curves makes it difficult to identify the root cause of the distortion.

The distorted signal $u'(t)$ can be calculated by applying a linear filter with the inverse transfer function $H(f,r_i)^{-1}$ to the measured sound pressure signal $p(r_i)$. The Equivalent Harmonic Input Distortion (EHID) can be determined by performing a spectral analysis of the distorted signal $u'(t)$. The EHID describes the distortion signal $d(t)$ at the output of the nonlinear system which simplifies the interpretation significantly.

4.5 Intermodulation Distortion Measurement

Loudspeakers and other electro-acoustical transducers generate significant intermodulation distortions in the audio band, which have a significant impact on the perceived sound quality. Distortion measurements using a single sinusoidal tone cannot produce these intermodulation components. As shown in Figure 10, a simple two tone signal, comprised of a variable tone f_1 and a second tone with constant frequency f_2 applied to a nonlinear system, will generate nth-order intermodulation components at the difference frequencies $(f_2 - (n - 1)f_1)$ and summed frequencies $(f_2 + (n - 1)f_1)$ for $n = 2, 3, \ldots$

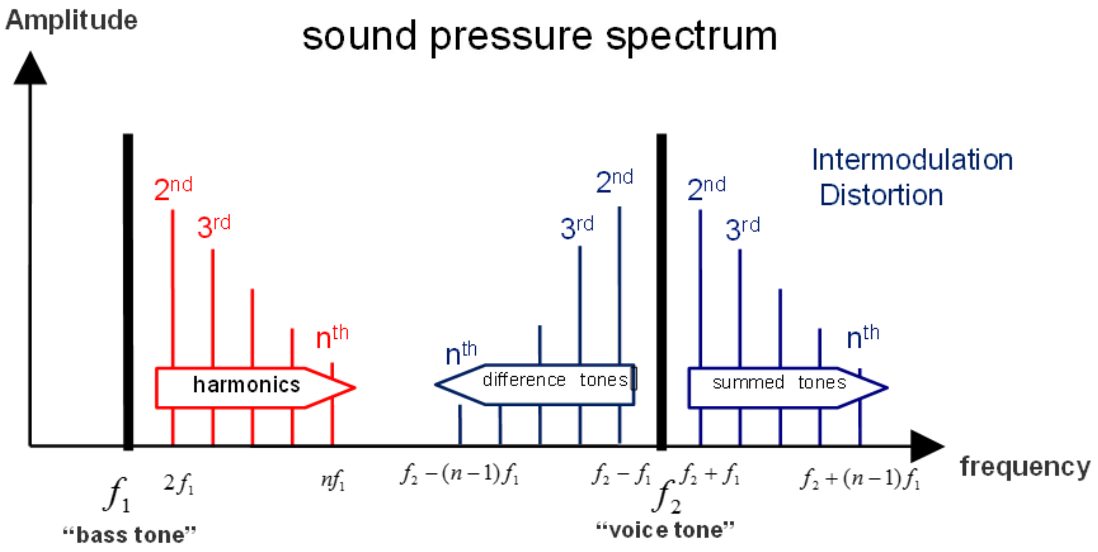

Figure 10: Intermodulation components

Intermodulation distortion is a critical symptom of the motor nonlinearities $Bl(x)$ and $L_e(x)$. IMD is also a critical symptom of Doppler Effect, which is an acoustical radiation nonlinearity. Exciting the loudspeaker with a two-tone signal produces both amplitude and phase (frequency) modulation distortion.

4.5.1 Amplitude and Phase Modulation

Amplitude **M**odulation (AM) only varies the instantaneous amplitude (envelope) of the voice tone but does not change its phase. The temporal variation of the envelope of the high-frequency signal generates a fluctuation and roughness in the perceived sound.

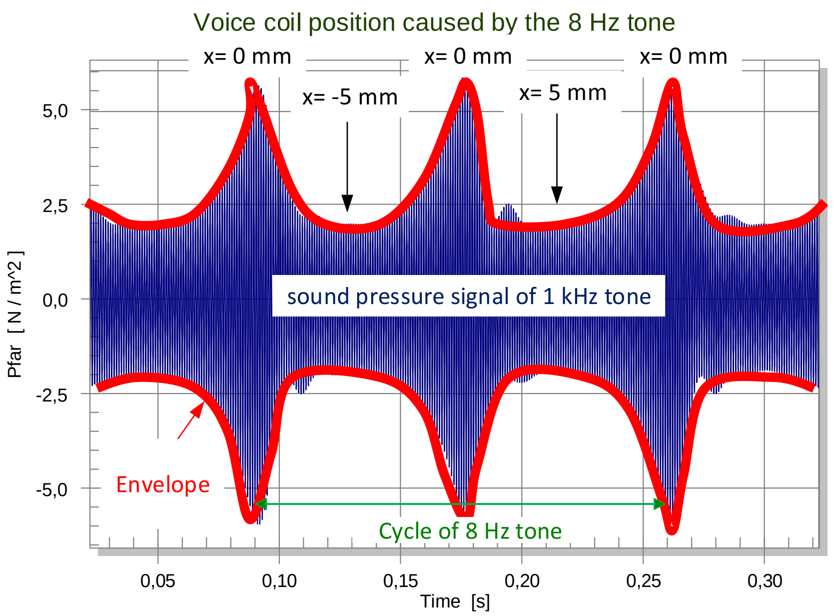

Figure 11: Amplitude Modulation (AM) generated by a nonlinear force factor *Bl*(*x*) when a two-tone signal (f_1=8 Hz and f_2=1 kHz) is applied

As shown in Figure 11, the characteristic force factor *Bl*(*x*) versus displacement *x* creates significant amplitude modulation of the high frequency tone f_1. If the voice coil is at the rest position $x = 0$, the instantaneous value of *Bl*(*x*=0) is maximal and generates the peak in the envelope of the high frequency tone. For negative and positive peak excursions where the voice coil is outside the gap the *Bl*-value reduction generates the bottom values of the envelope.

Unlike amplitude modulation, the phase or frequency modulation (FM) does not change the envelope of the voice tone but varies the instantaneous phase.

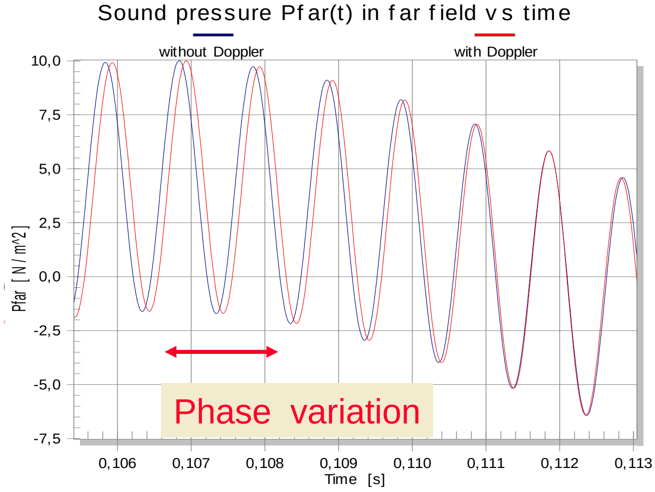

Figure 12: Frequency Modulation (FM)

The Doppler Effect causes mainly phase modulation due to the time delays associated with the changing distance between the moving diaphragm and the fixed listening point.

4.5.2 Useful Techniques for Intermodulation Measurements

There are two important setups for varying the excitation tone: “bass tone sweep” or “voice tone sweep”. In both cases, the frequency of one tone is constant while the frequency of the other tone is changed.

Figure 13 shows the recommended setup for the bass tone sweep technique. The frequencies are chosen for woofers as follows:

- f_1 is a varying frequency bass tone around the resonance: $0.5f_s < f_1 < 2f_s$
- f_2 is a constant frequency voice tone above the resonance: $f_2 \approx 7.5f_s$

The varying bass frequency f_1 causes a significant change in voice coil displacement and input current, which activates the dominant loudspeaker nonlinearities. The frequency f_2 represents a typical audio signal in the pass band where the generated sum and difference tones are clearly separated from the harmonics of the bass tone f_1.

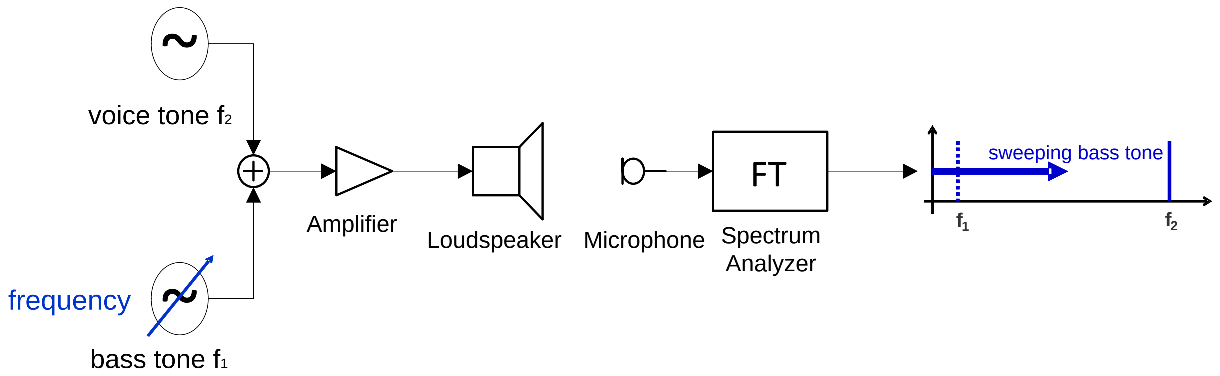

Figure 13: IMD measurement setup using the bass tone sweep method

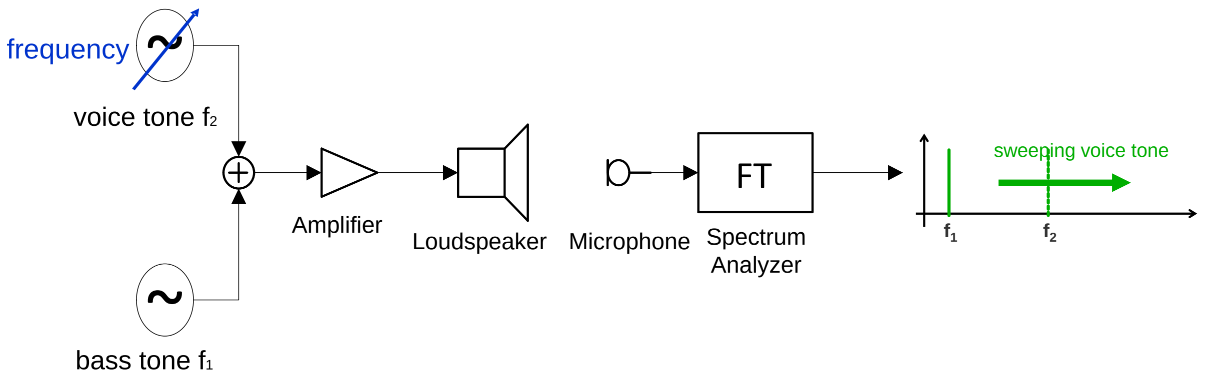

Figure 14: IMD measurement setup using the voice tone sweep method

Figure 14 shows the recommended setup for the voice tone sweep technique. The frequencies are chosen for woofers as follows:

- f_1 is a constant frequency bass tone below the resonance: $f_1 \approx 0.5f_s$
- f_2 is a varying frequency voice tone over the audio band with $8f_s < f_2 < 20f_s$

The voice tone sweep method ensures an almost constant peak voice coil displacement while allowing sufficient separation between the bass tone f_1 harmonics and the voice tone f_2 sum and difference tones. As the voice tone is swept the total IMD measurement varies. Thus, the voice sweep method reveals the frequency dependency of the intermodulation distortion on the frequency of the signal components in the audio band.

4.6 Multi-tone Distortion Measurement

The multi-tone complex signal is comprised of a multitude of tones at known frequencies. During the measurement, each tone generates harmonics. In addition, the tones interact providing a variety of difference and summed tones similar to an actual audio music signal. For this reason, the multi-tone complex is used to simulate the normal working conditions of a loudspeaker.

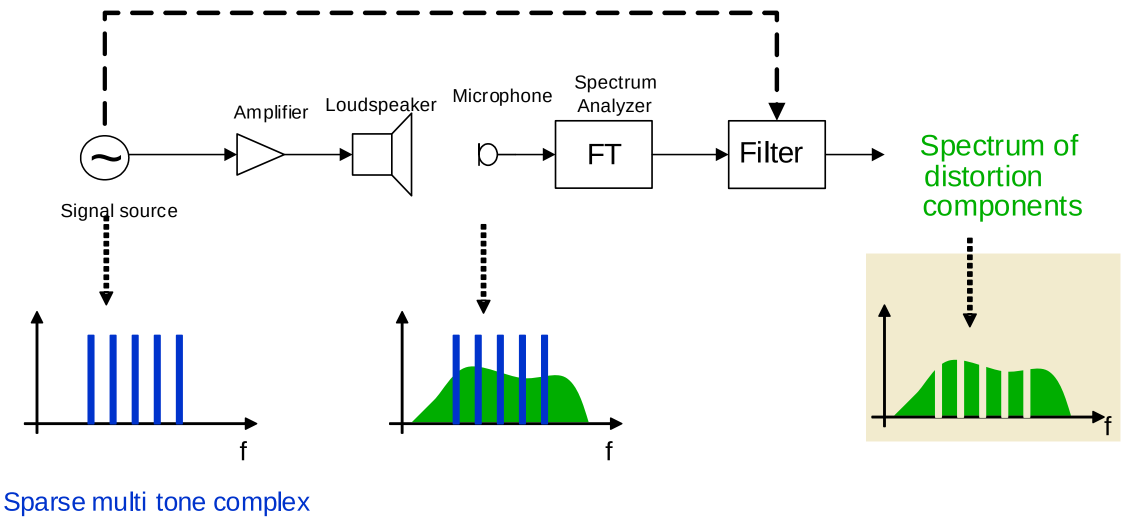

Figure 15: Multi-tone distortion measurement

When using the multi-tone complex signal, the nonlinear distortions are detected between the frequency tones. Therefore, the multi-tone signal has a sparse spectrum. The ambient noise floor can be measured by performing an additional measurement without any excitation signal. The bandwidth, shape and density of the multi-tone complex signal can easily be adjusted to satisfy the target application of a particular transducer or a complete loudspeaker system. It is important to note that the distortion spectrum resulting from this measurement reveals no diagnostic information regarding the symmetrical and asymmetrical shapes of the nonlinearities.

4.7 Rub & Buzz Measurement

In order to easily separate the excitation tone from the generated impulsive distortion, a single tone or a sinusoidal sweep is the preferred stimulus for exciting the distortions associated with irregular defects such as voice coil rubbing, buzzing, loose particles, air leakage and hard limiting of the suspension or voice coil former.

The distortions created by irregular defects are more impulsive than the nonlinear distortions generated by the regular nonlinearities in the motor, suspension and diaphragm. In a conventional total harmonic distortion (THD) measurement, the harmonic distortion generated by the regular nonlinearities are dominant. As a result, the irregular distortions cannot be detected in the measurement. In addition, the distortion from irregular defects may be audible even if the rms value of their higher order components ($n > 20$) are below the noise level in a conventional THD measurement.

4.7.1 Impulsive and Higher-order Distortion

The irregular defects listed in 4.7 above generate distortion having a particular impulsive fine structure in the time domain as shown in distortion signal $d(t)$ in Figure 16. The impulsive behavior is evident in higher-order harmonics and other signal components which are initiated and synchronized by the nonlinear defect. Thus, both amplitude and phase information of the distortion signal provide valuable clues in determining the source of the defect. Since the interpretation of a complex spectrum is difficult, it is more useful to analyze the high-pass filtered time signal $d(t)$ shown in Figure 16. Since irregular distortion is basically seen in higher frequencies, the lower order distortion components are separated from the original microphone signal by using a high-pass tracking filter.

At the output of the tracking filter some important measures are derived from the impulsive distortion signal $d(t)$:

- Peak value of higher-order distortion (PHD)
- RMS value of higher-order distortion (MHD)
- Crest factor of higher-order distortion (CHD)
- Instantaneous crest factor of higher-order distortion (ICHD)

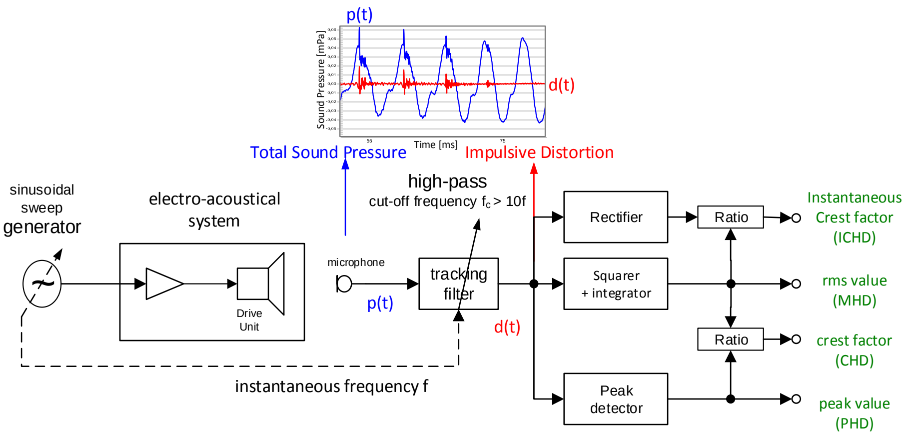

Figure 16: Measurement of defects by time domain analysis

4.7.2 Peak value of Impulsive Distortion

The most important characteristic of the impulsive distortion is the peak value (PHD). The rms-value (MHD) corresponds with the power of the irregular distortion while the peak value is a more sensitive measurement to show the impulsive symptoms in the fine structure of the waveform.

The PHD critical level is best evaluated relative to the loudspeakers passband sound pressure level.

Figure 17 shows the absolute PHD value exceeding a PHD limit set to 40 dB below the fundamental mean sound pressure level in the passband. The frequency band between 40 and 70 Hz shows significant distortion.

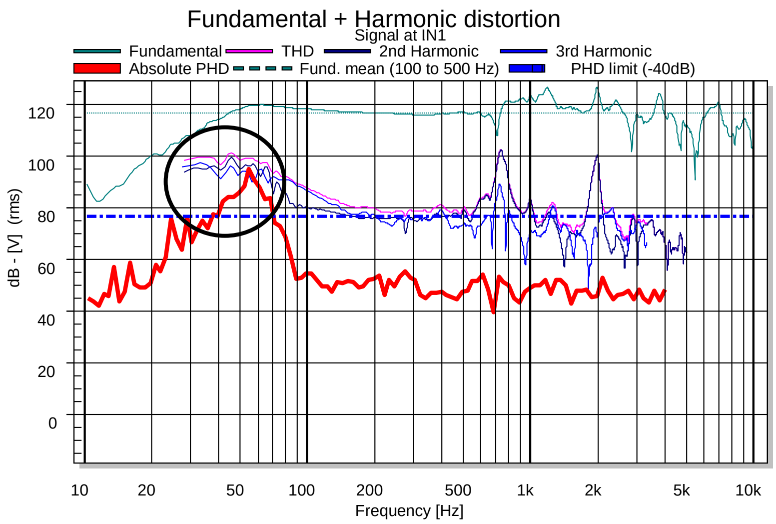

Figure 17: Absolut PHD (red thick line) and PHD limit (blue dashed line)

4.7.3 Crest Factor of Higher-order Distortion

The **C**rest factor of the **H**igher-order **D**istortion is a relative measure which describes the ratio between the peak value and the rms value of the distortion within one period of the fundamental. This value describes the impulsiveness of the distortion signal. A constant DC signal has a crest factor of 0 dB. A sinusoidal signal has a crest factor of 3 dB. A distortion generated by regular nonlinearities or measurement noise usually reaches a crest factor of 12 dB. Typical irregular distortions generated by loudspeaker defects can have crest factors larger than 12 dB.

The **I**nstantaneous **C**rest **H**igher-order **D**istortion (ICHD) describes the ratio between the absolute value instantaneous distortion value |d(*t*)| and rms value MHD. This measure is useful for investigating the fine structure of the impulsive distortion. When mapped to state variables, such as voice coil displacement or sound pressure output, valuable information about the physical cause (bottoming, coil rubbing, compression …) of the distortion can be determined.

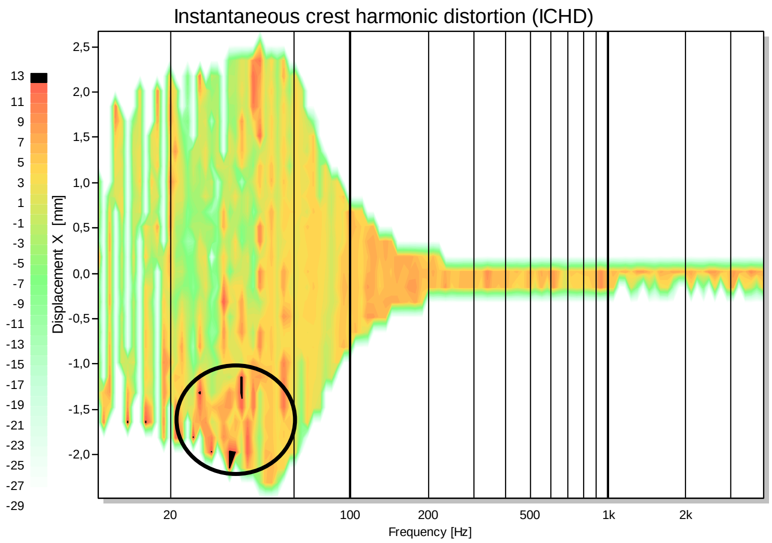

Figure 18: Instantaneous crest factor of higher-order distortion (color) versus frequency and voice coil displacement

Figure 18 shows the ICHD value as a function of the voice coil displacement (vertical axis) and the instantaneous frequency, which is correlated with the sweep time of the logarithmic chirp used as the stimulus. As shown in Figure 18, the black spots at negative excursions for the excitation frequencies between 20 and 50 Hz indicating a bottoming of the voice coil former against the back plate.

5 Preparatory Questions

Check your theoretical knowledge before you start Section 6. Answer the questions by selecting all correct responses (sometimes, there will be more than one).

QUESTION 1: Does the generation of the nonlinear distortion components depend on the frequency of the excitation tone(s) in the stimulus?

- □ **MC a:** Yes, the generation of the nonlinear distortions (harmonics, intermodulation and dc components) depend on the frequency of the excitation tone because the loudspeaker is a dynamic nonlinear system.
- □ **MC b:** No, the generation of the nonlinear distortions (harmonics, intermodulation and dc components …) are independent of the frequency of the excitation tone because the loudspeaker is a static nonlinear system without any memory (like a diode with a nonlinear transfer characteristic between input and output).

QUESTION 2: Does the generation of the nonlinear distortion components depend on the amplitude of the excitation tone(s) in the stimulus?

- □ **MC a:** No, the amplitude of the nonlinear distortion components generated in the output signal are independent of the amplitude of the spectral components in the stimulus.
- □ **MC b:** Yes, the amplitude of the nonlinear distortion components always increase when the amplitude of the spectral components in the stimulus are increased.
- □ **MC c:** Yes, the amplitude of the nonlinear distortion components may increase or decrease when the amplitude of the spectral components in the stimulus are increased.

QUESTION 3: Do the loudspeaker nonlinearities affect the fundamental component?

- □ **MC a:** No, the loudspeaker nonlinearities produce only harmonic distortions which are multiples of the fundamental component. The amplitude and phase of the fundamental component in the output signal can be calculated by multiplying the linear transfer function $\underline{H}(j\omega)$ with the fundamental in the stimulus.
- □ **MC b:** Yes, the fundamental component is also affected by the loudspeaker nonlinearities. In most cases, the measured amplitude of the fundamental in the acoustic output of a nonlinear loudspeaker is less than the predicted output using the linear transfer function. However, at the resonance frequency, the nonlinear loudspeaker may produce more output of the fundamental component due to the loss of electrical damping.

QUESTION 4: What does the measured DC component in the voice coil displacement reveal?

- □ **MC a:** A DC component is generated by asymmetries in the nonlinear parameters (motor and suspension nonlinearities) causing a partial rectification of the AC displacement. The sign of the DC component (indicating voice coil movement towards or away from the back plate) gives information about the orientation of the shape of the asymmetry.
- □ **MC b:** A DC component may also be generated by loudspeaker instabilities even when the loudspeaker nonlinearities are symmetrical.
- □ **MC c:** A DC component in the voice coil displacement is a symptom of loudspeaker nonlinearities but has no diagnostic value because a DC component is not audible.

QUESTION 5: Does the harmonic distortion measurement describe the nonlinear transfer behaviour of the loudspeaker completely?

- □ **MC a:** Yes, provided harmonic distortions are measured by using a sinusoidal sweep (chirp) covering the complete audio band.
- □ **MC b:** Yes, provided all orders of the harmonic distortion and the nonlinear compression of the fundamental are measured as a function of the frequency and amplitude of the sinusoidal stimulus.
- □ **MC c:** No, because a single tone stimulus cannot generate intermodulation distortion. IMD requires a stimulus comprised of more than one spectral component (two-tone).

QUESTION 6: Does the frequency response of the absolute harmonic distortion have the same curve shape as the relative harmonic distortion (as defined by IEC and other standards)?

- □ **MC a:** Yes, the curve shapes are identical when the relative harmonic distortion is expressed in dB (for example: -40 dB corresponds with 1 % distortion).
- □ **MC b:** No, the relative harmonic distortion is the amplitude of the absolute distortion referred to the amplitude of the total signal. Thus, the frequency response of the relative harmonic distortion will have a different curve shape.

QUESTION 7: How do you detect rub & buzz and other irregular loudspeaker defects reliably?

- □ **MC a:** The measurement of the 2^{nd}-order and 3^{rd}-order intermodulation components using a two-tone stimulus is a reliable technique for measuring rub & buzz.
- □ **MC b:** The measurement of the total harmonic distortion (THD) considers the amplitudes of all harmonic components. The THD generated by rub & buzz and other irregular loudspeaker defects is much less than the THD generated by the regular nonlinearities inherent in the motor and suspension. As a result, to detect the unique symptoms of irregular defects, an additional high-pass filter is required to suppress the low-order harmonic distortion components generated by regular loudspeaker nonlinearities.
- □ **MC c:** The irregular loudspeaker defects generate impulsive distortion components having a high crest factor (high ratio of peak to rms value). Therefore, a time domain analysis is required to consider the amplitude and phase of the distortion components at high frequencies.
- □ **MC d:** A multi-tone distortion measurement is a reliable technique for detecting rub &buzz provided the stimulus is comprised of a sufficient number of tones logarithmically distributed over the audio band.

6 Interpretation of Distortion Measurements (no hardware required)

Step 1: View the demo movie *Loudspeaker Distortion Measurements* located at www.klippel.de/training to see how practical distortion measurements are performed.

Step 2: Run the Software *dB-Lab* and open the file *Loudspeaker Distortion Measurements.kdbx*

Advice: Before submitting your answers, it is recommended to do the following exercises offline!

6.1 Harmonic Distortion Measurement

6.1.1 2^{nd}- and 3^{rd}- Order Distortion

Step 3: Open the operation ***1a TRF SPL Fund + Harm 1.5V***, which shows a transient measurement of the harmonic distortion of the transducer by using a sinusoidal sweep where the instantaneous frequency increases logarithmically over time. In the result window "***Fundamental + Harmonics***" compare the curves *2^{nd} Harmonic* with the *Total Harmonic Distortion (THD)***.**

QUESTION 8: In which frequency range is the absolute 2^{nd}-order distortion component dominating the THD?

- □ **MC a:** Below 40 Hz
- □ **MC b:** $40 < f < 250$ Hz
- □ **MC c:** For all frequencies above 250 Hz

QUESTION 9: In which frequency range is the absolute 3^{rd}-order distortion component dominating the THD?

- □ **MC a:** Below 40 Hz
- □ **MC b:** $40 < f < 250$ Hz
- □ **MC c:** For all frequencies above 250 Hz

Step 4: In the same operation ***1a TRF SPL Fund + Harm 1.5V*** look at the result window "***Y1 (f) Spectrum***" and compare the *Signal lines* (blue), which is the reproduced stimulus, with the *Noise floor*, found in the microphone signal with muted stimulus (black)**.**

QUESTION 10: Is the measurement of 2nd- and 3rd-order harmonic distortion for excitation frequencies between 40 and 250 Hz corrupted by steady-state noise (e.g. generated by air conditioning)?

- □ **MC a:** No, the measurement is not corrupted by steady-state noise because the signal to noise ratio (SNR) is greater than 40 dB in the frequency range 80 Hz < f < 750 Hz where the 2nd- and 3rd- order harmonic distortions are being measured.
- □ **MC b:** Yes, the signal to noise ratio (SNR) is about 40 dB at 40 Hz which is not sufficient for measuring the harmonics at 40 Hz.

Step 5: Open the operation ***1b TRF SPL Fund + Harm 6V***, which shows the harmonic distortion of the same transducer using the same measurement setup but at a much higher sinusoidal stimulus voltage (6 V instead of 1.5 V). In the result window "***Fundamental + Harmonics***" observe the *3rd Harmonic* and answer the following question:

QUESTION 11: What is the physical reason for the absolute level of the total harmonic distortion to be decreasing at frequencies above resonance ($f_s < f < 4f_s$)?

- □ **MC a:** The sound pressure output is almost constant in this frequency range.
- □ **MC b:** The electrical input current increases from resonance (120 Hz) to the minimum of the electrical impedance curve (450 Hz).
- □ **MC c:** The displacement decreases above the resonance frequency by 12 dB per octave.

6.1.2 Displacement

Step 6: Open the operation ***2a TRF X Fund + Harm 6V*** and inspect the frequency range from 150 Hz to 1.5 kHz in the result windows "***Fundamental + Harmonics***", "***Harmonic Distortion (relative)***" and "***Y2 (f) Spectrum***".

QUESTION 12: What causes the increase of the relative total harmonic distortion, shown as curve *THD in X*, in result window "***Harmonic distortion (relative)***" in this frequency range?

- □ **MC a:** The harmonic distortion measurement is corrupted by measurement noise. As shown in the result window "***Y2 (f) Spectrum***" of the displacement, the measured voice coil displacement disappears into the noise floor at 1 kHz. The resulting low signal-to-noise ratio (SNR) causes an almost constant value of absolute distortion (-50 dB) in this frequency range. The relative distortion, which is the ratio between absolute distortion and total displacement, increases because the total displacement decreases by 40 dB.
- □ **MC b:** The loudspeaker generates more distortion at higher frequencies.
- □ **MC c:** The laser sensor is limiting and generates this distortion.

Step 7: In the same operation ***2a TRF X Fund + Harm 6V*** open the result window "***Harmonic Distortion (relative)***". Compare the curve *THD in X*, representing the relative THD found in the displacement signal, with the curve *THD in SPL*, representing the relative THD found in the sound pressure signal (this curve has been copied from result window "***Harmonic Distortion (relative)***" in the operation ***1b TRF SPL Fund + Harm 6V***.

QUESTION 13: Why is the relative THD measured in sound pressure higher than the relative THD measured in displacement at low frequencies (f < 150 Hz)?

- □ **MC a:** The radiation and propagation of the sound is highly nonlinear and increases the distortion at low frequencies.
- □ **MC b:** The amplitudes of both relative measurements depend on the frequency response of the fundamental component. Below resonance, the fundamental of the SPL measurement decreases but the fundamental of the displacement measurement is almost constant.
- □ **MC c:** The noise generated by the microphone increases the THD in the acoustical measurement.

Step 8: Open the operation ***2b DIS X Fund., DC, Short*** and inspect the result window "***DC Component***" which shows displacement versus frequency.

QUESTION 14: Is the DC Component independent of the frequency of the sinusoidal stimulus?

- □ **MC a:** No, the direction of the DC Component depends not only on the frequency but also on the shape of the nonlinearity. In this case the DC Component is maximal at the resonance frequency.
- □ **MC b:** No, the direction of the DC Component depends not only on the frequency but also on the shape of the nonlinearity. In this case the DC Component is maximal at frequencies below 50 Hz and above 300 Hz.
- □ **MC c:** Yes. The DC Component is always maximal at low frequencies, where there is more displacement.

Step 9: Open the operation ***2c DIS X Motor stability*** and inspect the result window "***DC Component***" which shows displacement *versus* voltage.

QUESTION 15: How does the dc displacement vary versus voltage?

- □ **MC a:** The DC displacement becomes less at higher voltages because the progressive suspension produces a symmetrical increase in the stiffness at positive and negative displacements, which improves the stability of the loudspeaker.
- □ **MC b:** The DC displacement is relatively small at low voltages but increases rapidly at critically higher voltages. This behaviour reveals instability of the electro-dynamic motor.
- □ **MC c:** The DC displacement increases slowly with rising voltage. This behaviour is typical for a loudspeaker which is stable but has significant asymmetries in the nonlinear parameters.

6.1.3 Equivalent Input Distortion

Step 10: Select the operation ***3a TRF SPL EHID 6 V.*** Open the property page ***Processing*** and ensure that the checkboxes *Curve* and *Level* in section *Reference* are disabled. Open the result window "***Fundamental + Harmonics***". *C*opy the curve *Fundamental* onto the clipboard and paste it into the EDIT button found in section *Reference* on the property page ***Processing***. The imported curve is used for inverse filtering of the microphone signal. The response of the fundamental component should now become almost flat. By doing this, the nonlinear distortion at the sensor (e.g. microphone) has been transformed to the input of the loudspeaker (e.g. electrical terminals).

QUESTION 16: Why is the calculation of the equivalent input distortion useful?

- □ **MC a:** The dominant nonlinear distortions generated by motor and suspension nonlinearities are generated in the one-dimensional signal domain close to the electrical input. These nonlinearities produce the same amount of equivalent harmonic input distortion (EHID) as measured by the microphone (independent of the microphone position). If the EHID measurement shows a dependency on the microphone position, the nonlinearities are located in the multi-dimensional signal path (e.g. cone vibration or sound radiation).
- □ **MC b:** The amplitude response of the microphone has no influence on the Equivalent Harmonic Input Distortion (EHID) measurement because the inverse filtering removes the linear properties of the fundamental and the linear properties of the microphone.
- □ **MC c:** The influence of room reflections and room modes are removed from the Equivalent Input Distortion (EHID) measurement because linear sound propagation can be compensated by inverse filtering the fundamental component.

Step 11: Open the result window "***Harmonic Distortion (relative)***" in the Operation ***3b X EHID 6 V*** which shows curves *2^{nd} EHID X* and *3^{rd} EIHID X* of the laser measurement. Compare these curves with the curves *2^{nd} EHID SPL* and *3^{rd} EHID SPL* that have been copied from corresponding result window of the microphone measurement ***3a TRF SPL EHID 6 V***.

QUESTION 17: Are the relative Equivalent Harmonic Input Distortions (EHID's) derived from the microphone and the laser measurement similar at all frequencies?

- □ **MC a:** No, at low frequencies (f < 100 Hz), the laser and microphone measurement give almost the same EHID. However, at high frequencies (f > 150 Hz), the EHID calculated from the laser is corrupted by measurement noise resulting in insufficient SNR.
- □ **MC b:** Yes, the laser and the microphone measurement give almost the same EHID at any frequency.

6.2 Intermodulation Distortion Measurement

6.2.1 Intermodulation Distortion

Step 12: Open the operation ***4a DIS SPL IMD (bass sweep)*** and view the frequency response of the intermodulation distortion in the result windows "***2nd Intermod, %***" and "***3rd Intermod, %***".

QUESTION 18: At which frequency of the bass tone are the relative intermodulation distortions (in percent) maximal?

- □ **MC a:** At 35 Hz.
- □ **MC b:** At 95 Hz (close to the resonance frequency).
- □ **MC c:** At 235 Hz.

Step 13: In the same operation ***4a DIS SPL IMD (bass sweep)*** view the result window "***Fundamental Component***" to compare the *6.00V* curve with the ***Fundamental X 6V*** and the ***Fundamental I 6V*** curves, imported from the operations **4b DIS X (bass sweep)** and **4c DIS current (bass sweep)**, respectively.

QUESTION 19: Which state variable has the largest magnitude when the generation of intermodulation distortion increases in the frequency range 20 Hz < f < 120 Hz?

- □ **MC a:** Displacement
- □ **MC b:** Current
- □ **MC c:** Sound Pressure

6.2.2 AM / FM Distortion

Step 14: Open the operation ***4d DIS SPL IMD (voice sweep)*** and inspect the result window "***Waveform YI***" showing the sound pressure signal for a two-tone signal comprised of a first voice tone f_1 = 1.9 kHz and a second bass tone f_2 = 23 Hz.

QUESTION 20: What does the sound pressure time signal reveal?

- □ **MC a:** The envelope of the voice tone at 1.9 kHz varies over time. The distance between the maxima is about 43 ms which corresponds with the period of the bass tone at 23 Hz.
- □ **MC b:** The phase of the voice tone varies over time.

Step 15: Open result window "***Modulation***" in the same operation ***4d DIS SPL IMD (voice sweep)*** and compare the curve *AM distortion(Lamd)*, representing pure Amplitude Modulation (AM) distortion, with the total intermodulation distortion *Ldm (cumul)*, which considers both Amplitude and Frequency Modulation (AM + FM).

QUESTION 21: What causes the variations in the voice tone envelope at 1.9 kHz?

- □ **MC a:** Amplitude Modulation (AM) because the value of *AM distortion(Lamd)* is close to the value of the total intermodulation distortion *Ldm (cumul)*.
- □ **MC b:** Frequency Modulation (FM) because the value of *AM distortion(Lamd)* is much less than the value of the total intermodulation distortion *Ldm (cumul)*.

6.2.3 Distortion in the Input Current

Step 16: Open the operation ***5a TRF CURRENT Harm 6 V*** and in the result window "***Fundamental + Harmonics***" view the absolute *2nd Harmonic* and *3rd Harmonic Distortion* in the input current versus frequency.

QUESTION 22: How does the 2nd-order harmonic distortion in the input current change over the frequency range from 200 Hz to 1.75 kHz?

- □ **MC a:** The absolute value of the harmonic distortion in the input current is constant.
- □ **MC b:** The absolute value of the harmonic distortion in the input current decreases approximately 5 dB per octave.
- □ **MC c:** The absolute value of the harmonic distortion in the input current decreases approximately 12 dB per octave.

Step 17: In the same operation ***5a TRF CURRENT Harm 6 V*** and the same result window "***Fundamental + Harmonics***" view the frequency response of the fundamental component of the current in the frequency range from 200 Hz to 1.75 kHz. Copy the corresponding curves of the sound pressure in ***1b TRF SPL Fund-Harm 6 V*** and displacement in ***2a TRF X Fund + Harm 6 V*** and paste them into this window and compare the curves.

QUESTION 23: Which state variable of the loudspeaker decreases in this frequency range? Hint: it may activate a loudspeaker nonlinearity that generates distortion in the input current between 200 Hz and 1.75 kHz?

- □ **MC a:** Voice coil displacement is decreasing by approximately 12 dB/octave.
- □ **MC b:** Current is slightly decreasing due to the effect of the voice coil inductance.

Step 18: Open the result window "***3rd Intermod, %***" located in the operation ***5b DIS CURRENT IMD (bass sweep)*** and compare the *6V* curve, representing the 3rd-order intermodulation distortion in the input current, with the *IMD SPL 6V* curve, corresponding with the 3rd-order intermodulation found in the sound pressure copied from the operation ***4a DIS SPL IMD (bass sweep)***.

QUESTION 24: Does the 3rd order intermodulation distortion in current and sound pressure have a common cause (the same loudspeaker nonlinearity)?

- □ **MC a:** No, because the 3rd-order IMD in the sound pressure is much larger in amplitude than the 3rd-order IMD in the current signal. Thus, the source of the distortion is in the mechanical or acoustical domains.
- □ **MC b:** Yes, because the values are similar. Thus, the source of the distortion is in the electrical domain.

Step 19: Open the operation ***5c DIS CURRENT IMD (voice sweep)*** and view the result windows "***2nd Intermod %***" and "***3rd Intermod %***". Compare the *3V* intermodulation curve, found in the electrical input current, with the *IMD SPL 3V* curve, found in the sound pressure output copied from the corresponding result windows in operation ***4d DIS SPL IMD (voice sweep)***. Note: both 2nd-order and 3rd-order intermodulation distortions have a local maximum (exceeding 30 %) at 1.9 kHz.

QUESTION 25: Where is the nonlinearity which causes the high intermodulation at 1.9 kHz located?

- □ **MC a:** The nonlinearity is located in the electrical domain because the intermodulation distortions found in the input current are similar to the IMD found in the sound pressure output.
- □ **MC b:** The nonlinearity is located in the mechanical or acoustical domain because the IMD found in the electrical input current is much less than the IMD found in the sound pressure.

6.3 Multi-tone Distortion Measurement

Step 20: Open the operation ***6a LPM MTD 1/10th oct*** and inspect the result window "***P(f) Spectrum***". Compare the two curves *Noise Floor* and *Noise+Distortion*.

QUESTION 26: Is the distortion-to-noise ratio sufficient to separate the nonlinear distortion components from steady-state ambient noise?

- □ **MC a:** No, the noise floor curve is similar to the *Noise+Distortion* curve.
- □ **MC b:** Yes, the noise floor curve is 30 dB below the *Noise+Distortion* curve.

Step 21: In the same operation ***6a LPM MTD 1/10th oct*** and the same result window "***P(f) Spectrum***" compare the *Noise+Distortion* spectrum with the corresponding spectrum in the result window "***Current (f) Spectrum***".

QUESTION 27: Use the cross cursor to read the approximate level difference between the fundamental component and the distortion peak value at 2 kHz. Where is the dominant cause of the distortion located?

- □ **MC a:** It is located in the electrical domain because the input current shows dominant distortion (the same level difference exists between distortion and fundamental in both the input current and the sound pressure output).
- □ **MC b:** It is located in the mechanical or acoustical domain because the input current shows less distortion (a level difference of 30 dB exists between distortion and fundamental in the input current, which is much higher than the corresponding difference of 15 dB in the sound pressure output).

Step 22: Select the operation ***6b LPM MTD 1/10th oct hp*** which shows the multi-tone distortion measurement where the frequency components below 300 Hz are attenuated using a high-pass filtered stimulus. In the result window "***P(f) Spectrum***", compare the *signal line (high-pass*) curve with the *signal line (full band)* curve that was copied from the corresponding result window in the operation ***6a LPM MTD 1/10th oct***. Open the result window "***Multi-tone distortion***" and compare the *MTD high pass* curve with the *MTD full band* curve. Note: the distortions are significantly reduced (20 dB) when the high-pass filtered stimulus is used.

QUESTION 28: Which state signal, that has a significant influence on the generation of MTD above 300 Hz, is significantly reduced by applying a high-pass filter?

- □ **MC a:** voice coil displacement
- □ **MC b:** electrical input current
- □ **MC c:** terminal voltage
- □ **MC d:** sound pressure output at microphone

6.4 Rub & Buzz Measurement

Step 23: Select the Operation ***7c TRF peak harmonics 8V*** and open the result window "***Fundamental+Harmonics***". Activate the cross cursor and view the difference in the magnitude between the *THD* curve, representing the absolute total harmonic distortion at 70 Hz, and the magnitude of the *21st harmonic* component at 70 Hz.

Comment: If the 21st-order harmonic is not displayed press "*c*" or right click in the graph area and select ***customize***. Select property page ***Subset***, press Ctrl on the keyboard and select curve ***21th***.

QUESTION 29: Is the measurement of the 21st-order harmonic component a reliable way for detecting rub & buzz and other irregular defects?

- □ **MC a:** Yes, because the regular nonlinearities inherent in the motor and suspension do not generate significant contributions to higher-order harmonics.
- □ **MC b:** No, a higher-order harmonic component (n > 10) has low spectral energy and is located close to the noise floor. Only significant defects can be detected by reading a higher order single harmonic. Sensitive measurement techniques in the time domain and spectral information between the harmonic components in the frequency domain are required to determine the amplitude and phase information of multiple harmonics (10 < n < 300).

Step 24: Open the result window "***Instantaneous Distortion***" in the same operation ***7c TRF peak harmonics 8V*** and view the peak value of the high-pass filtered PHD with n > 10. This window shows the *Distortion* curve and the *Distortion 1V* curve that was copied from operation ***7a TRF peak harmonics 1V*** using a lower input terminal voltage.

Step 25: Open the result window "***Fundamental+Harmonics***" in the same operation ***7c TRF peak harmonics 8V*** and view the *Absolute PHD* curve**.** Search for the frequency range where the *Absolute PHD* exceeds the permissible *PHD limit* curve which is 40 dB lower than the mean value of the fundamental component.

Step 26: Select the Operation ***7d TRF Crest harmonics 8V*** and inspect the result window "***Instantaneous Distortion***". Search for the frequency range where the instantaneous crest factor ICHD exceeds the permissible limit *Thresh* curve, which has been set to 12 dB.

QUESTION 30: Do the peak value PHD and crest factor ICHD exceed the corresponding limits at the same frequency?

- □ **MC a:** Yes, a loudspeaker irregular defect that has both a high PHD and a high ICHD value in the frequency range from 40 to 60 Hz indicates the generation of impulsive distortion.
- □ **MC b:** No, the instantaneous crest factor ICHD exceeds the limit value in the frequency range 40 to 60 Hz but the PHD is more than 40 dB below the fundamental. Although the impulsive properties of the distortion indicate an irregular defect, the small peak value shows that the defect is negligible.
- □ **MC c:** No, although the peak value PHD exceeds the permissible limit, the low instantaneous crest factor ICHD indicates that the distortions are not impulsive. The peak value may be caused by measurement noise and is not critical.

Step 27: View the result window "***Instantaneous Distortion 3D***" in the same operation ***7d TRF Crest harmonics 8V*** and search for the conditions (frequency, displacement) where the instantaneous crest factor ICHD exceeds the 12 dB threshold corresponding with "black spots" in the diagram.

QUESTION 31: Determine the voice coil displacement where the impulsive distortion occurs and the crest factor exceeds the 12 dB threshold.

- □ **MC a:** In the frequency range 35 < f < 65 Hz the impulsive distortion occurs at the positive peak of the displacement (at 3 mm peak).
- □ **MC b:** In the frequency range 35 < f < 65 Hz the impulsive distortion occurs at the negative peak of the displacement (at 3 mm peak).
- □ **MC c:** In the frequency range 35 < f < 65 Hz the impulsive distortion occurs at the negative peak of the displacement (at 1 mm peak).

Step 28: View the result window "***Modelled and Measured Response***" in the same operation ***7d TRF Crest harmonics 8V*** which shows the distortion waveform plotted as the *Residual* curve versus instantaneous frequency. Zoom into the *Residual* curve at 60 Hz to see the impulsive distortion generated at the maximum negative sound pressure signal which corresponds to a positive displacement maximum (sound pressure is proportional to acceleration).

7 Performing Measurements (Hardware required)

7.1 Setup the Hardware

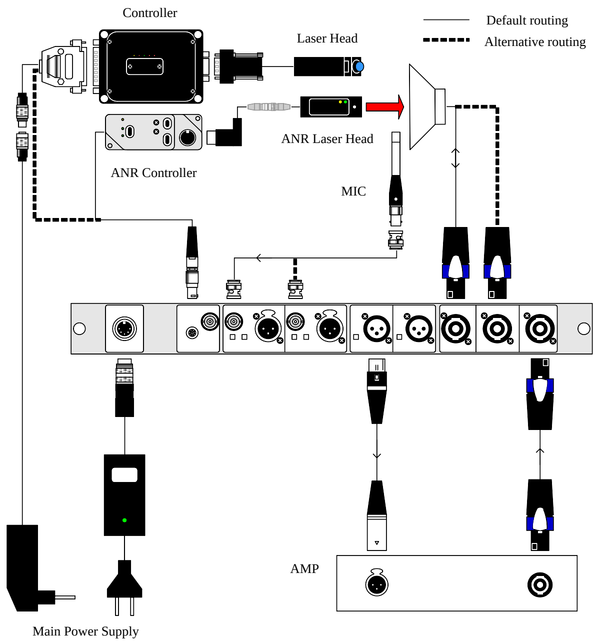

Figure 13 Pin Assignments DA2

Step 29: Connect XLR Output ***OUT 1*** located on the rear side of the *Distortion Analyzer (DA)* to the XLR Input of the Amplifier.

Step 30: Connect the Output of the Amplifier to the ***AMPLIFIER*** SPEAKON input connector located on the rear side of the *DA*.

Step 31: Connect the *DA* SPEAKON Output ***Speaker 1*** to the loudspeaker terminals. Use the special speaker measurement cable supplied.

Step 32: If you have a Laser, connect the laser head with the controller and link the LEMO plug of the controller with the input marked ***LASER*** located on the rear side of the *DA*.

Step 33: Connect the ***USB*** input located on the front of the *DA* to the PC.

Step 34: Select a transducer for testing. A woofer having a resonance frequency below 100 Hz is recommended. Solidly clamp the transducer into the speaker stand ensuring that the cone is positioned orthogonal to the laser beam.

Step 35: Apply a white dot to the middle of the cone (use TippEx® or a small white sticker).

Step 36: Aim the laser beam at the white dot.

Step 37: Bring the laser head to its working range.

Step 38: Place the microphone in the near field of the loudspeaker and connect it to ***INPUT 1*** of the *DA*.

7.2 Multi-tone Distortion

Step 39: Press the button *NEW OBJECT* on the tool bar and select the KLIPPEL template *Diagnostics WOOFER*. Enter a useful name for the device under test.

Step 40: Open the property page Stimulus in the operation *4g LPM Multitone distortion*. Adjust the input voltage to the transducer (start with a small value around 1 V to avoid damaging the speaker). Start the measurement.

Step 41: Open the result window *P(f) Spectrum* of the microphone signal. Inspect the multi-tone distortion and the noise floor.

Step 42: Open the result window *Multitone Distortion* and read the maximum of the distortion. If the distortion maximum is below -20 dB (less than 10 %), increase the voltage in the property page *Stimulus* and repeat the measurement.

Step 43: For future measurements, make note of the final voltage determined in step 42.

Step 44: Open the result window *X(t)* and read the peak value of the displacement.

Step 45: Open the result window *Current (f) spectrum*. Determine the relative level of the distortion in the current by finding the level difference between the multi-tone distortion maximum and the fundamental in the current signal. Compare this result with the relative level of the distortion in the sound pressure output. Is the dominant nonlinearity in the electrical domain?

7.3 Voice Coil Displacement

Step 46: Open the property page Stimulus for the operation *3a DIS X Fundamental* DC and enter the voltage used in the *4g LPM Multitone distortion* for U end (use 0.1 V for U start). Start the measurement.

Step 47: Determine the peak and bottom value of the displacement in the result window *PEAK + BOTTOM*.

Step 48: In result window *COMPRESSION*, inspect the amplitude compression at frequencies below resonance.

Step 49: Inspect the DC-displacement generated by the woofer. Find the frequency where the transducer generates the largest DC-displacement and compare it with the transducer resonance frequency f_s, as shown in the result window *Table Linear Parameters* for the operation *4g LPM Multitone distortion.*

Step 50: Open the property page Stimulus for the operation *3b DIS motor stability* and enter the critical frequency $f = 1.5\, f_s$. Ensure that the voice coil temperature monitoring is enabled in property page *Protection* and set the maximal allowed temperature increase to 60 K. Start the measurement. (The protection based on temperature monitoring makes it possible to use a maximal voltage U end which is higher than the voltage used in operation *4g LPM Multitone distortion*.)

Step 51: Read the maximal value in the result window *DC component* and compare it with the AC component shown in result window *Fundamental Component*. Is the motor stable?

7.4 Harmonic Distortion

Step 52: Open the property page Stimulus for the operation *4a TRF SPL + harmonics* and adjust the stimulus voltage to the value used in *4g LPM Multitone distortion*. Ensure that the check box “noise floor monitoring” is enabled. Start the measurement.

Step 53: Open the result window *Y1(f) Spectrum* and check the signal to noise ratio. Do you have an SNR better than 30 dB in the frequency range of interest?

Step 54: Open the result window *Impulse response* and set the left and right window cursors around the direct sound so that room reflections arriving at a later time will be suppressed.

Step 55: Inspect the harmonics in the result window *Fundamental + Harmonics*. Determine which order of harmonic distortion dominates the total harmonic distortion (THD).

Step 56: Open the result window *Harmonic Distortion* and search for the frequency of maximum distortion above f_s. Compare the harmonic distortion at this frequency with the multi-tone distortion at the same frequency in operation *4g LPM Multitone distortion*. Explain the difference.

7.5 Equivalent Input Distortion

Step 57: Copy the operation *4a TRF SPL + harmonics* and paste it under the same object in dB-lab. Rename the operation *4a TRF Equivalent Harmonics*. Open the result window *Fundamental + Harmonics* and copy the *Fundamental* curve onto the clipboard. Open the property page *Processing* and paste this curve into the *IMPORT* button.

Step 58: View the window *Fundamental + Harmonics* and check that the *Fundamental* curve has become almost flat and located close to zero. The harmonic distortion curves now shown in this window represent the equivalent input distortion. Why are the EID's almost constant for frequencies below resonance?

Step 59: Open the result window *Harmonic Distortion* and compare the EID relative distortion with the relative distortion found in the sound pressure output in the corresponding window of operation *4a TRF SPL + harmonics*.

7.6 Intermodulation Distortion

Step 60: Open the property page *Stimulus* of the operation *4e DIS IM Dist. (voice sweep)* and enter the voltage used in the *4g LPM Multitone distortion* for U end (use 0.1 V for U start). Start the measurement. Listen for fluctuation and roughness in the reproduced high frequency tone.

Step 61: View the result windows *2nd Intermod, %* and *3rd Intermod, %* and search for the frequency of maximum intermodulation distortion f_{max}. Compare this value with the harmonic distortion at the same frequency f_{max} in operation *4a TRF SPL + harmonics*.

7.7 Rub&Buzz and other Irregular Distortion

Step 62: Open the property page *Stimulus* of the operation *5 TRF Rub and Buzz* and enter the voltage value 1 V. Rename the measurement *5 TRF Rub and Buzz 1V.* Start the measurement. Listen for bottoming, voice coil rubbing and other symptoms of irregular defects.

Step 63: Open the result window Fundamental + Harmonics and search for the maximum in the *Absolute PHD* curve. Does the curve exceed the *PHD limit (-40 dB)* curve? Repeat the measurement to ensure that this maximum is reproducible and not caused by ambient noise. A loudspeaker without any defect should produce only noise (a flat line which is independent of frequency).

Step 64: Hold an obstacle (pen, screw driver) at a close distance from the cone and start the measurement again. If the cone has struck the obstacle and produced an impulse you will see a distinct increase in the Absolute *PHD*. Open the property page *I-Dist* and select *ICHD* located under *Measure*. Open the result window *Instantaneous Distortion 3D* showing the instantaneous crest factor versus displacement and frequency. Find the positive displacement and frequency required for the cone's surface to hit the obstacle.

Step 65: Duplicate the measurement *5 TRF Rub and Buzz 1V* and enter the voltage used in the *4g LPM Multitone distortion.* Rename the operation *5 TRF Rub and Buzz high voltage*. Start the measurement. Listen for bottoming, voice coil rubbing and other symptoms of irregular defects. Open the result window *Fundamental + Harmonics* and search for the maximum in the *Absolute PHD* curve. Repeat the measurement to ensure that this maximum is reproducible and not caused by ambient noise. Does the curve exceed the *PHD limit (-40 dB)* curve? If not, increase the voltage and start the measurement again.

Step 66: Open the result window *Instantaneous Distortion 3D* of the measurement *5 TRF Rub and Buzz high voltage* showing the instantaneous crest factor versus displacement and frequency. Search for black spots in the 3D window and read the position of the voice coil where the impulsive distortion is generated.

Step 67: Discuss the possible cause which could lead to the measured impulsive distortion.

8 Further Literature

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

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

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

Paper “*Measurement of Equivalent Input Distortion”*:

http://www.klippel.de/uploads/media/Measurement_of_Equivalent_Input_Distortion_03.pdf

Paper “*Measurement of Impulsive Distortion, Rub and Buzz and other Disturbances”:*

http://www.klippel.de/uploads/media/Measurement_of_Rub_and_Buzz_03.pdf