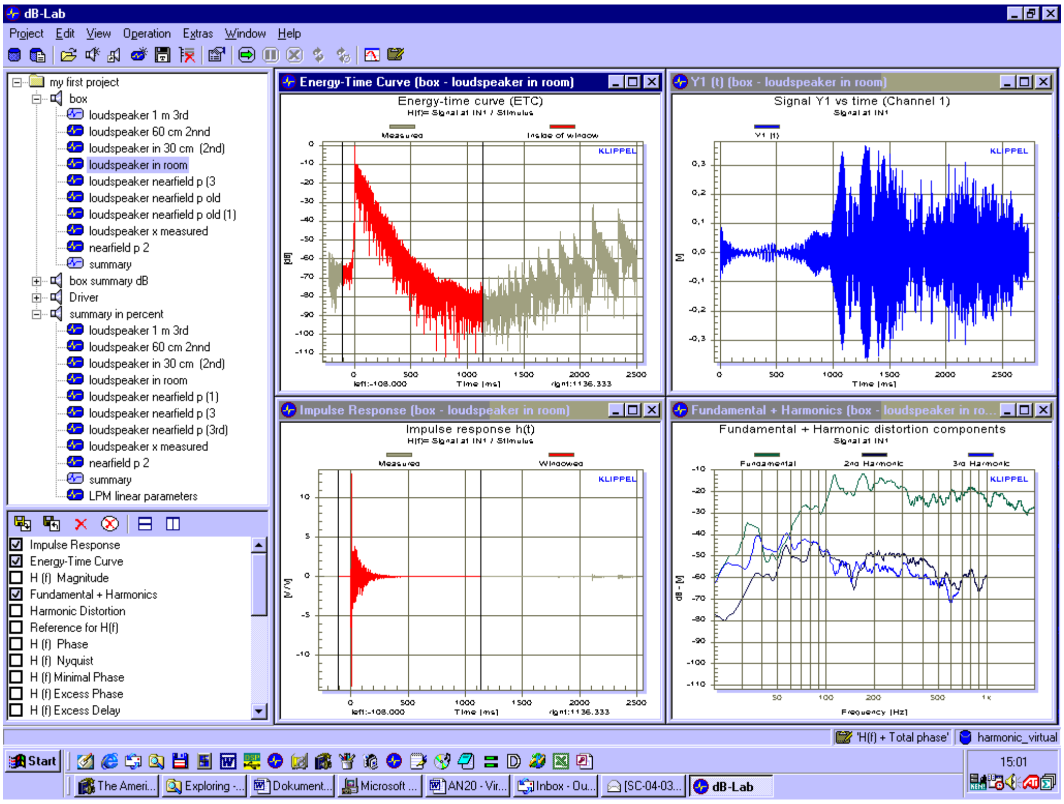
Measurement of Equivalent Input Distortion **AN 20**

Application Note to the KLIPPEL R&D SYSTEM

Traditional measurements of harmonic distortion performed on loudspeakers reveal not only the symptoms of the nonlinearities but also the effect of linear loudspeaker parameters, radiation into the sound field and the interactions with the room. Thus, the interpretation and comparison of results are difficult if the acoustical conditions change. This problem can be solved by transforming the harmonic distortion measured in the sound pressure into equivalent distortion at the voltage input. The equivalent distortion is almost independent of the radiation, sound propagation, room acoustics and the linear properties of the sensor (Laser, microphone). The equivalent harmonic distortion are not only a minimal set of information but make it possible to predict the traditional harmonic distortion according (IEC standard) at any point r in the sound field by performing a simple filtering with a linear transfer function.



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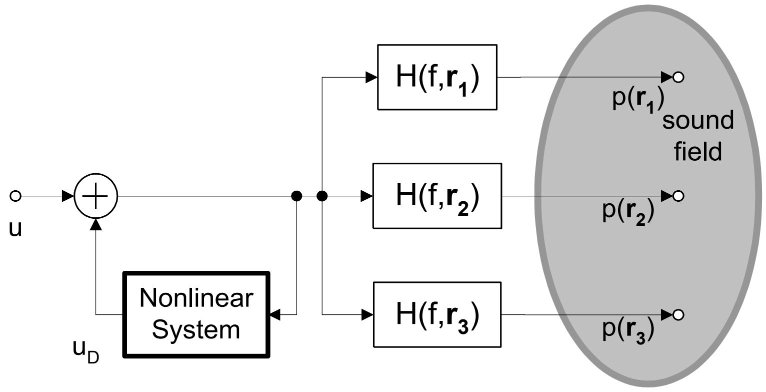
Modeling

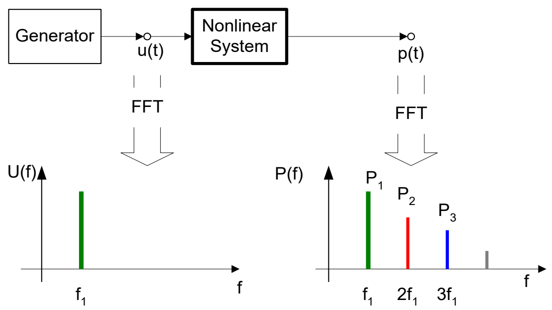
Loudspeaker Model



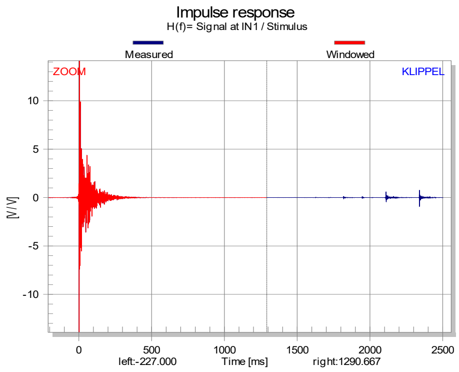
The loudspeaker may be considered as a system having a single input at the terminals and multiple outputs in the sound field. The small amplitudes the signal flow between input U_1 and output P(**r**) at point **r** may be represented by a linear transfer function H(f,**r**). In the large signal domain nonlinearities inherent in the loudspeaker produce distortion. Most of the nonlinearities are located in the electrical and mechanical domain where the signal flow is still one-dimensional. In this case the nonlinearities can be lumped into one system generating distortion U_D which are added to the input signal U and then dispersed via the linear systems H(f, **r**) to any point in the far field.

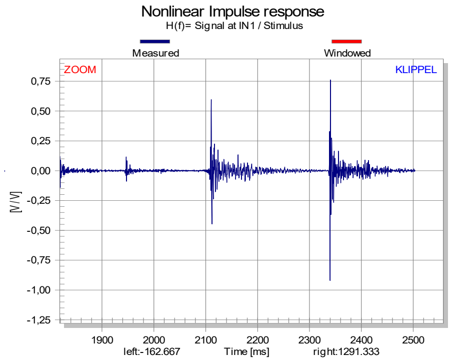
Measurement at one point in the sound field

Harmonic distortion are usually measured only at few points in the sound field to make practical performance feasible. After performing a spectral analysis the reponses of the fundamental component P_1(f) and the 2nd-harmonic component P_2(2f) and higher-order harmonics are measured versus excitation frequency f.



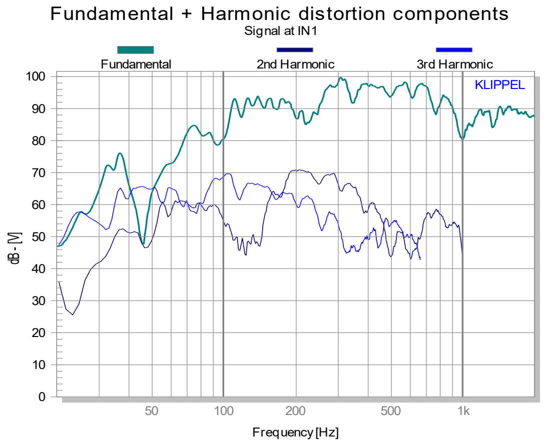
Impulse response





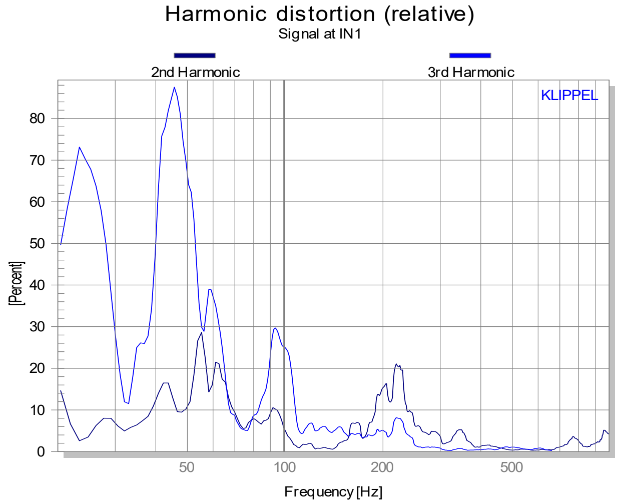
The Transfer Function Module (TRF) uses a sinusoidal sweep as excitation signal to separate the linear impulse response (0-0.2 s) from the nonlinear responses of 2nd, 3rd and higher (appear as peaks between 2 –2.5 seconds). Due to the diffuse field in the listening room both the linear and the nonlinear responses have a relatively long decay time. Thus, higher-order responses interfere with each other and can not completely be separated by using time windowing.

Amplitude Response



Applying an FFT to the windowed linear and nonlinear impulse responses leads to the magnitude of the fundamental $P_1(f)$, 2nd-order component $|P_2(2f)|$ and the 3rd -order component $|P_3(3f)|$. At 1 m distance from the loudspeaker the diffuse field in the room generates significant variation of the sound pressure values.

Relative Harmonic Distortion in the output signal (IEC standard)



According to the IEC 60268-5 the nth-order harmonic distortion components generated by an excitation frequency f are referred to the rms amplitude of the total output signal P(f) and may be expressed in percent

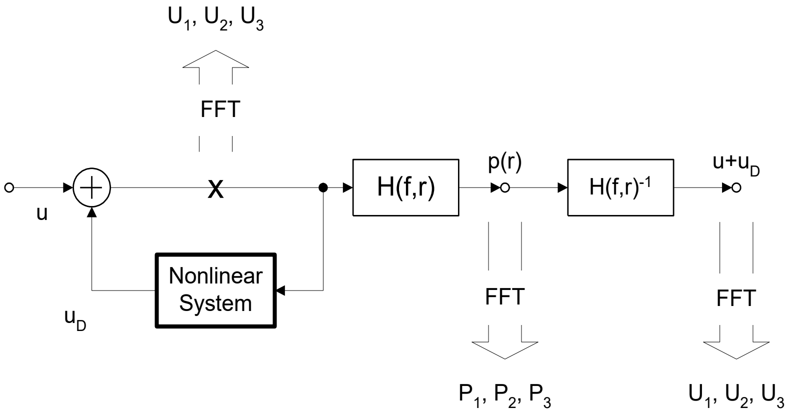
$$d_n^{(p)} = \frac{|P_n|}{P_t} 100\% \quad ,\text{n > 1 with } P_t = \sqrt{\frac{1}{T}\int_0^T p(t)^2 \, dt} \qquad (1)$$

or in dB

$$L_n^{(p)} = 20\log\left(\frac{d_n^{(p)}}{100\%}\right) \text{, n > 1.} \tag{2}$$

This measure reflects the interactions between loudspeaker and room at the point $\mathbf{r}_1$ in the sound field. The sparse density of room modes generates fluctuations which vary with the measurement position $\mathbf{r}_1$.

Equivalent Input Distortion



By performing a filtering of the input measured sound pressure signal with the inverse system function $H(f)^{-1}$, the effect of the radiation and room interactions can be compensated and the equivalent harmonic distortion

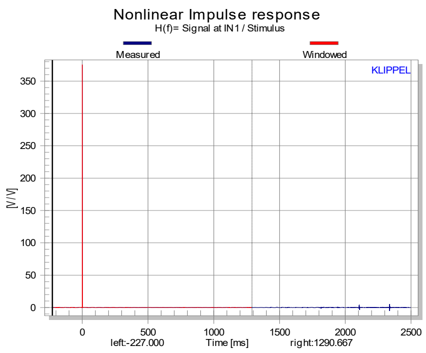
$$U_n = \frac{P_n}{H(nf)} = \frac{P(nf)}{H(nf)} \text{, n > 1} \tag{3}$$

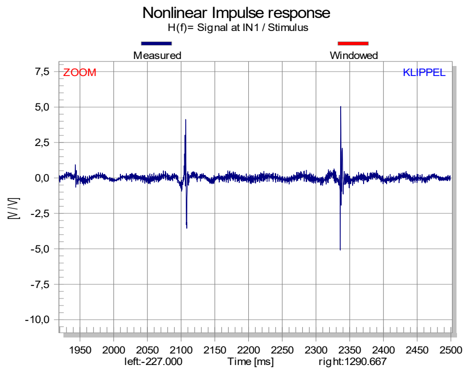
can be calculated.

The transfer function can be calculated by

$$H(f,\mathbf{r}_1) = \frac{P_1(f,\mathbf{r}_1)}{U_1(f)} \tag{4}$$

using the fundamental $P_1(f)$ and the input voltage $U_1(f)$.

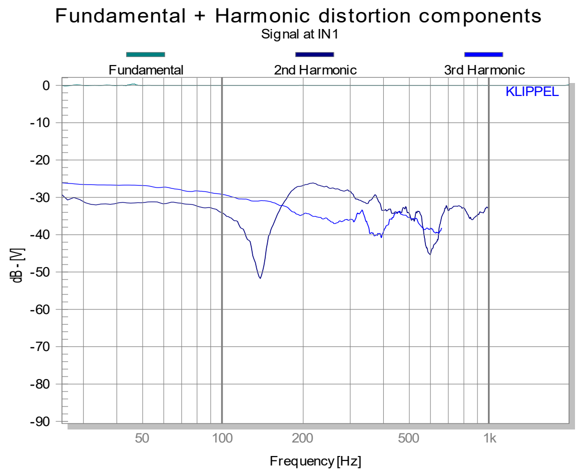




Thus, the linear impulse response becomes close to an ideal Dirac impulse and the nonlinear impulse responses become much shorter because the effect of the room response is removed.

Absolute Equivalent Distortion Components

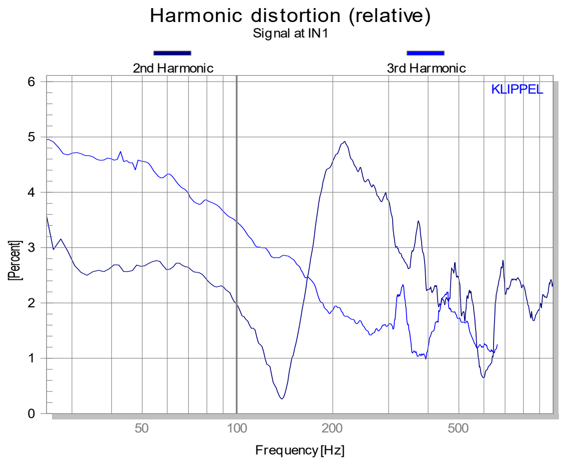
(voltage)



The "inverse filtering" of the measured fundamental and harmonic responses with the linear transfer function $H(f,r_1)$ generates to the almost constant linear transfer function $U_1(f)$ and 2nd-order component $U_2(2f)$ and the 3rd-order component $U_3(3f)$. These curves describe the nonlinear distortion at the source and are almost independent on the radiation, sound propagation, room interactions and the properties of the sensor.

Relative Distortion

(Percent)



In an analogue way as in the IEC standard the amplitude of the nth-order harmonic distortion is referred to the rms value of the total voltage signal.

$$d_n^{(u)}(f) = \frac{|U_n(nf)|}{\sqrt{|U_1(f)|^2 + |U_2(2f)|^2 + |U_3(3f)|^2 + ...}} 100\% \qquad (5)$$

Assessing loudspeaker system

The large signal performance of loudspeakers can be more easily assessed by comparing the equivalent harmonic distortion in the input voltage because the influence of the acoustical environment (room), distance and the sensor is minimal. It is also possible to predict the harmonic distortion at a preferred listening position (drivers seat) based on equivalent harmonic distortion (representing the loudspeaker) and the linear transfer function H(f) describing the room acoustics.

Performing the Measurement

Requirements

The following hardware and software is required

- Distortion Analyzer
- PC
- Software modules (TRF, dB-Lab)
- Sensor microphone (or laser)
- Power Amplifier (set gain to maximum)

Setup

Connect the terminals of the driver with SPEAKER 1. Switch the power amplifier between OUT1 and connector AMPLIFIER. Connect the microphone to input IN1,or connect a laser head to the connector LASER and adjust the laser beam to a white dot on the diaphragm.

Preparation

1. Create a new database
2. Open the database within dB-Lab
3. Create a new object DRIVER based on the template **Equivalent Input Dist. AN 20**.

Measurement	1. Adjust sensor. When using a microphone prefer a measurement point giving sufficient SNR (nearfield measurement or distance < 1m is preferable !!). 2. Start the measurement "1. **TRF Small Signal Measurement**". 3. Open Property Page IM/EXPORT in the measurement "**1. TRF Small Signal Measurement**" and select the transfer function "Fundamental + Total Phase". Press the button "Export to Clipboard". Select the measurement "**2. TRF Equivalent Distortion**". Open the Property Page PROCESSING and press button IMPORT under “Reference”. Press the button "From Clipboard" to transfer data. 4. Open the Property Page STIMULUS of the measurement "**2. TRF Equivalent Distortion**". Adjust the voltage U of the stimulus in dBU according to the permissible load. Start the measurement. 5. Open the Result Window "Energy-Time Curve" and adjust the marker of the time window to separate the linear response from the nonlinear impulse responses (small spikes at later times).

Setup Parameters for the TRF Module

Template

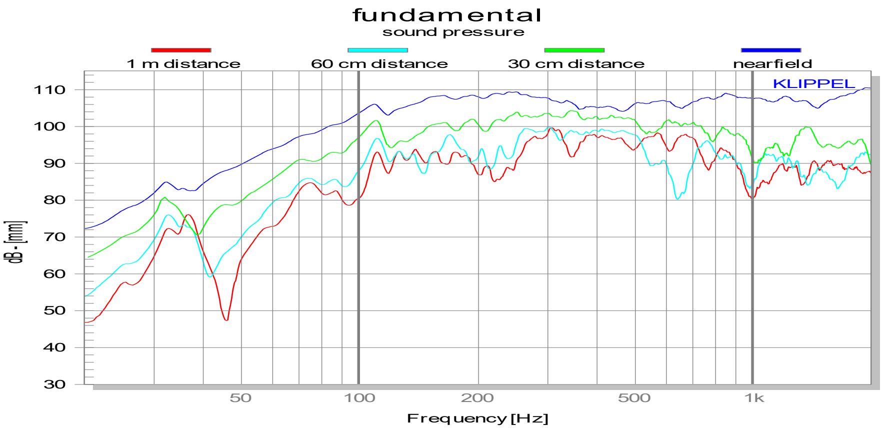
Create a new Object, using the object template **Equivalent Input Dist. AN 20** in dB-Lab. If this database is not available you may generate an object **TRF Equivalent Distortion AN 20** based on the general TRF module.

1. Generate an object **Equivalent Input Dist. AN 20.**
2. Assign to the object a measurement "**1. TRF Small Signal Measurement**" and "**2. TRF Equivalent Distortion**" based on the default TRF measurement.
3. Select "**1. TRF Small Signal Measurement**". Open the PP STIMULUS and set starting frequency Fmin= 20 Hz and Fmax =20kHz. Set resolution to 0.73 Hz. Set Voltage to 0 dBU at SPEAKER 1 Terminals. Select 16 times averaging. Open PP INPUT, select IN1 at Channel 1 and disable Channel 2. Open PP Processing and select “No Window”.
4. Select "**2. TRF Equivalent Distortion** ". Open the PP STIMULUS and set starting frequency Fmin= 20 Hz and Fmax =20kHz. Set resolution to 0.73 Hz. Set Voltage to 12 dBU or higher at SPEAKER 1 Terminals. Open PP INPUT, select IN1 at Channel 1 and disable Channel 2. Open PP Processing and use default setting of “Rectangular Window”.

You may also modify the setup parameters according to your needs.

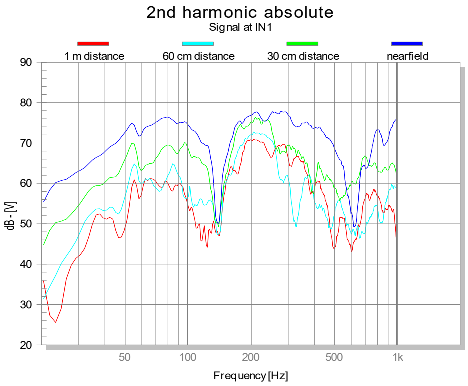
Example

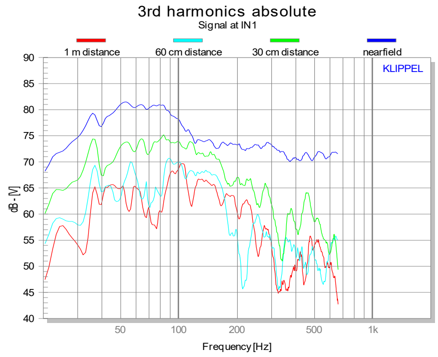
Linear Amplitude Response



The sound pressure response of a loudspeaker was measured at four positions (nearfield, 30 cm, 60 cm and 1 m distance). With rising distance the diffuse field in the room causes significant variations in the response (> 20 dB variation at 50 Hz) .

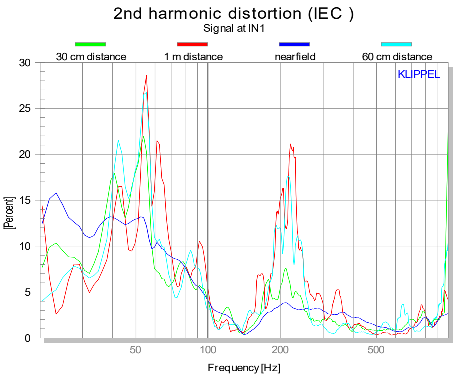
Harmonic Distortion Components

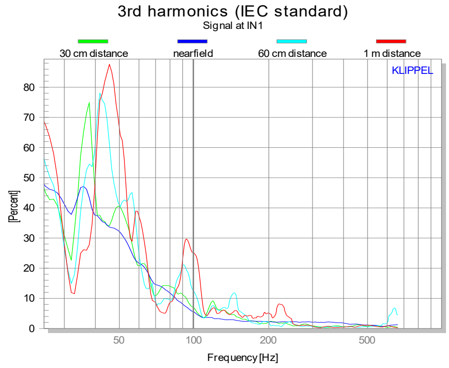




The absolute sound pressure response of the 2nd and 3rd-order harmonics vary significantly within the sound field.

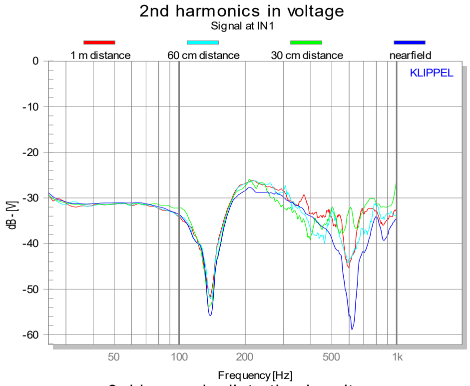
Relative Harmonic distortion in sound pressure

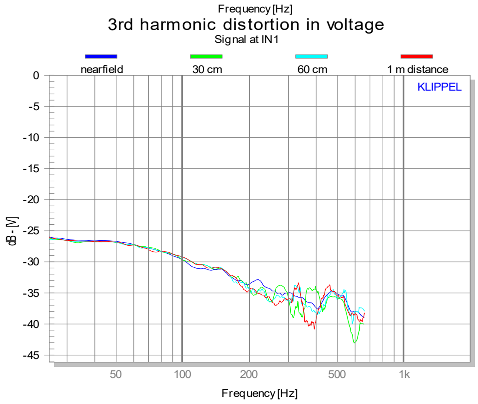




Calculating the 2nd and 3rd-order harmonic distortion in percent according to the standard IEC 60268 the variation of the fundamental are considered in the assessment of the distortion. However, there are substantial variations due to the interferences with the room modes.

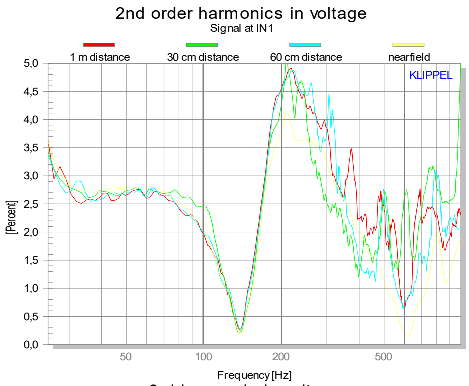
Equivalent harmonic distortion in voltage

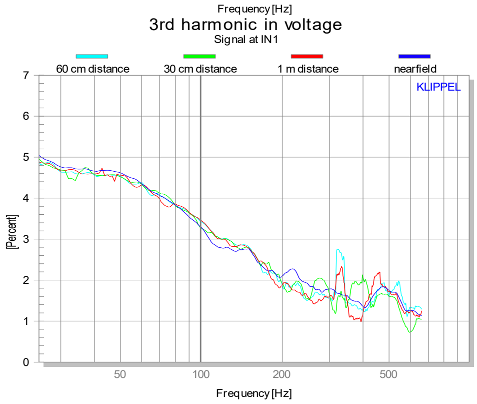




The 2nd and 3rd-order harmonics measured at the point $\mathbf{r}_i$ are filtered with the inverse linear transfer function $H(f,\mathbf{r}_i)^{-1}$ to generate the equivalent harmonic distortion in the sound pressure input. If the nonlinearities are located in the electrical, mechanical and acoustical part before the sound disperses the equivalent harmonic distortion are independent of radiation, sound propagation and room interferences.

Equivalent harmonic distortion in voltage input (in percent)





The equivalent harmonic distortion may be referred to the total voltage signal and expressed in percent.

Glossary of Symbols

$H(f,\mathbf{r})$ linear transfer function between voltage and sound pressure at point **r**

u voltage at terminals

$p(\mathbf{r})$ sound pressure at point **r**

$P_1(f)$ fundamental component in sound pressure versus excitation frequency f

$P_2(2f)$ 2nd-order harmonic component in sound pressure versus excitation frequency f

$P_3(3f)$ 3rd-order harmonic component in sound pressure versus excitation frequency f

$U_1(f)$ *voltage at terminals* versus excitation frequency f

$U_2(2f)$ equivalent 2nd-order harmonic component in voltage versus excitation frequency f

$U_3(3f)$ equivalent 3rd-order harmonic component in voltage versus excitation frequency f

$G_2(f_1,f_2)$ 2nd-order system function generating 2nd-order harmonics

$G_3(f_1,f_2,f_3)$ 3rd -order system function generating 3rd-order harmonics

$d_n^{(p)}(f)$ nth-order harmonic distortion in the sound pressure output signal in percent (according to IEC 60268)

$d_n^{(u)}(f)$ equivalent nth-order harmonic distortion in the input voltage in percent

More Information

Literature	W. Klippel, "Measurement of Equivalent Input Distortion Presented at the 115th Convention of the Audio Eng. Society, 2003 October 10–13, New York, USA, Preprint 5913.
Related Specification	"TRF", S7
Software	User Manual for the KLIPPEL R&D SYSTEM.

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