Measurement of Impulsive Distortion, Rub and Buzz and other Disturbances

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ABSTRACT

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

1   Introduction

Quality Control becomes more and more an important issue in loudspeaker manufacturing and requires a reliable technique for selecting defect speakers in a short time even under noisy conditions. A variety of objective techniques [1 – 4] have been developed over the years. However, the human ear is still the preferred tool at many factory sites. A trained tester is able to detect malfunctions producing even small symptoms which are almost inaudible for the untrained ear. However, fatigue and distraction reduce the attention and cause wrong pass/fail decision.

One of the reason for the superiority of the human tester is his learning capability. Any loudspeaker produces some kind of distortion which are considered as typical for the "good" driver and the tester has to distinguish between regular and excessive distortion.

The target of this paper is to discuss this process from an objective point of view and to suggest a new method which allows to select defect drivers more reliably. Whereas quality control is mainly interested in correct pass/fail decisions the loudspeaker design also needs a better tool to verify subjective assessments and to find the physical causes of malfunction.

2   Glossary of Symbols

f	frequency
t	time
$x(t)$	input signal
$y(t)$	measured output signal
$y'(t)$	modeled output signal
$e(t)$	residual distortion signal
f_s	resonance frequency of the driver
ID	instantaneous distortion
IHD	instantaneous harmonic distortion
MD	mean value of distortion
MHD	mean value of harmonic distortion
PD	peak value of distortion
PHD	peak value of harmonic distortion
ICD	instantaneous crest factor of distortion
ICHD	instantaneous crest factor of harmonic distortion
CD	crest factor of distortion
CHD	crest factor of harmonic distortion
THDN	total harmonic distortion plus noise

3   Causes of Signal Distortion

If we compare the input and output signal of the system we find significant deviations in the waveform:

    1.   Constant amplitude and time delay distortion

Varying the distance between loudspeaker and listener will cause significant change in the amplitude and in arrival time of the signal. Clearly this is a trivial variation of the waveform and has not much impact on subjective evaluation. However, monitoring the sensitivity of the driver is required in manufacturing to ensure that the magnet is fully magnetized.

    2.   Linear amplitude and phase distortion

The vibration of diaphragm, the radiation into 3D-space and the acoustical environment affect the waveform of the reproduced signal at the listening position. At small amplitudes where the loudspeaker behaves linearly the distortion may completely be described by the variation of the complex transfer function versus frequency. This affects the transferred signal and produces "*linear distortion"* in the output. The human ear is much more sensitive to amplitude variations than to phase variations. To assess the linear distortion of a loudspeaker we have to consider the free-field response of the drivers at all possible angles in the 3D space. The directivity pattern of the speaker and room properties produce significant linear distortion in the diffuse field. Fortunately, the listener tolerates a certain amount of linear distortion as normal and might even perceive it as pleasant or interesting. This gives the loudspeaker designer freedom to shape the linear transfer behavior to some extend in order to have the best performance under normal use. Manufacturing is supposed to duplicate the final design model as precisely as possible and all linear distortion are considered as desired properties of a good speaker.

3.   Regular nonlinear speaker distortion

The force factor *Bl(x)*, the compliance $C_{ms}(x)$ of the mechanical suspension and the inductance $L_e(x)$ of the voice coil depend on the instantaneous displacement *x*. As the dominant nonlinearities inherent in electro-dynamical transducers they generate substantial nonlinear distortion. For a multi-tone stimulus we may identify them as harmonic and intermodulation components. In contrast to the linear distortion the regular nonlinear distortion are negligible in the small signal domain but rise rapidly with the signal amplitude. The radiation and acoustical environment have no direct influence on the generation process because the dominant nonlinearities of the speaker are located in the electro-mechanical part. The distortion components once generated are radiated exactly like the linear components. High second-order distortion reveals a distinct asymmetry in the nonlinear characteristic. In most cases it is caused by defects in design or manufacturing and can be fixed at low cost. The third-order distortion corresponds with the symmetrical characteristic and is mainly caused by the trade-off between size, weight and cost of the driver. Thus, some of the driver nonlinearities and the resulting nonlinear distortion are excepted as part of the intended loudspeaker design.

4.   Triggered distortion (rub & buzz)

Improper functioning of the speaker is a infinite source of signal distortion. If the voice coil rubs on the pole tips, the lead wire strikes the diaphragm or a loose glue joint starts vibrating the collective term *rub and buzz* distortion is used. The expression *triggered distortion* would be more appropriate because the distortion depends on the stimulus, is only initiated under certain conditions of the state variables and is reproducible. The dependence on signal amplitude reveals a strong nonlinear mechanism which transforms signal energy to other frequency components. In many cases parts of the speaker (loose parts, leads) start vibrating at their own resonance and produce a buzzing sound. In other cases there are only short clicks which contribute not much energy to the signal. The human ear is quite sensitive for triggered distortion because its signal properties do not fall into the pattern of common audio signals. Thus, Rub and Buzz distortion is not acceptable in loudspeaker design and has to be detected in quality control.

5.   Random distortion

There are other distortion which are not reproducible under identical conditions. For example a loose connection in the soldering of the wire interrupts the current almost sporadically. However, the random disturbances also depend on the input signal

because some energy has to be provided to the speaker. Since the random disturbances might only occur rarely it is most difficult to find them in quality control.

6.   Independent disturbances

The sound produced by the speaker will be disturbed by any ambient sound and noise generated by the measurement equipment (microphone, amplifier, AD-converter). These disturbances are completely independent of the signal transferred. Unlike random disturbances they also occur when no signal is supplied to the speaker. Independent disturbances limit the speed of a measurement system especially if triggered distortion or random distortion have to be detected.

4   Frequency Domain Analysis

The different kinds of distortion have a completely different impact on the subjective sound perception. Some kinds of distortion are irrelevant, other ones are tolerated by the ear or even desired as a nice effect. Others can be easily recognized as a defect of the speaker. Thus we need an objective way to separate distortion. The traditional approach is to transform the time signal into the frequency domain as shown in Fig. 1.
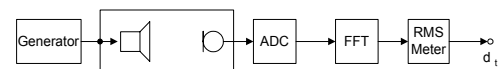

Fig. 1: Measurement of signal distortion by a spectral analysis (FFT)

Comparing the spectrum of the input and output signal shows the distortion of a linear system directly. If the system is nonlinear we find additional spectral components which are not part of the input signal. To distinguish the fundamental from the harmonic and intermodulation distortion we have to use a special stimulus comprising a limited number of spectral components (single tone or sparse multi-tone signal). This technique proves to be very powerful for measuring the linear and regular nonlinear distortion of a driver in short time while ensuring sufficient signal to noise ratio. However, this technique seems to be not optimal for assessing the triggered distortion caused by rub and buzz phenomena. Here the human ear seems still superior in finding speaker defects. The reason for this will be illustrated on two units of the same type where a small difference was just audible. In a almost perfect acoustical environment the radiated sound pressure has been measured in the near field of the driver with a relatively long single-tone stimulus (5 s) at about 50 Hz. The sound pressure signal was analyzed by an FFT-length which was just a multiple of the tone's period to dispense with any windowing.
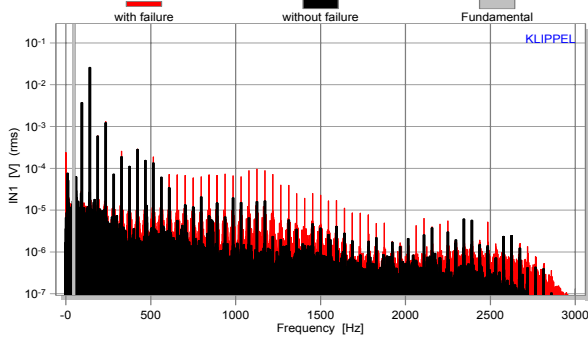
Fig. 2: Spectrum of the reproduced sound pressure signal measured on two units (driver A) with and without failure.

Under these ideal conditions the noise level was less than 100 -120 dB below the fundamental component. Fig. 2 shows the spectrum of the properly working speaker as thick black lines. The regular nonlinearities in the speaker produce about 10 % harmonic distortion with a dominant third-order component. Their amplitude rapidly decreases to 80 dB below the fundamental at the 12th harmonic. However, the higher harmonics up to 60th order are still 20 dB above the noise floor. The spectrum of the defect unit is depicted as red, thin lines in Fig. 2. At low frequencies where the regular nonlinearities of the driver are dominant it coincides almost perfectly with the reference driver. The higher-order harmonics (15 < order < 40) are about 15 dB above the reference. Most of the rub and buzz and any other triggered distortion also fall into the harmonic overtone structure but will be distributed over a large number of harmonics. Thus the energy of each component is extremely low (80 dB below the fundamental). The conditions required to measure such low levels are not feasible in quality control. Performing a complete frequency sweep over a wide frequency range in less than 1-2 seconds produces problems with the signal to noise ratio.

Assessing the mean-power or rms-amplitude of the higher-harmonics exploits only half of the information we get from the triggered distortion. The harmonic analysis provides us for each harmonic not only an amplitude value but also the phase information. The interpretation of the phase information is difficult and not common in distortion measurements. However, the amplitude and phase information is required to reconstruct the time signal of the distortion in its fine structure. There the triggered distortion accumulates to significant peak values at certain times while the mean value is much lower. Thus analyzing the distortion in the time domain is a convenient way to use both amplitude and phase information.

5   Time Domain Analysis

5.1   Separation of Distortion

The distortion we are interested in can be separated by calculating the difference between the measured output $y(t)$ and a desired output $y(t)'$ at any time instance t.

The resulting *residual distortion*

$$e(t) = y(t) - y'(t) \tag{1}$$

is a time signal depending on the properties of the stimulus and the system under test. The desired output $y'(t)$ is generated by a model

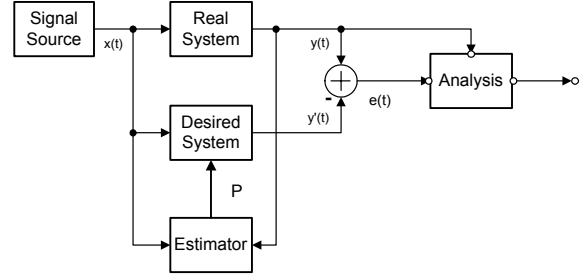of the real system which is excited by the same stimulus $x(t)$ as illustrated in Fig. 3.

Fig. 3: Measurement of the residual distortion signal by comparing measured and predicted signal

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

The outlined method can be illustrated on the following example (example B). A woofer is exited by a continuous sine sweep $x(t)$ at high amplitude. Operated in the large signal domain the driver produces the sound pressure output $y(t)$ shown as blue, bold line in Fig. 4. Variation of the signal amplitude reveals the linear distortion. Also, the motor and suspension nonlinearity cause significant distortion at the beginning of the sweep where the excitation frequency is low. Both the linear and regular nonlinear behavior of loudspeaker is identified and modeled. The corresponding desired system output $y(t)'$ is represented as gray line in Fig. 4.
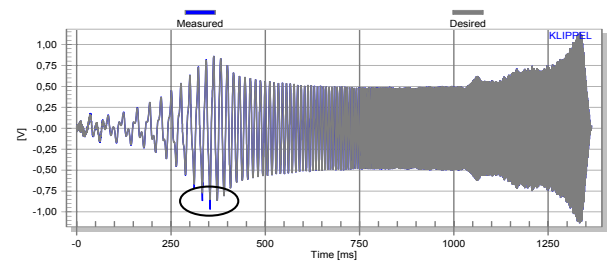

Fig. 4: Measured and desired sound pressure of example driver B.

The measured and desired waveform are almost identical. However, the negative half waves deviate between 300 and 350 ms. A detailed plot of this region is depicted in Fig. 5.
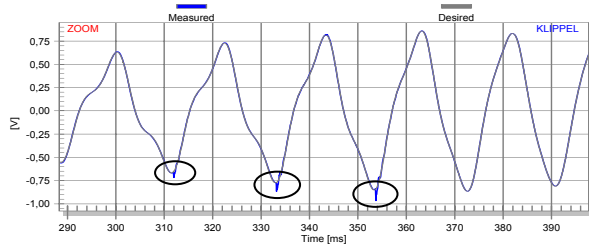
Fig. 5: Detail of the measured and desired sound pressure time signal shown in Fig. 4.

There are three little peaks in the measured sound pressure curve near the turning points of the negative half wave.
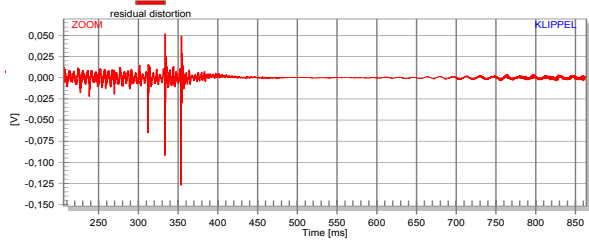

Fig. 6: Detail of the residual distortion signal *e(t)*

Fig. 6 shows the residual distortion *e(t)* versus measurement time t which comprises the triggered distortion between 300 – 350 ms, measurement noise and some error caused by imperfect modeling of the regular distortion.

5.2 Dependency on signal properties

The residual distortion signal *e(t)* is subjected to further analysis as shown in Fig. 3. The dependency of the residual distortion *e(t)* on measurement time is not very instructive. In order to understand the causes of the distortion it is useful to plot the residual distortion versus relevant properties of the signal *y(t)* such as frequency, amplitude, etc.

If a continuous sine sweep is used as stimulus the instantaneous excitation frequency is of particular interest. The sweep excites just a single frequency at a single instance of time. As there is a unique mapping between time and frequency the time axis can be replaced by a frequency axis as shown in Fig. 7. It can be seen that the peaks occur around the resonance frequency f_s=50 Hz where the voice coil displacement is maximal.
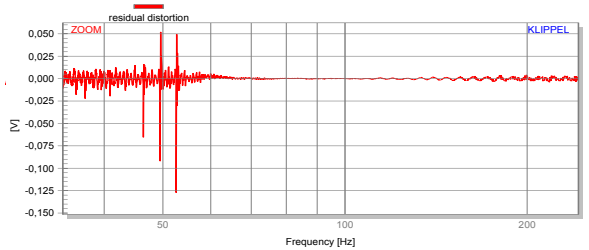

Fig. 7: Residual distortion signal *e(f)* versus the instantaneous frequency f of example B (continuous sine sweep).

Further insight into the particular distortion generation process can be gained by plotting the distortion signal versus the state variables of the system. The sound pressure signal and the voice coil displacement are two interesting candidates. Most of the triggered distortion are initiated by the movement of the cone. The displacement measurement requires a special sensor. Similar information can also be derived from the sound pressure measured in the near field as it is the second derivative of the displacement. Fig. 8 shows the residual distortion plotted versus excitation frequency and voice coil displacement. It can be seen that peaks occur if the displacement exceeds + 10 mm. At this point the diaphragm is hitting the metal screen of the driver.
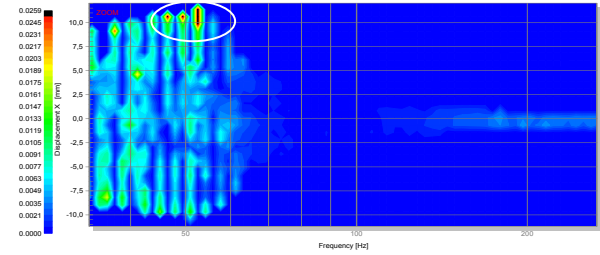

Fig. 8 Residual distortion signal *e(f)* versus instantaneous excitation frequency and voice coil displacement.

5.3 Distortion Measures

In order to compare distortion and to define acceptance thresholds the data has to be processed. At first we suggest general measures which are function of the time *t* and can be applied for any stimulus (music, noise, sweep, ...). For a sinusoidal sweep where we have a clearly defined mapping between time *t* and instantaneous frequency *f* we also derive special harmonic measures which are comparable with traditional THDN (total harmonic distortion + noise).

For the practical calculation of the distortion measures the time scale is divided into *N* small intervals. The boundaries of the kth interval are denoted by t_k and t_{k+1} with k= 0,1,…, N-2.

5.3.1 Instantaneous Distortion (ID)

A very useful distortion measure is obtained by referring the magnitude of the residual distortion to the root-mean-square (rms) value of the total signal giving the relative *instantaneous distortion* (ID)

$$d_{ID}(t) = \frac{|e(t)|}{y_{RMS}} \tag{2}$$

for the interval $t_k \leq t < t_{k+1}$ using

$$y_{RMS} = \sqrt{\frac{1}{t_{k+1}-t_k} \int_{t_k}^{t_{k+1}} y^2(t)dt}. \tag{3}$$

This measure preserves the fine structure of the distortion. However, the sign of the residual distortion is neglected to express the ID in dB or in percent.

Using a sweep as stimulus Eq. (2) the time can be mapped to instantaneous excitation frequency yielding the *instantaneous harmonic distortion*

$$d_{IHD}(f) = \frac{|e(f)|}{y_{RMS}} \qquad (4)$$

Fig. 9 shows the instantaneous harmonic distortion IHD (thin, red line) versus the instantaneous frequency for the example B discussed in section 5. The points where the diaphragm hits the metal screen can be seen as peaks well above the mean level.
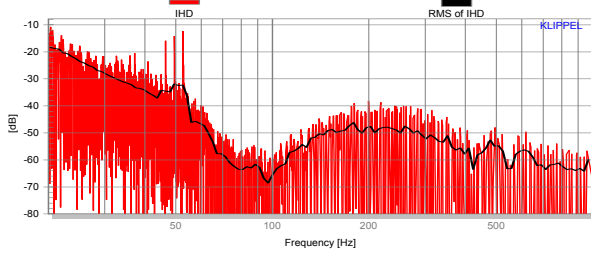

Fig. 9: Instantaneous harmonic distortion (IHD) and mean harmonic distortion (MHD) versus excitation frequency

5.3.2   Mean Value of Distortion (MD)

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

$$d_{MD}[k] = \frac{e_{RMS}}{y_{RMS}} \qquad (5)$$

for each interval $t_k \le t \le t_{k+1}$ using the rms-value of the residual distortion signal

$$e_{RMS} = \sqrt{\frac{1}{t_{k+1}-t_k} \int_{t_k}^{t_{k+1}} e^2(t)dt}. \qquad (6)$$

and the rms-value of the measured signal $y(t)$ according to Eq. (3). This measure may also be expressed in dB or in percent. Representing each interval by one number the mean distortion reduces the amount of data significantly. For stationary distortion the loss of information is negligible.

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

$$d_{MHD}[f_k] = \frac{e_{RMS}}{y_{RMS}}. \qquad (7)$$

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

The Fig. 9 shows the mean harmonic distortion of example B as bold line.

5.3.3   Peak Value of Distortion (PD)

Peaky, short time, low-energy distortion leads to large peaks in ID which are not represented by the mean distortion MD. In this case the peak value of ID in the interval $t_k \le t \le t_{k+1}$ is an valuable information that exploits both phase in amplitude information. The *peak distortion* (PD)

$$d_{PD}[k] = \frac{e_{peak}}{y_{RMS}} \qquad (8)$$

is the ratio between the absolute peak value

$$e_{peak} = \max_{t_k \le t < t_{k+1}} \left[\left| e(t) \right|\right] \qquad (9)$$

and the rms-value of the measured signal according to Eq. (3). It is also a relative quantity and can be expressed in dB and percent.

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

$$d_{PHD}[f_k] = \frac{e_{peak}}{y_{RMS}}. \qquad (10)$$

Fig. 10 depicts the PHD of example B. The PHD represents the main shape of IHD by using far less data points. Nevertheless, it contains all the important information.
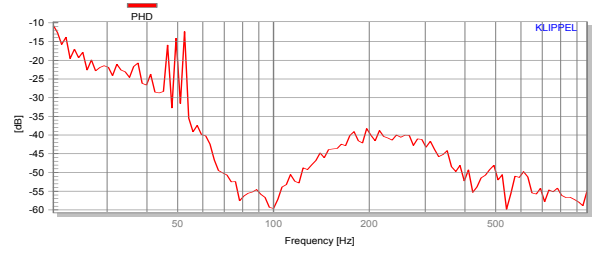

Fig. 10: Peak value of harmonic distortion (PHD) versus excitation frequency

5.3.4   Crest Factor of Distortion (CD)

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

$$d_{CD}[k] = \frac{e_{peak}}{e_{RMS}} \qquad (11)$$

is defined as the ratio between the peak value and the rms-value of the $e(t)$, as defined in Eq. (9) and Eq. (6), respectively. This measures uses the phase information of the harmonics and is almost independent of the power of the distortion.

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

$$d_{CHD}[f_k] = \frac{e_{peak}}{e_{RMS}} \qquad (12)$$

The CHD of example B is shown in Fig. 11. The crest factor is below 8-10 dB for the regular distortion. The triggered distortion at 50 Hz have a significant higher crest factor.
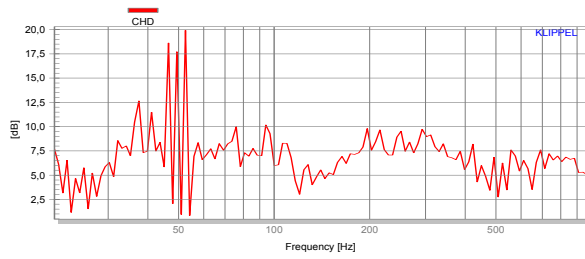
Fig. 11: Crest factor of harmonic distortion (CHD) versus excitation frequency

5.3.5 Instantaneous Crest Factor of Distortion (ICD)

In order to preserve the distortion fine structure we replace the peak value e_{peak} by the actual (absolute) value of $e(t)$ and define the *instantaneous crest factor of distortion* (ICD) by

$$d_{ICD}(t) = \frac{|e(t)|}{e_{RMS}} \qquad (13)$$

where e_{RMS} is defined in Eq (6). This measure is useful for investigating the relationship between crest factor variation and state variables (sound pressure or voice coil displacement) to derive more information about the physical cause of the distortion.

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

$$d_{ICHD}(f) = \frac{|e(f)|}{e_{RMS}} \qquad (14)$$

The ICHD of example B depicted in Fig. 12 shows a similar pattern as Fig. 11 but at much higher resolution.
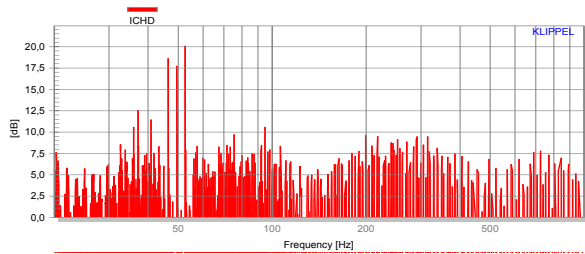
Fig. 12: Instantaneous crest factor of harmonic distortion (ICHD) versus excitation frequency

6   Diagnoses on Examples

6.1   Wire Beat

We take up the example A (see Fig. 2) where we have already discussed the spectrum of the steady-state response at 50 Hz. Using a sinusoidal sweep from 20 Hz up to 1 kHz the sound pressure output was measured in 1.5 s.
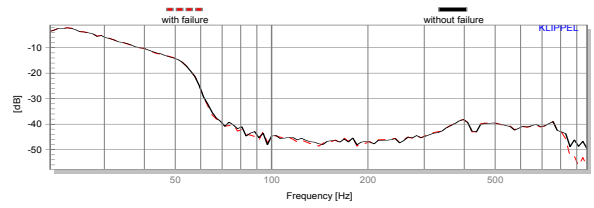

Fig. 13: Total harmonic distortion plus noise (THDN) of example speaker A with and without failure.

Fig. 13 shows the total harmonic distortion measured on the units with and without defect. The regular nonlinear speaker distortion dominates the THDN at low frequencies and there is no significant difference about 50 Hz. After modeling the linear behavior and regular nonlinearities of the motor and suspension and calculating the difference to the measured waveform we get the residual distortion $e(t)$. The mean harmonic distortion (MHD) is shown Fig. 14.
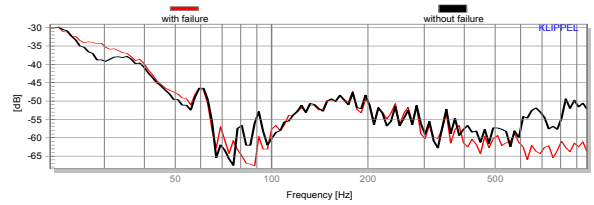

Fig. 14: Mean harmonic distortion (MHD) versus frequency of the driver A with and without defect

Like the total harmonic distortion (THDN) the curve of the unit with failure almost coincides with the curve of the properly working unit. At the critical frequency of 50 Hz there is only a minor difference of 1 – 2 dB which is compared to the other deviations not significant to recognize the defect.
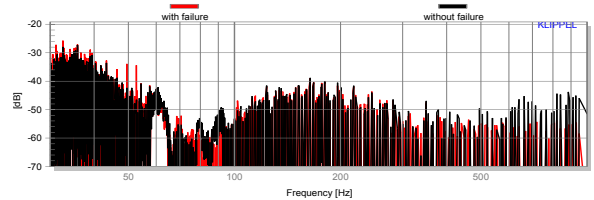

Fig. 15: Instantaneous harmonic distortion (IHD) versus frequency of the driver A with and without defect

However, the instantaneous harmonic distortion (IHD) depicted in Fig. 15 reveals two narrow peaks at 50 and 55 Hz. The distance between the two peaks is caused by the speed of the sweep used and corresponds with the temporal distance of the two transient events as shown in Fig. 16.
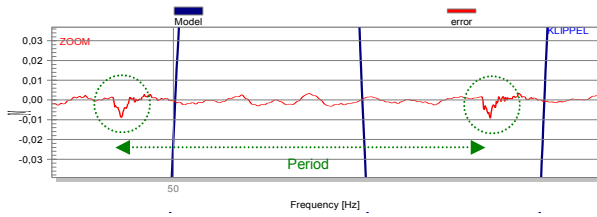
Fig. 16 Deviation between modeled and measured time response mapped versus frequency.

The bold line in Fig. 16 represents the modeled waveform *y'(t)* of the speaker output which is 100 times larger than the residual distortion depicted as thin line. Although modeling of the regular nonlinearities reduced the residual distortion by 35 dB at 50 Hz we still find stationary vibrations which are typical for regular driver distortion.
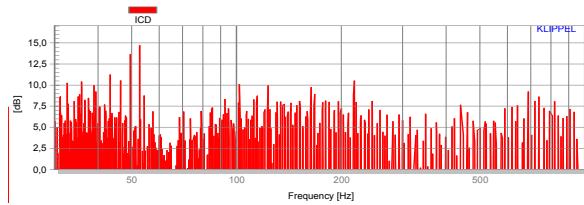

Fig. 17: Instantaneous crest harmonic distortion (ICHD) of defect loudspeaker A

The triggered distortion are quite visible in the time response and we can trace them more easily in the instantaneous crest factor as depicted in Fig. 17. The triggered distortion go up to 15 dB while regular nonlinear speaker distortion have a ICHD of about 6 - 10 dB.
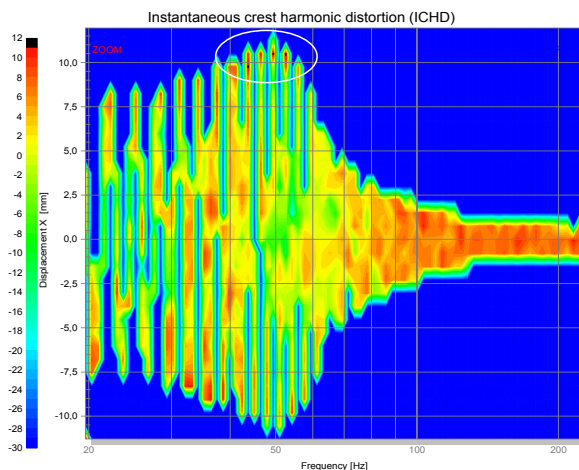

Fig. 18: Instantaneous crest harmonic distortion (ICHD) of defect speaker A versus frequency and displacement x

Fig. 18 shows the instantaneous crest factor mapped versus frequency and voice coil displacement. The triggered distortion occurs at +10 mm displacement when the wire hits the spider. Distortion with a high crest factor initiated at the negative

maximum of displacement is typical for a voice coil former hitting the back plate.

6.2   Voice Coil Rubbing

Any mechanical contact of the voice coil or voice coil former with the pole tips causes audible signal distortion and will damage the speaker eventually. Fig. 19 shows the instantaneous harmonic distortion (IHD) of a driver with and without this defect. Short but high peaks are also typical for this kind of distortion occurring over a relatively wide frequency range (40 Hz …120 Hz). The instantaneous distortion of the "good" driver varies also with frequency but have a much smoother fine structure. Comparing the mean harmonic distortion (MHD) would lead to a much smaller difference between both units.
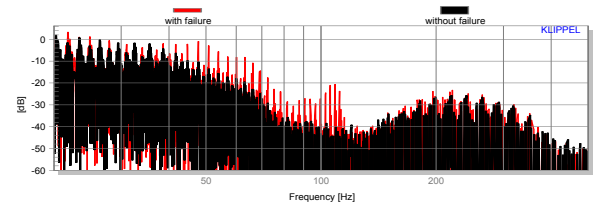

Fig. 19: Instantaneous harmonic distortion versus time (frequency) of a driver with and without voice coil rubbing

Clearly the harmonic distortion of the driver with and without rubbing have a different crest factor. The regular distortion and noise in the driver without rubbing cause a crest factor of 8 –10 dB shown as bold black lines in Fig. 20. The rubbing causes a significantly higher crest factor of 18 dB depicted in Fig. 20 as thin red lines. Since the crest factor can be interpreted on an absolute scale we may find a defect of the driver without having a reference speaker.
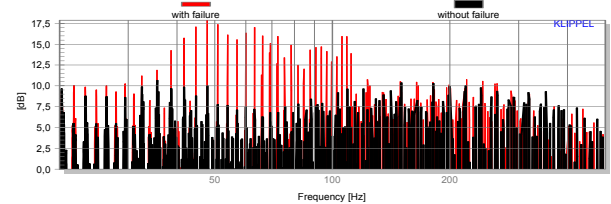

Fig. 20: Instantaneous crest factor of harmonic distortion (ICHD) of a driver with and without rubbing voice coil

Fig. 21 shows the time signal of the residual distortion over five periods at frequencies which correspond with a logarithmic frequency sweep from 35 to 55 Hz. The distortion pattern in each period is regular and shows a deterministic process. Fig. 22 reveals that at each negative peak of the voice coil displacement the crest factor (ICHD) goes far beyond 10 dB. This happens at low frequencies where the negative peak is more than 10 mm and above 100 Hz where the displacement is only 1 mm. Thus the negative turning point might cause a staggering of the coil. The Fig. 22 also reveals an asymmetrical displacement of the voice coil above the resonance frequency caused by the regular motor distortion.
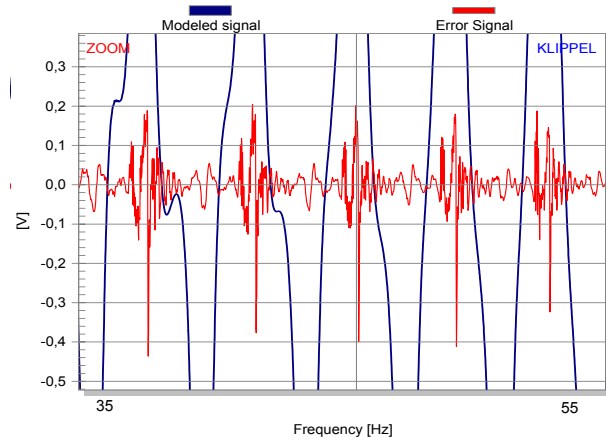
Fig. 21: Residual distortion *e(t)* and modeled signal *y(t)* versus time (frequency) of the speaker with rubbing voice coil
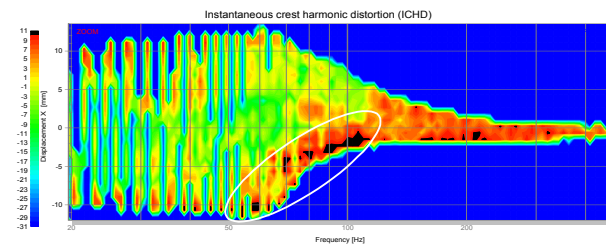

Fig. 22: Instantaneous crest factor of harmonic distortion (ICHD) versus frequency and displacement of the driver with voice coil rubbing

6.3   Defect Glue Joint

A loose part somewhere in the mechanical system may vibrate in its own mode. This defect also produces typical spikes in the fine structure of the instantaneous harmonic distortion (IHD) as shown in Fig. 23. They are 20 dB above the regular distortion between 100 and 300 Hz.
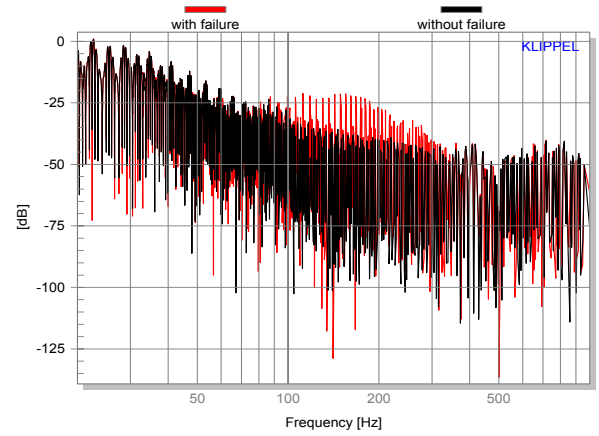

Fig. 23: Instantaneous harmonic distortion (IHD) of a speaker with and without a glue defect

Fig. 24 shows the crest factor of the harmonic distortion versus voice coil displacement and frequency. The crest factor exceeds 12 dB at the negative peak of the voice coil displacement between 100 and 300 Hz. The distortion generating mechanism is almost independent of the peak amplitude of the displacement which varies from 3 mm down to 0.5 mm. However, it clearly depends on the phase of the fundamental component. When the diaphragm is moved in positive direction (outwards) and the acceleration is maximal the two parts of the glue joint are separated and the surround starts vibrating at a particular frequency (about 3 kHz) determined by the mass and stiffness of the loose part.
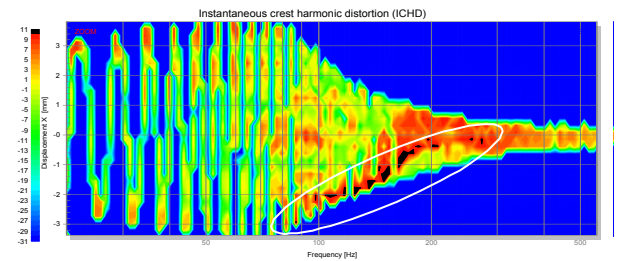

Fig. 24:Instantaneous crest harmonic distortion (ICHD) versus frequency and voice coil displacement of a driver with a defect glue joint
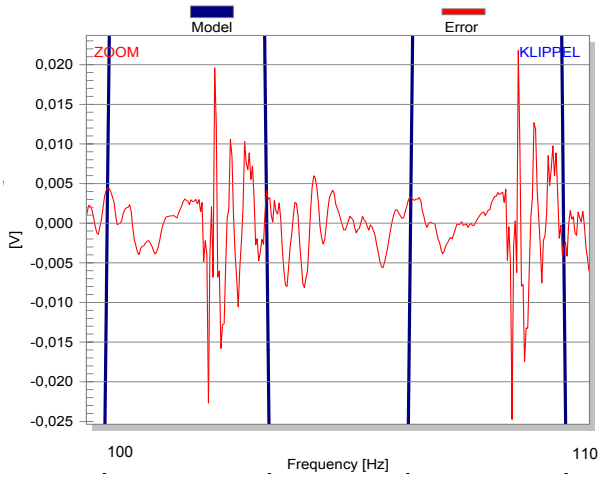
Fig. 25 Residual distortion *e(t)* signal versus time (frequency) of speaker having a defect glue joint

6.4   Defects in Electronic Circuits

The instantaneous distortion is also very useful for assessing electronic systems others than speakers. This may be illustrated on an amplifier producing almost constant total harmonic distortion THDN ≈ –56 dB over a wide frequency range as shown as bold black line in Fig. 26. However, the instantaneous harmonic distortion shown as thin red line reveals narrow peaks up to – 40 dB. Comparing the peak and rms-value of the distortion we find a crest factor of 16 dB.
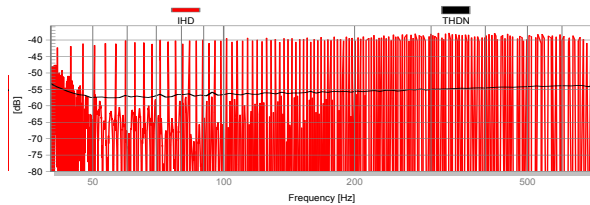

Fig. 26: Instantaneous harmonic distortion (IHD) and total harmonic distortion + noise (THDN) versus frequency

The peaks occur in each period of the measured signal as shown in Fig. 27. Plotting the instantaneous crest factor of harmonic distortion (ICHD) versus instantaneous voltage and frequency in Fig. 28 shows that the distortion are generated exactly at the negative maximum at all frequencies. This is caused by a limiting effect in the amplifier.
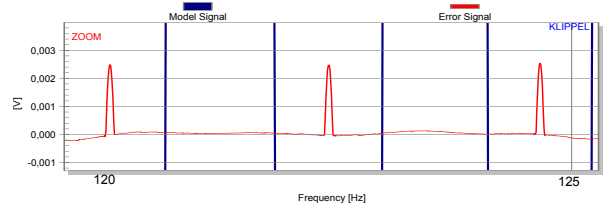

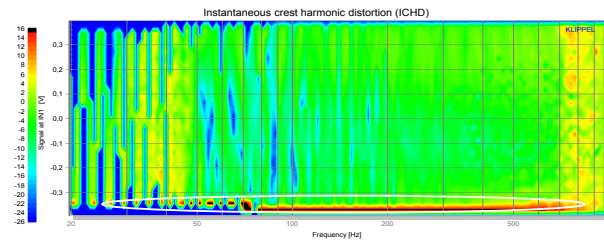Fig. 27: Deviation between modeled and measured signal

Fig. 28: Instantaneous crest factor of harmonic distortion (ICHD) of the electronic circuit

7   Conclusion

There are many sources for signal distortion in loudspeakers and in other electro-acoustical transducers. Some distortion have a small impact on the perceived sound quality and are tolerated by the listener as regular speaker distortion. Other kinds of (excessive) signal distortion are more easily audible and are directly related with a malfunction of the speaker. The mean power spectrum is a rather weak criteria to distinguish between both kinds of distortion and to identify speaker defects objectively. Exploiting both amplitude and phase information or comparing the measured with a desired response in the time domain allows to separate the excessive distortion more reliably from the regular distortion especially under the presence of noise. The residual distortion shows the fine-structure of the excessive distortion without any smearing effects caused by filtering or windowing. Statistical measures such as the rms-value and peak value can be applied to the residual distortion. Using a sinusoidal sweep excitation and a linear model the rms-value is comparable with traditional measures such as the THDN. The peak value of the instantaneous distortion summarizes both amplitude and phase information of the instantaneous distortion. The crest factor of the instantaneous distortion summarizes the phase information to a simple and meaningful number. The crest factor distinguishes stationary from transient signal distortion but is independent of the signal power. Since excessive distortion caused by rub and buzz have a more transient characteristic than the regular distortion the crest factor is a simple but reliable criteria for quality control using a fixed threshold (≈ 10 dB). The new measures derived from the residual distortion are also valuable for the speaker design. Plotting the distortion versus signal properties (instantaneous frequency and amplitude) shows the conditions (voice coil position) under which the excessive distortion is generated. The new measurement technique is not restricted to transducers but is also valuable for detecting triggered distortion produced by analogue and digital electronics.

8   References

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