Non-Linear Amplitude Modulation (NLAM) of Loudspeakers

Do you believe that a perfectly linear loudspeaker doesn't generate distortions?
That is wrong. This Q&A was the introduction of a first article about the Phase Modulation a.k.a. Doppler Effect. Here is now the second article about the Nonlinear Amplitude Modulation.

Even a perfectly linear loudspeaker produces two types of distortions because of the movement of its membrane. First Doppler distortions, a.k.a. Phase Modulation, and second distortions stemming from the Nonlinear Amplitude Modulation.
Doppler distortions generate non-harmonic spectral lines, which are independent from the listening distance and need to be eliminated for a good listening experience.
The Nonlinear Amplitude Modulation generates harmonic and intermodulation spectral lines, but decreasing with listening distance.
This article characterizes the Nonlinear Amplitude Modulation distortions and shows how to eliminate them.

Background:
A perfect sound reproduction requires a system able to translate the sound pressure of a source into a usually storable signal (e.g. an electrical audio signal) and translates that signal back into sound pressure without modification.
In order to do that system theory [1] tells that one needs **first** a **linear** and **second** a **time invariant** system.

One can fulfill the first requirement on linearity using a loudspeaker e.g. with Acceleration Feed Back (AFB) as described in [2]. By this technique the linearity of a loudspeaker can be improved such, that harmonic and intermodulation distortions are reduced to below -60 dB (0.1%) relative to the fundamental frequency.
Figure 1 shows the Sound Pressure Level (SPL) frequency response of an acceleration-controlled loudspeaker together with the harmonic distortions *k2* and *k3*.

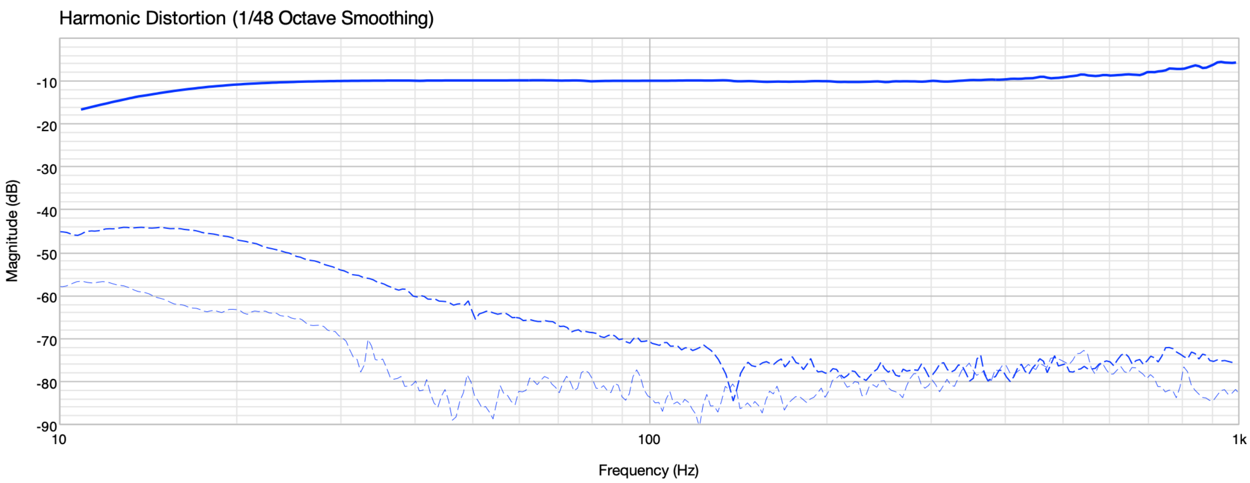

Figure 1: SPL Frequency Response of an Acceleration Controlled Loudspeaker SEAS L15 RLY/P with an AudioChiemgau AC-PAR75 Amplifier. Solid blue SPL amplitude, upper dashed line harmonic distortion *k2* and lower dashed line harmonic distortion *k3*.
Measurement parameters: AudioChiemgau ModeCompensator [6], Sweep Time 10 Seconds, Time Window 10 Seconds, Frequency resolution 0.1 Hz. The stimulus signal is High Pass filtered at 15 Hz (-3 dB) in order to limit the membrane amplitudes at frequencies below 15 Hz.

Obviously, with acceleration control technology, a sufficient linear system can be achieved with moderate effort. So the first mentioned **requirement on linearity** can be sufficiently fulfilled. Above 100 Hz the harmonic distortions are below -60 dB (0.1 %). However, at lower frequencies especially the harmonic *k2* is rising towards lower frequencies. The reason lies inter alia in the violation of the second above mentioned **requirement on time invariance**.
During recording of the sound, the membrane of a condenser microphone moves with an amplitude of about 1 μm at low frequencies (20 Hz) and high sound pressure levels (100 dB). The same is valid for the tympanic membrane of the human ear.
However, during play back a loudspeaker membrane may move up to 1 cm amplitude in order to reproduce the recorded sound with the above mentioned frequency and sound pressure level.

To put it in simple terms: We record at an almost fixed location in the sound space, but we reproduce at a variable location in the sound space. **This 1 to 10000 relation of the membrane amplitudes between recording and reproduction leads to well audible distortions, even for a perfect linear loudspeaker.**

The Generation of the Nonlinear Amplitude Modulation

Figure 2 visualizes the generation of the non-linear Amplitude Modulation.
The loudspeaker membrane travels with the amplitude s during reproduction of the sound.
As a consequence the distance D between the rest position and the Receiver (Ear, Microphone) varies by the travel amplitude s. Because of the inverse proportional law [3] between sound pressure and distance, the sound pressure varies at the receiver inversely to the actual distance.

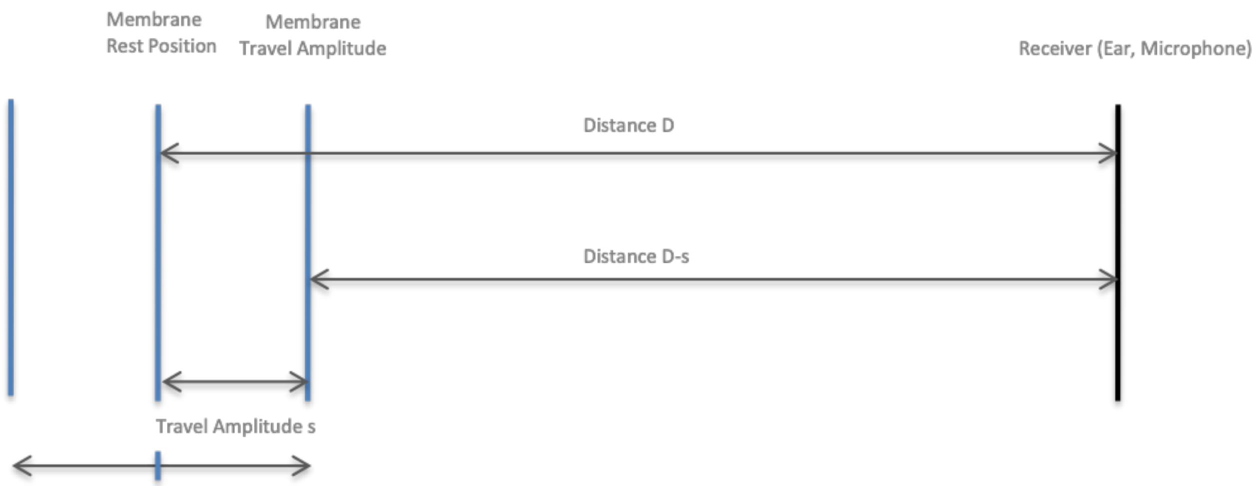

Figure 2: The membrane of the loudspeaker travels by s(t) and is located at a distance D to the receiver. The Phase Modulation is not taken in consideration as it is treated in another article [4].

Thought experiment:
One can imagine a high note being played back by the loudspeaker. Simultaneously a low note with considerable membrane travel amplitude is being played back. The location at which the high note is generated travels with the membrane position back and forth. Because of the inverse proportional law between sound pressure and distance, the high note is “louder”, i.e. has a higher sound pressure when the membrane is closer to the receiver and vice versa.
This principle is a general one and is valid for any note or frequency, even, when only one frequency is played back. The **amplitude** of the sound pressure at the receiver is modulated by the factor

$$\frac{D}{D-s(t)}$$

With the distance D between the loudspeaker membrane rest position and the receiver. The loudspeaker membrane moves as function of the time by s(t).
The above modulating factor is a non-linear function of time as an inspection of Figure 3 reveals. That is the reason one can talk about a non-linear Amplitude Modulation (NLAM) of the sound pressure at the receiver.

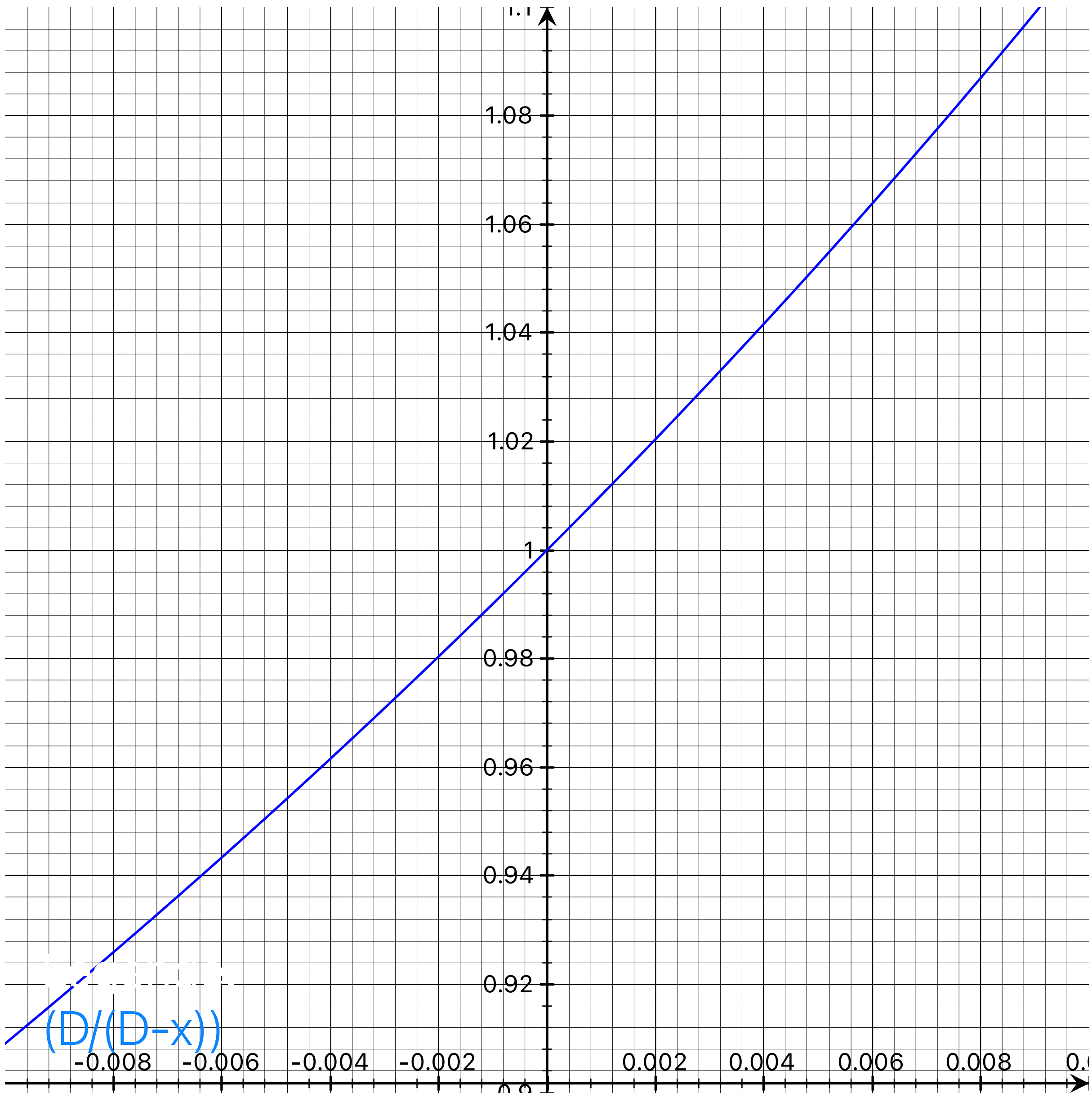

Figure 3: Amplitude modulating factor D/(D-s) as function of the amplitude of the membrane travel Amplitude s in Meters.

Consideration in the time and frequency domain

The time domain plot in Figure 4 shows in blue the SP at the receiver considering the inverse proportional law. The difference between the ideal sound pressure (red) generated by an acoustically steady membrane (an extremely large area membrane with negligible membrane travel amplitude) and the real sound pressure considering the inverse proportional law is shown in **green** (10 times magnified) at the receiver. The distance D is 10 cm for the example.

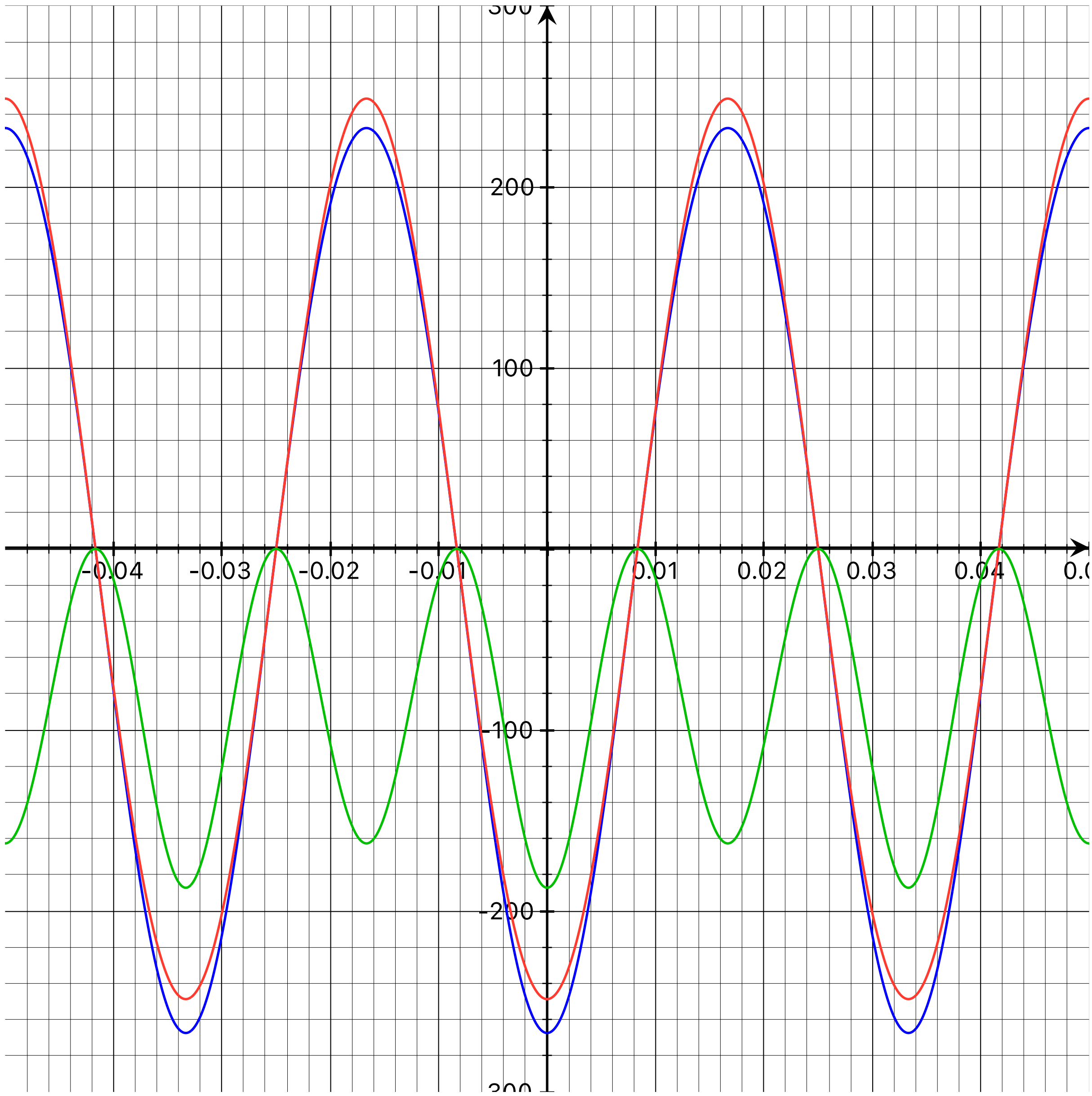

Figure 4: Time domain display (X axis: Time in Seconds, Y axis: SP in arbitrary units).
Red the ideal SP at the receiver generated by an acoustical steady membrane for a 30 Hz frequency with 7 mm membrane travel amplitude. **Blue** the real sound pressure waveform considering the inverse proportional law. **Green** (10 times magnified) the difference between the ideal and the real sound pressure at the receiver at a distance of D=10 cm.

Obviously this difference is always negative. It's minimum is at the positive and negative maxima of the waveform and zero at the zero crossings of the waveform. This difference is identified as the Nonlinear Amplitude Modulation (NLAM) generated by the moving loudspeaker membrane.

From Figure 4 it is obvious that the resulting main difference frequency is twice the fundamental frequency, i.e. mainly a second harmonic is generated.

In the Frequency Domain the Fourier Transformation of the waveform at the receiver in Figure 5 shows that there are further harmonic frequencies, however with smaller amplitudes.
The second harmonic for the above example is 29 dB (3.5%) below the fundamental frequency and the third harmonic is about 58 dB below the fundamental frequency amplitude.
There is a DC component generated too, which makes the compensation of that distortion difficult, as also the compensation function is generating a DC component. We come later to this.

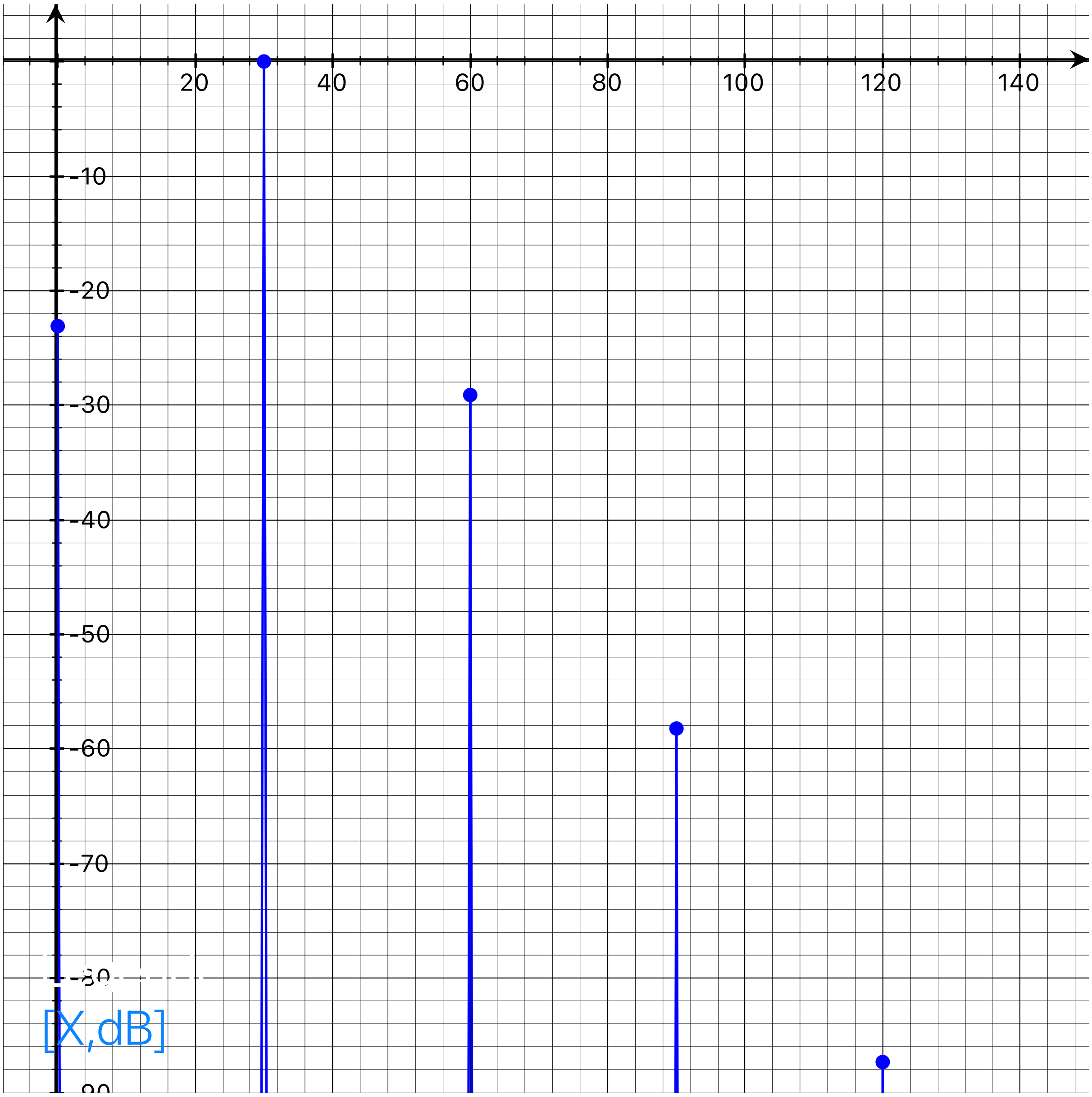

Figure 5: Frequency Domain plot of the Non Linear Amplitude Modulation (X axis: Frequency in Hz, Y axis: SPL in dB). The moving loudspeaker membrane generates harmonic distortions at the receiver. Displayed are a DC component and the harmonic spectral lines k2, k3 and k4.

Consequences for the near field measurement of loudspeakers

We see that the moving loudspeaker membrane creates, because of the inverse proportional law, a non-linearity for the SPL at the receiver, which results in the generation of harmonic frequency components and intermodulation in case more frequencies are involved. Figure 6 shows the resulting spectrum for two frequencies of 30 Hz and 300 Hz at the same SPL.

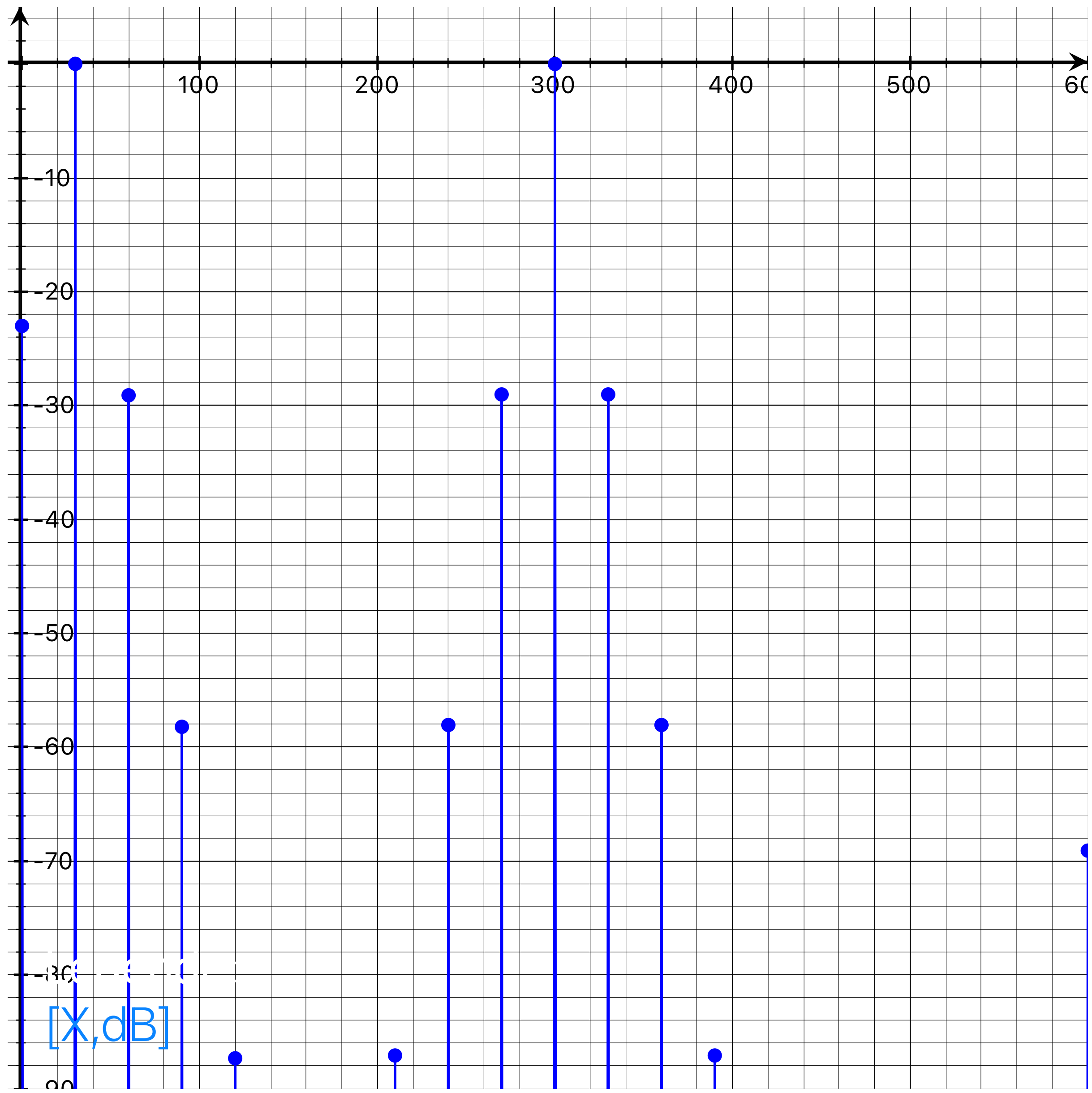

Figure 6: Intermodulation Spectrum between 30 Hz and 300 Hz at the same sound pressure level (X axis: Frequency in Hz, Y axis: SPL in dB).

It is obvious that the spectrum is filled with intermodulation products quickly when there are more frequencies present.

The calculation of the NLAM dominating second harmonic versus frequency in Figure 7 shows a drop by 12 dB per frequency octave, which corresponds exactly to the drop of the membrane amplitude as function of the frequency at constant SPL. A variation of the microphone distance, or the membrane travel amplitude results in graphs parallel shifted along the y-axis.

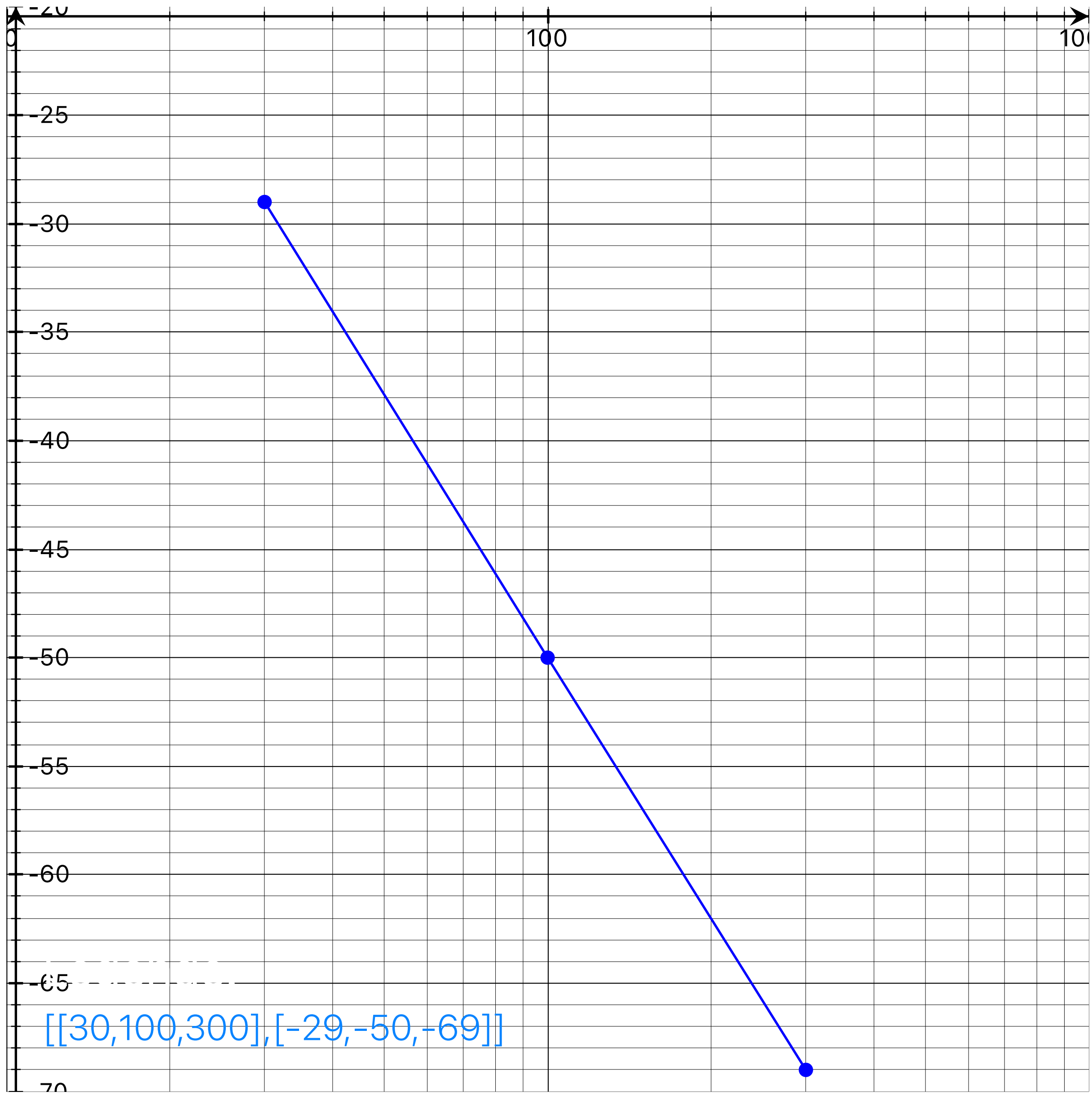

Figure 7: Frequency dependency of the generated second harmonic for constant sound pressure of the fundamental frequency.
X axis: Fundamental frequency in Hz (the displayed amplitude of the second harmonic ist at twice that frequency),
Y axis: SPL in dB wrt. the fundamental frequency amplitude following exactly the membrane travel amplitude at constant sound pressure.

From these results the following **general rules** for the effects of the Nonlinear Amplitude Modulation can be derived:

- The NLAM, i.e. the generated harmonic spectral lines are approximately proportional to the quotient of membrane travel amplitude and the distance but the exact dependency is non-linear. See Figure 2.

- The NLAM is independent from the frequency, but generates intermodulation products between different frequencies. Intermodulation products are non-harmonic and very disturbing to a good listening experience.

- The NLAM decreases with growing distance from the transmitter / loudspeaker by 6 dB per doubling of the distance and can usually be neglected above 3 Meters even for loudspeakers with small diameter woofers with significant membrane travel amplitudes.

- For distortion measurements the NLAM needs to be carefully considered, especially for near field measurements.

Consequences for listening

The significant relationship by a factor of approximately 10000 of the membrane travel amplitudes between recording and play back leads to a non-linearity, generating inter alia harmonic and intermodulation distortions. At deep notes and high SPL these distortions can easily dominate the distortions generated by a good loudspeaker chassis at relatively small listening distances. This may e.g. be the case for monitor rooms. As a consequence one must not go below a minimum listening distance (e.g. 3 Meters) to achieve a fairly undistorted sound experience.

Elimination of the NLAM

As the generation of the NLAM results in a multiplication of the sound pressure by the factor D/(D-s(t)), it can be compensated/eliminated, however exactly only for a defined listening distance D. The elimination is based on an amplitude modulator e.g. in a Digital Audio Processor [5] using the membrane position as control signal. By modulating (multiplying) the audio signal with the inverse NLAM factor (1-s(t)/D), it is possible to compensate (pre-distort and such eliminate) the NLAM for a defined distance D. This requires the exact knowledge of the membrane position s(t) at any point in time, which can be calculated as double integral of the audio signal versus time. Using acceleration controlled chassis, which have negligible phase of their sound pressure transfer function allows a nearly perfect cancellation of the NLAM.

Comparison of the two physical effects PM (Doppler Effect) and NLAM:

Both distortion mechanisms exist because of the 1 over 10000 ratio between the membrane travel amplitudes between recording and reproduction. A reproduction with an extremely large area loudspeaker with negligible membrane travel amplitude (e.g. 1 µm) would avoid these distortions.

Both distortions are proportional to the membrane travel amplitude, which is highest at low frequencies.

The Phase Modulation is independent from the listening distance. The two horizontal lines (red for 5 mm membrane amplitude, yellow for 2.5 mm membrane amplitude) show this in Figure 8. However the PM is proportional to the frequency, as the phase increases with frequency for a given membrane travel amplitude. Such it is mainly a problem for two-way loudspeakers with a high crossover frequency, but definitely a general problem for high-end loudspeakers.

The NLAM (blue line in Figure 8) decreases by 6 dB for each doubling of the listening distance. It is mainly a problem for close listening and measurement distances and can be neglected for listening distanced above 3 Meters.
The usual characterization of loudspeaker chassis at a measuring distance of 0.5m in an anechoic chamber, which is then re-computed for 1 m distance is definitely not adequate for the measurement of harmonic distortions at significant membrane travel amplitudes.

Figure 8: Sound Pressure Level in dB versus distance in cm, logarithmic x-axis).
The PM is independent from the listening distance (Bessel spectral lines red for 5 mm and yellow for 2.5 mm membrane amplitude), while the NLAM's dominating second harmonic is decreasing with 6 dB per doubling of the distance.
Different distance and membrane travel amplitudes result in NLAM graphs shifted parallel along the y-axis.

In conclusion:

At close listening / measurement distances the NLAM generated harmonic and non-harmonic intermodulation distortions may dominate distortions generated by the chassis itself.
A measurement of distortions of a loudspeaker in the near field needs therefore to carefully consider the NLAM.
The NLAM generated non-harmonic intermodulation spectral lines lines show the same level as the dominating second harmonic.

At larger listening distances the distance independent PM sets the minimum achievable distortion level, if the PM is not compensated.
The non-harmonic Bessel lines of the PM have significant amplitudes at any distance, while the non-harmonic intermodulation spectral lines of the NLAM drop 6 dB with each doubling of the distance.

Both distortion mechanisms set a limit for the dynamic of the measurement of harmonic and intermodulation distortions especially at low frequencies. E.g. 7mm membrane travel amplitude results at 1 m distance to -50 dB for the generated second harmonic, compare Figure 8.

Both distortion mechanisms generate a dense non-harmonic spectral floor, which result in the loss of transparency in real music signals. From the point of view of the author, High-End loudspeakers should compensate the distance independent PM and also the NLAM in case of listening distances below 3 Meters. At 3 Meters distance and 7 mm membrane travel amplitude the generated second harmonic is at about -59 dB.

AudioChiemgau offers all electronic modules for building High-End acceleration (motion) feed back loudspeakers including a Digital Audio Processor which can eliminate both, the Doppler effect of the moving loudspeaker membrane as well as the NLAM.

References:

1. Karl Küpfmüller, Wolfgang Mathis, Albrecht Reibiger: *Theoretische Elektrotechnik – Eine Einführung*. 18. Auflage. Springer, 2008, ISBN 978-3-540-78589-7.
2. https://audioxpress.com/article/improving-the-transfer-function-of-electrodynamic-loudspeakers
3. Damping of sound level (decibel dB) vs. distance: http://www.sengpielaudio.com/calculator-distance.htm
4. https://audioxpress.com/article/Doppler distortions eliminated (TBD)
5. Web page www.AudioChiemgau.de -> Digital Audio Processor AC-DAP01
6. https://audioxpress.com/article/measuring-loudspeaker-spl-response-and-harmonic-distortion-at-low-frequencies