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# Implementation and evaluation of a fast method for THD analysis of sound transducers under demanding operating conditions

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## Abstract

The total harmonic distortion (THD) is an important parameter to specify an instrument performance. In general, this parameter is specified in a single frequency and as the maximum level allowed in the operational range of the device under test (DUT). Müller (2007) formulated the sweep level method for fast THD evaluation, in which THD vs. level is obtained using a single tone modulated by a sweep level envelope. Thus, is possible to evaluate the THD in the function of carrier frequency and also of the input level of the system. Moreover, the relationship between the input and output levels can be used to analyze the non-linear behavior of the DUT. The method was validated using both numerical and experimental comparison with the classical stepped sine method. In the numerical model, a hyperbolic tangent was applied to the excitation signal to simulate non-linear behavior. The most important conclusion obtained refers to the level parametrization that was made in an increment of the signal amplitude per wave cycle, instead of increment per time unit. Using an increment of 0,05 dB per wave cycle, the maximum error between sweep level and stepped sine was less than 0,01%. Using a loudspeaker and microphone, the THD of the system was tested in the frequency range between 125 Hz and 4 kHz. The results for the maximum THD vs level presented strong frequency dependence. Thus, the linear operational range of electroacoustical transducers is frequency dependent and it is important to specify for high fidelity applications, mainly if loud signals must be reproduced by the system. As the next step, it is intended to purpose a standardization of THD measurement and results presentation. So, is possible to offers a more detailed parameters to the customers.

**Keywords:** harmonic distortion; electroacoustic measurement; non-linearity.

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## 1. INTRODUCTION

The Linear and time-invariant systems (LTI) theory is very important in electroacoustic products, because these systems can be entirely characterized by their impulse response, and respective frequency response and transfer function[1]. However, as no system is perfectly linear or deterministic in any circumstance, real systems are modeled by an deterministic and linear approximation in a specific operation and frequency range.

Furthermore, in some applications non-linearity can be desired: microphone coloration in music applications [2], dynamic controllers such as limiters, compressors and expanders [3] can be cited as examples. These devices are used in artistic contexts and rarely a metrological rigor is required. In fact, these processes are governed largely by subjective aspects in these contexts.

On the other hand, non-linear effects are completely undesired in measurement instruments, such as transducers. For instance, in microphones and loudspeakers used for acoustical measurements, non-linear effects cause systematic errors in the obtained data. Commonly, these effects are complex to remove mathematically and the measurements could be strongly impaired. Also, non-linear effects are detrimental to accuracy in audiological tests, such as pure tone audiometry. While very high sound pressure levels must be produced to stimulate the hearing system, especially at low frequencies and for hearing impaired people, loudspeakers must be fairly linear and should not produce frequencies others than those of the intended pure tone excitation signal.

Generally, a physical LTI system is linear within an operational range limited by its inherent background noise at low excitation levels and the total harmonic distortion (THD) at high excitation levels. Figure 1 illustrates this range in the frequency domain. For signals with low amplitudes, a minimum Signal-to-noise ratio (SNR) is required for each particular application, as well, for high amplitude signals, a maximum acceptable THD is required. Thus, if background noise or the system excitation is too high, the measurement quality can be compromised.

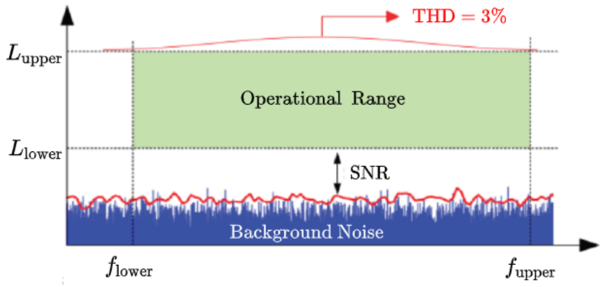



**Figure 1:** Illustration of the operational range of a generic system with a desired SNR and a THD of 3 %. Source: [4] (adapted).

Normally manufactures provide the specification of the background noise as a global value and a single valued THD at a single frequency and amplitude. Nevertheless, for high accuracy applications, more detailed information about the devices' performance must be available.

The most common method to achieve THD at different levels and frequencies is the step-sine method: tonal signals with different discrete amplitudes are used to excite the device under test (DUT) and, as result, the THD is obtained at discrete frequencies and levels. This method demands a lot of time and can stress the DUT due the long time working with high energy tonal signals. So, this method can cause some undesired time variances in the DUT, and bias will be appear in the results. An example is the power compression in loudspeakers, where the temperature of the coil increases during the measurement and the sensitivity of the device is reduced. Over longer high power excitation periods, these devices can therefore present variation in the systems parameters as presented by [5].

To deal with this situation, Müller [6] *et. al.* proposed the Sweep Level Method (SLM), where the DUT is excited by a tonal signal modulated by an envelope curve. The result obtained is the THD over a continuous level domain at a single frequency. Repeating the experiment with different carrier frequencies, a multidimensional analysis of the THD can be performed over the

[1]Commonly, transfer function and frequency response are defined as synonyms. However, a more precise definition is found in Oppenheim [1], where the frequency response and transfer function are obtained respectively by applying the Fourier and Laplace transforms on the impulse response of the system.

domain of discrete frequencies and continuous levels.

Besides giving THD, the method provides the possibility to analyze other parameters: for instance, the relationship between input and output signal energy is a very good indicator of linearity of the DUT. Also, the spectrogram of the DUT's response is an easy way to visualize the harmonic generation over the input level, as shown in Figure 2. In this case, a hyperbolic tangent system is simulated and, at low amplitudes, the output is very close to input and a linearization is possible. For high amplitudes, harmonics appear and the system can not be approximated by linearization anymore.

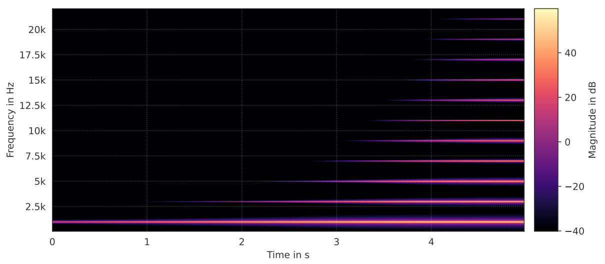



**Figure 2:** Spectrogram of the output signal of an hyperbolic tangent function $y(t) = \tanh\{x(t)\}$. The excitation signal $x(t)$ is a sweep level with -30 dBFS at start, 20 dB at the end, carrier frequency of 1 kHz and 5 seconds of duration. Spectrogram calculated with 1024 samples per block and 50% of overlap between blocks.

In this work we performed an analysis of the method, starting with a review of the THD concept and explaining the SLM. Then our algorithm that provides a mean for automatic calculation of excitation and response parameters is discussed. Finally, numerical and experimental results are presented and discussed.

## 2. A SHORT REVIEW OF TOTAL HARMONIC DISTORTION

In a perfect linear system the response to any tonal signal is another tonal signal with the same frequency and some amplitude factor and phase shift. In fact, a tonal signal is mathematically defined as an eigenfunction of any linear system [1].

If the system shows some non-linear behavior, its response to a tonal signal will feature harmonics and the waveform will be distorted, as illustrated in Figure 3. The amount and amplitude of the harmonics tend to raise with the increase of the excitation signal's amplitude. This behavior is illustrated in Figure 2.

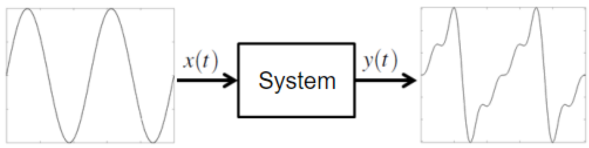



**Figure 3:** Illustration of a system excited by a tonal signal $x(t)$. The system's response is the signal $y(t)$ with harmonic distortion. Adapted of [7].

In this context, the Total Harmonic Distortion (THD) is a quantity largely used to describe the amount of non-linearity of a system's response when excited with a tonal signal. Mathematically, the THD of a signal, that can be the system's response, is expressed by:

$$\mathrm{THD} = \frac{\sqrt{\sum_{k=2}^{N} X_k^2}}{X_1}, \tag{1}$$

being $X_k$ the amplitude of the $k$-th harmonic of the signal, $X_1$ the amplitude of the first harmonic and $N$ the number of harmonics used in the calculation.

Periodic signals can be represented by the sum of a finite or infinite number of harmonics, generally with decreasing energy as the order of the harmonics increases. Therefore, any periodic signal can be reasonably well represented by a finite number of harmonics. For very high accuracy applications, a convergence analysis increasing the harmonic order can be performed.

This analysis can be made estimating the relative error between the THD with $N$ and $N+1$ harmonics:

$$\varepsilon_N = \frac{|\mathrm{THD}_{N+1} - \mathrm{THD}_N|}{\mathrm{THD}_N}, \tag{2}$$

where $\varepsilon_N$ denotes the relative error for the THD with $N$ harmonics. As THD is always positive and increases with increasing harmonic number, the absolute operator in $|\mathrm{THD}_{N+1} - \mathrm{THD}_N|$ can be omitted. Thus, Eq. (2) can be expressed as:

$$\varepsilon_N = \frac{\sum_{a=2}^{N+1}}{\sum_{a=2}^{N}} - 1. \quad (3)$$

Using this formulation, an iterative algorithm can be implemented using an adequate stopping criterion to estimate THD. Another option is to use the entire harmonics obtained in the FFT spectrum. However, the result can diverge for higher harmonics because the background noise can be high compared with the lower distortion energy caused by the system.

Another problem can be found for higher fundamental frequencies, close to the sampling frequency. In this case, all harmonics exceeding the Nyquist frequency must be rejected by the anti-aliasing filter of the ADC and the obtained THD may be underestimated. Therefore, for high frequencies, an oversampling may be required to obtain a good estimation of the system's THD.

Figure 4 shows the convergence analysis and THD's relative and absolute error for a square wave signal. The analytical THD of this signal is approximately 48.3% [8] and allows to compute the absolute error in THD estimation using the FFT spectrum, given by:

$$\varepsilon_{N,\mathrm{abs}} = \frac{0.483 - \mathrm{THD}_N}{0.483}. \quad (4)$$

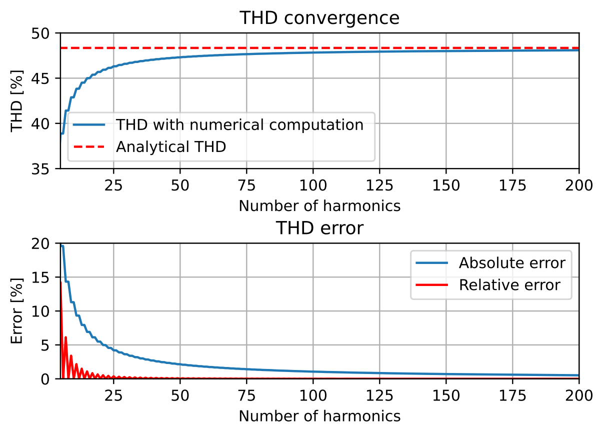



**Figure 4:** THD's convergence analysis of a square wave signal.

The absolute THD error given in Figure 4 is not smooth but stepped because even harmonics are computed by the algorithm and these frequency components are null in the square wave. A similar effect is observed in the relative error curve in Figure 4, being the error very close to zero for even harmonic components. This phenomenon can be a problem if there is significant noise in the measurement, because spurious energy is summed to the harmonic components and the THD is overestimated. Therefore, the THD plus the noise (THD+N) is more useful to estimate noise and THD together, but this quantity is not representative to evaluate the non-linear behavior of systems.

One way to deal with the noise contamination can be to set a minimum energy level for each harmonic to be computed as distortion. Nevertheless, high THD appears at high excitation amplitudes, generally. So a good estimation is provided in most cases.

## 3. THE SWEEP LEVEL MEASUREMENT METHOD

As cited previously, the idea behind the fast THD measurement is to excite the DUT with a tonal signal modulated by an envelope. An overview of the entire process is illustrated in Figure 5: a tonal signal is multiplied by an envelope and excites the DUT. So the DUT's response $y(t)$ is divided by the envelope and the signal analysis is performed using the excitation, response and corrected response signals.

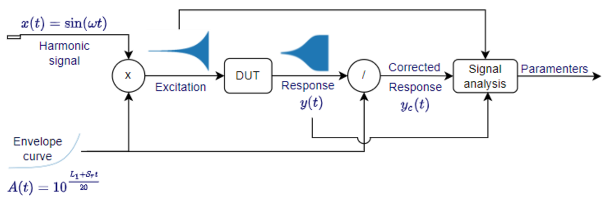



**Figure 5:** Overview of the SLM.

The main parameters obtained are:

- DUT's input level: calculated from the excitation signal $x[t_n]$
- DUT's output level: calculated from the response signal $y[t_n]$
- Output THD: calculated from the corrected output signal $y_{\mathrm{corr}}[t_n]$

The SLM method can be divided in three steps: excitation synthesis, system measurement and signal analysis, as shown in the subsequent subsections.

### 3.1 Excitation synthesis

The excitation signal can be expressed as:

$$x[t_n] = A[t_n]\sin(2\pi f_c t_n), \tag{5}$$

being $f_c$ the carrier frequency of the signal, $A[t_n]$ the time-varying envelope of the signal and $t_n$ the discrete time vector with a defined number of samples spaced in $1/f_s$ steps, where $f_s$ is the sampling rate of the signal.

An envelope function which provides a linear amplitude increase in the dB scale is given by:

$$A[t_n] = 10^{\frac{L_1 + S_r t_n}{20}}, \tag{6}$$

being $L_1$ the initial amplitude in dB and $S_r$ the amplitude build up ratio or sweep rate, expressed in dB/s. A good parameter to evaluate the method's accuracy is the wave period sweep rate in dB per wave period, given by:

$$S_{rp} = S_r f_c \quad [\text{dB}/\text{wave period}], \tag{7}$$

which defines the change in level in each period of the signal. Fixing this parameter, high carrier frequencies require more time than lower frequencies to keep the same $S_{rp}$.

To avoid smearing and leakage artifacts in the spectrum of the signals, commonly an optimal time-window is applied. However, as the signals are time sampled, the carrier frequency can be tuned as a sub-multiple of the sampling rate to ensure a perfect periodicity in the digital version of the excitation and response signals [6]. To achieve this, a desired carrier frequency $f_{cd}$ can be set. Therefore, the closest sync frequency of $f_{cd}$ is given by:

$$f_{c,\text{sync}} = \frac{f_s}{\text{round}(f_s/f_{cd})}. \tag{8}$$

Rounding the desired carrier frequency to its respective synchronous version allows to obtain a spectrum with an entire number of periods within a rectangular window. To divide the signal in blocks with entire periods, the local maxima of the signal can be used.

Another good practice in the signal's synthesis is to set a fade-in and fade-out at the start and end of the signal, respectively. So, abrupt excitation of the DUT is avoided. In the signal analysis stage, the fade-in and fade-out can be cut from the excitation and response signals. An illustration is shown in Figure 6 and more details of the division into blocks will be discussed in Section 3.3.

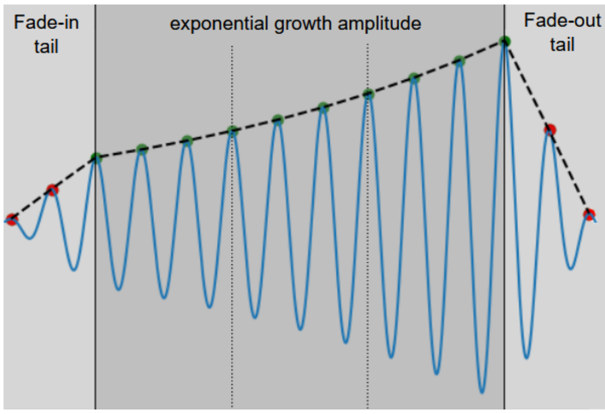



**Figure 6:** Illustration of a sweep level signal with a fade-in tail at the beginning, exponential amplitude increasing and fade-out tail at the end.

### 3.2 System measurement

Figure 7 shows the systems measurement flowchart. The excitation signal is converted to the continuous domain by the digital-to-analog converter (DAC) and excite the DUT. The systems response is converted to the digital domain using an Analog-to-digital converter (ADC).

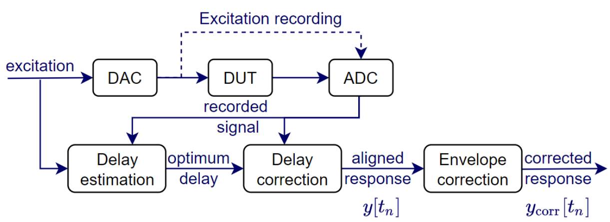



**Figure 7:** System measurement schematic.

For accurate measurement a signal alignment must be performed to compensate delays caused by the measurement chain. An optimal delay can be obtained by the cross-correlation between excitation and response signal as illustrated in Figure 7. In this case, to obtain a good estimate of the system's latency a broadband signal must be used, being exponential sweep sines a very good option. Exciting the measurement chain with a single frequency signal can generate erroneous estimations of the delay. Another option is to record the DUT's input signal, as illustrated by the dashed line. This approach is similar to the 2 channel FFT described in [9], so a good synchronization is provided. An advantage of recording the excitation signal is that a direct relationships between input and output of the DUT can be obtained, avoiding the possible bias of the measurement chain, for instance, the gain of pre-amplifiers used in the ADC and DAC. Another example is the microphone's THD measurement, as described in [2, 7]. In this case, a reference microphone is used to record the sound pressure, which is the input signal of the microphone under test. This experiment requires a high-performance loudspeaker to ensure an input sound pressure with THD as low as possible to evaluate only the THD of the microphone under test .

### 3.3 Signal analysis

The first step of the analysis is to divide the excitation and response signals into blocks. Using the synchronous carrier frequency, the blocks' limits can be obtained using the local maxima of the excitation signal, as shown in Figure 6. In this case, each block contains 3 periods. The number of periods of each block is an important parameter of the SLM. However, if the excitation signal is recorded (Figure 7), the local maxima can not be used to divide the signal in whole multiples of periods due to some noise and quantization errors. In this case, a better way to divide the signal into blocks is to use a fixed number of samples per block, such as:

$$N_{\text{samples,block}} = \frac{f_s}{f_{\text{c, sync}}}, \tag{9}$$

being $N_{\text{samples,block}}$ an integer if the carrier frequency is synchronous as expressed in Eq. 8.

The input signal's envelope can be obtained directly from the signal synthesis parameters, as shown in Section 3.1, Eq. (6). Nevertheless, if the DUT's input signal is recorded together with the response, the envelope is easily estimated by interpolating the local maxima of the signal in the time vector $t_n$ or index vector $n$ of the signal. Another form to estimate the envelope is computing the RMS value of each excitation block.

The next step is obtain the corrected output signal $y_{\text{corr}}[t_n]$ as illustrated in Figure 7 and expressed by:

$$y_{\text{corr}}[t_n] = \frac{y[t_n]}{\bar{A}[t_n]}, \tag{10}$$

being $\bar{A}[t_n]$ the estimated envelope curve of the recorded signal.

Having the signals segmented in $k$ blocks, excitation levels $L_{\text{in}}[k]$, response levels $L_{\text{out}}[k]$ and corrected response levels $L_{\text{out,corr}}[k]$ can be calculated from the root mean square (RMS) in each block:

$$L_{\text{in}}[k] = 20\log_{10}\{\text{RMS}(x_k[t_n])\}, \tag{11a}$$

$$L_{\text{out}}[k] = 20\log_{10}\{\text{RMS}(y_k[t_n])\}, \tag{11b}$$

$$L_{\text{out, corr}}[k] = 20\log_{10}\{\text{RMS}(y_{k,\text{corr}}[t_n])\}. \tag{11c}$$

In a perfectly linear system, $L_{\text{out, corr}}[k]$ is a constant function, because the correction curve just cancels the effect of the envelope. However, in the non-linear range of real systems, commonly this parameter will decrease as the input level increases.

Finally, the THD of the corrected response signal is calculated for each block of the signal:

$$\text{THD}[k] = \frac{\sqrt{\sum_{m=2}^{N} Y_{k,\text{corr}}^2[m]}}{Y_{k,\text{corr}}[1]}, \tag{12}$$

being $Y_{k,\text{corr}}^2[m]$ the $m$-th amplitude of the FFT component of block $k$ of the corrected excitation signal and $Y_{k,\text{corr}}[1]$ the amplitude of the fundamental frequency of the same block.

## 4. MODEL VALIDATION

To evaluate the accuracy of the SLM, a simulation was performed comparing the results with the stepped sine method. A hyperbolic tangent function was applied for both sweep level and stepped level signals, and THD's obtained were compared. Signals where sampled with 44.1 kHz.

Figure 8 shows the results for SLM and step sine using a signal with 100 Hz carrier frequency and sweep level increase of 0.5 dB/ wave period. The maximum absolute difference between stepped and SLM THD is close to 0 dBFS, when THD is close to 15%.

This discrepancy happens close to the inflection point of the input level vs THD curve. So, a possible explanation is the fast increase of THD with the change in the input level. In other words, the THD is sensitive to small changes in the input amplitude.

A bigger discrepancy happens above 20 dBFS input level, reaching more than 0.03%. Possibly this happens because the envelope correction reduces the amplitude of the output signal, once the system is saturated and the output is an almost square wave. Nevertheless, the THD is very high in this region, and the model validation is not useful in practical applications for linear systems evaluation.

In fact, most usually amplitudes are given for 3% or 10% THD, as shown in gray in Figure 8.

Figure 9 shows the THD difference $|\Delta\text{THD}_{\text{step level vs SLM}}|$ for 100 Hz and 1 kHz carrier frequencies. Curves are similar for both cases, with the difference tending to increase with input level and leveling off at high amplitudes. However, the 100 Hz curve presents less difference and anti-peaks. This phenomenon is related to the relationship between the carrier frequency and sampling rate.

In the region between 3% and 10% of THD, the difference tends to decrease by increasing the excitation length. So, in each particular application, the user must choose a trade-off between measurement time and accuracy.

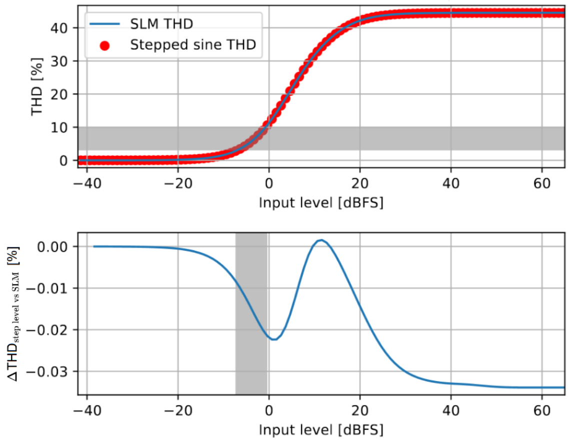



**Figure 8:** THD and THD difference ($|\Delta\text{THD}_{\text{step level vs SLM}}|$) of a hyperbolic tangent function excited by a sweep level signal with 100 Hz carrier frequency and 0.5 dB per wave period amplitude increase. Gray area corresponds to THD between 3% and 10%. SLM THD obtained from blocks of 2 wave periods.

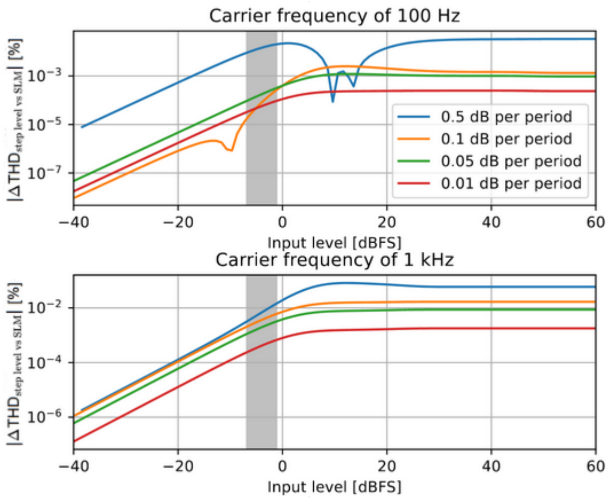



**Figure 9:** Absolute THD difference ($|\Delta\text{THD}_{\text{step level vs SLM}}|$) between SLM and step sine for 100 Hz and 1 kHz carrier frequencies. Gray area corresponds THD between 3% and 10%. SLM THD obtained in blocks of 2 wave periods.

Figure 10 shows the absolute difference $|\Delta\text{THD}_{\text{step level vs SLM}}|$. The worst case is found for 8 periods/ block, amounting to difference close to 5%. It can be concluded that the number of periods in each block must be as small as possible. Additionally, it is noted that the synchronous frequency is also a very important requirement in favor of small signals with a rectangular window. Otherwise FFT smearing and leakage can compromise strongly the results. Also, noise components tend to be more suppressed in long-duration tonal signals.

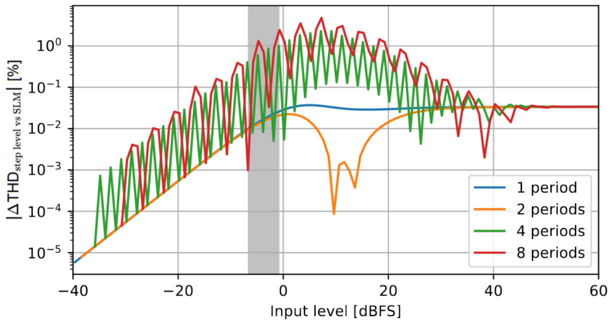



**Figure 10:** Absolute difference $|\Delta\text{THD}_{\text{step level vs SLM}}|$ using distinct wave periods per block. Carrier frequency of 100 Hz and increasing amplitude of 0.5 dB per wave period.

## 5. EXPERIMENTAL ANALYSIS

The SLM to obtain THD was verified using the setup illustrated in Figure 11. The signal was generated by a computer, converted by an audio interface and send to the amplifier and loudspeaker. For the recording, the microphone was connected to a mixer for signal conditioning. So, the response signal is recorded using the ADC of the audio interface. Signal analysis was performed using PyTTa, a Python based toolbox for acoustics [10][2].

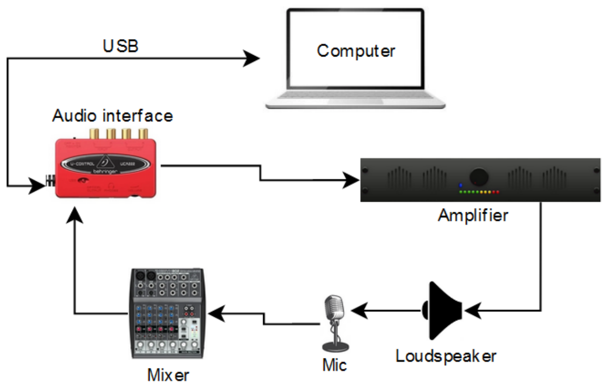



**Figure 11:** Workbench flowchart of the measurement setup.

Figure 12 shows the THD vs. input level for 125 Hz and 1 kHz carrier frequencies. For the 125 Hz signal, the sweep rate per wave period do not causes large effects on THD. However, for the 1 kHz signal THD tends to increase with increasing sweep rates. This phenomenon can be caused by the signal duration, which is much shorter for high frequencies (see Equation 7).

Another important effect is the influence of the background noise. As the measurements were not conducted in a controlled acoustic room, THD tends to increase for very low input levels. This phenomenon is related to the signal to noise ratio and not to non-linear behavior of the system. So, to improve the THD estimation at low SNRs, a noise controlled room is required. Nevertheless, for most transducers under demanding operating conditions a good SNR is achieved.

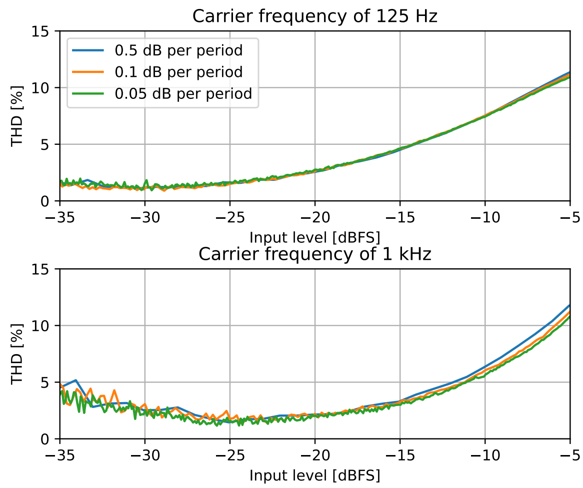



**Figure 12:** Experimental SLM THD of the system with different sweep rate levels and carrier frequencies.

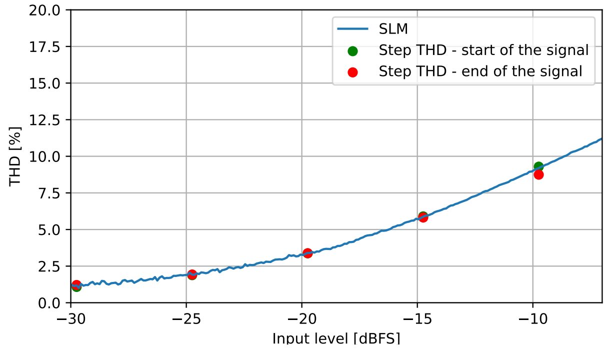
Figure 14 show the comparison between THD obtained with SLM and stepped method. For the stepped method, tonal excitation signals with 10 s of duration were generated with different amplitudes. Two blocks were cut off from the response signal, the first at beginning and the second at the end.

Results in Figure 14 show an almost perfect match between SLM and stepped THD, using the initial block of the response signal in the stepped THD approach. However, in the second block, the THD tends to be underestimated noticeably for high input levels. The effect is more prominent for the 1 kHz signal and this happens mainly due to the heat build-up at the voice coil [3], increasing the loudspeaker's impedance. It is an important time variance effect and must be taken into account for high-performance applications.

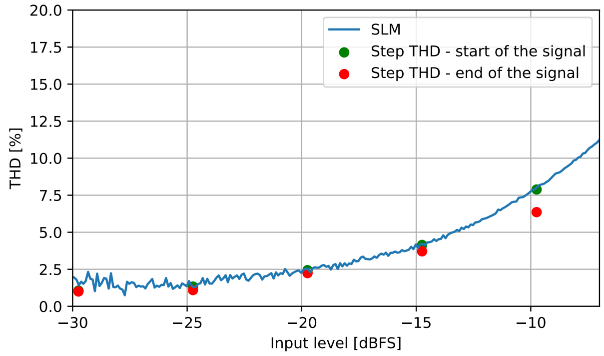
Figure 14 show the input levels that result in 3% and 10% THD for different carrier frequencies

[2]The toolbox is available in: https://github.com/PyTTAmaster/PyTTa

[3]Effect often called Power compression.




**(a)** $f_c$ =125 Hz




**(b)** $f_c$ = 1 kHz

**Figure 13:** Step and SLM THD of the system with sweep rate of 0.05 dB/wave period. Step sine with 10 seconds.

and sweep rates. In this case, sweep rate was set as level per time unit, to have signals with equal duration in the whole frequency range. Using sweep rate/wave periods, low frequency signals would be very long and high frequency signals very short.

Results show a few discrepancies using different sweep levels. This difference can be explained by the difference in the increasing THD curves shown in Figure 12, that cause a considerable difference in the points at 3% and 10% THD. To investigate better this effect, a more detailed uncertainty analysis can be conducted with repetitions and respective dispersion analysis.

Another very interesting effect in this results is the frequency dependence of the level at 3% and 10%. For 3% THD the maximum level is close to -17 dBFS for 500 Hz and the minimum is close to -30 dBFS for 2 kHz, resulting in a range close to 13 dB in the frequency range analyzed. For 10% THD, the maximum level is close to -2 dBFS and the minimum is close to -19 dBFS , at 500 Hz and 4 kHz respectively.

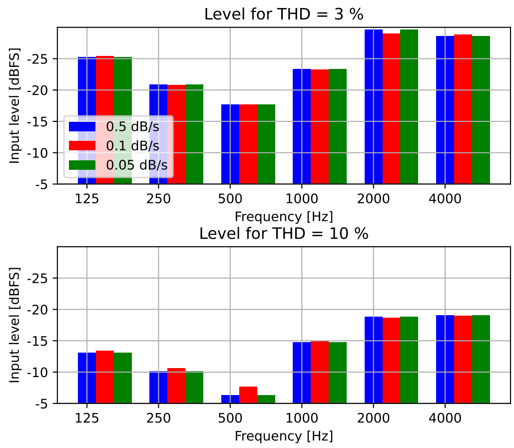



**Figure 14:** Levels for THD of 3% and 10% obtained with sweep level rates of 0.5 dB/s, 0.1 dB/s and 0.05 dB/s.

## 6. CONCLUSIONS AND FURTHER DEVELOPMENTS

In this work the sweep level method (SLM) was studied for characterization of THD and estimation of the linear dynamic range of electroacoustic systems. Compared with stepped level, SLM is much faster, allowing a complete analysis in a wide frequency and level range.

Numerical analysis of the method showed a slight dependence of the results obtained by SLM on the amplitude growth factor used. Also, for high amplitudes and THD, there is a constant difference between THD obtained by SLM and the stepped approach.

The experimental results showed a good match between SLM and stepped level THD. SLM is preferred as it does not load the system to much, reducing power compression in the case of loudspeakers for instance. SLM showed that input levels that corresponds to 3% and 10% THD are highly frequency dependend. Therefore, presenting THD for a single frequency as usually done in products chart can be not representative for the whole operational range.

In future works, we intend to perform an uncertainty analysis of the SLM under a more controlled acoustic conditions using better instrumentation too. Also, we intend to build a data base for electroacoustic devices and work towards some standardization of SLM measurements.

## 7. ACKNOWLEDGMENTS

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