Loudspeaker Distortion – Measurement and Perception
Part 1: Regular distortion defined by design

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ABSTRACT

A major part of the signal distortion generated by loudspeaker systems are directly related with the geometry and properties of the material used in loudspeaker design and found in all good units passing the assembling line. Those regular distortions are the result of an optimisation process giving the best compromise between perceived sound quality, maximal output, cost, weight and size. This paper discusses the physical causes of the regular distortion, their modelling by using lumped and distributed parameters, the objective assessment using modern measurement techniques and the perception by the human ear.

1. Introduction

The target of an audio reproduction system is to generate at the listening position an output signal $p_{out}(t)$ which is similar to the input signal $p_{in}(t)$ at the source point. The difference between the two time signals may be considered as a distortion signal $p_{dist}(t)=p_{out}(t) - p_{in}(t)$ generated somewhere in the audio chain. After introducing digital signal processing, transmission and data storage the weakest part is the electro-acoustical conversion (loudspeaker) and in the interaction with a acoustical environment (room).
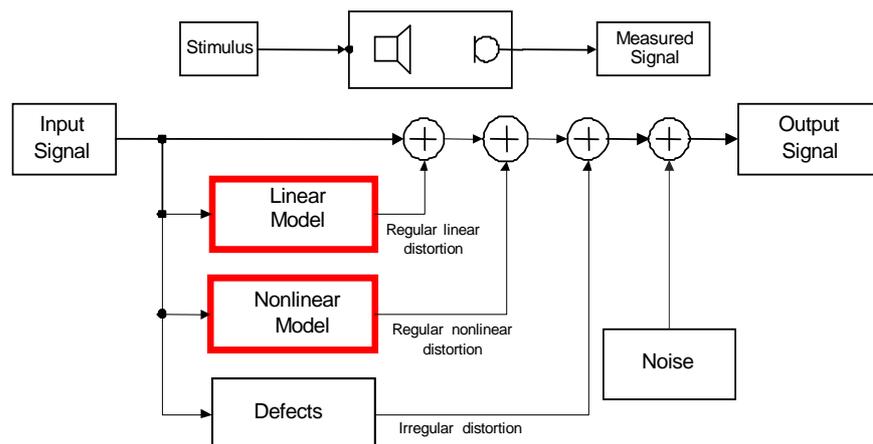

Figure 1: *Signal flow diagram showing the generation of signal distortion in a loudspeaker system.*

The generation of signal distortion can be modelled by a flow chart as shown in Fig. 1. It comprises a linear and a non-linear model, a black box system describing further defects and faults in the system and an independent noise source. The linear and the non-linear model

describe the target performance of the loudspeaker which should be materialized in the golden reference units at the end of loudspeaker development. The outputs of the linear and non-linear models are regarded as regular distortions because they are accepted within the design process and are an result of optimisation process giving the best compromise with other constraints (weight, size, cost, …). Irregular distortion are generated defects caused by the manufacturing process, ageing and other external impacts (overload, climate) during the later life cycle of the product. A rubbing voice coil, buzzing parts, loose particles and air leaks are typical loudspeaker defects which produce irregular distortion which are quit audible and not acceptable. A related paper [12] will discuss the physical causes and measurement techniques in greater detail.
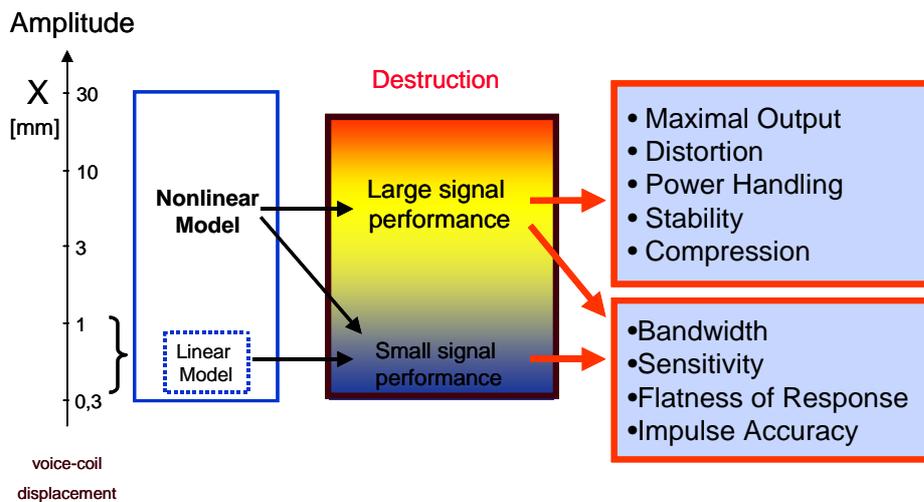

Figure 2: *Prediction of the regular transfer characteristics of loudspeakers by using a linear and nonlinear model*

This paper here focuses on the regular distortion generated by the linear and non-linear models which are theoretical basis of the loudspeaker design process. Linear modelling based on lumped parameter modelling (Thiele/Small parameters) has a long history in loudspeaker design. More complex models using distributed parameters have been introduced to explain the cone vibration and sound radiation at higher frequencies. The linear modelling fails in describing the large signal performance of the loudspeaker which is directly related to maximal output and cost, size and weight of the loudspeaker. Therefore, modelling and direct measurement of loudspeaker nonlinearities is an important part of modern loudspeaker design.

2. Linear Distortion

Table 1 gives an overview on dominant causes of linear distortion caused by transducer and system design and by the acoustical environment in the final application. The first causes are in the one-dimensional signal path close the input of the transducer which can be modeled by a network comprising lumped elements. Electrical measurements of voltage and current at the terminals gives the electrical impedance which is the basis for identifying to basic lumped parameters and other derived Thiele/Small parameters which describe the properties of electrodynamical transducer, mechanical resonator and acoustical load.

Causes of Linear Distortion	Measurements	Characteristics
Fundamental mechanical resonator (coil, cone, suspension)	Voltage and current at terminals of driver operated in free air,	Resonance frequency, loss factor, moving mass, stiffness, compliance
Electrical input impedance (voice coil, iron path, magnet)		dc resistance, inductance
Acoustical load (baffle, enclosure)		Acoustical impedance, box volume, port resonance
Partial mechanical vibration (diaphragm, dust cap, surround) cone, diaphragm	Displacement of radiator's surface	Accumulated acceleration AAL, natural frequencies, modal loss factor, modal functions
Sound Radiation (radiator's surface, horn)	Sound pressure radiated by loudspeaker into anechoic environment	Far field SPL response, polar plot, sound power response directivity index
Diffraction (edges of the enclosure)		
Early reflections (walls, floor, ceiling, panels)	Sound pressure at listening position	Impulse response
Room modes (room size and geometry)		Frequency-time analysis (cumulative decay spectrum)

Table 1: Overview on meaningful measurements for assessing the linear signal distortion generated in loudspeaker systems and identifying their physical causes.

At higher frequencies the radiator (cone or diaphragm) vibrates not as a rigid body anymore but breaks up into higher-order modes. Here a more complex model using distributed parameters and multiple state variables such as the displacement $X(\mathbf{r})$ on sufficient points r on radiator's surface is requried. New mechanical measurements using laser scanning techniques provide the displacement and the geometry of the vibrating surface. The generated sound pressure in the near field or in the far field at the listening position depends not only from the sound radiation but also on the diffraction at the edges of the enclosure, early reflections on room boundaries and room modes. In micro-speakers, headphones, horn compression drivers the acoustical sound field may generate a force $F(\mathbf{r})$ at any point of the vibrating surface which is not negligible and may be also detected in the electrical signals at the terminals.
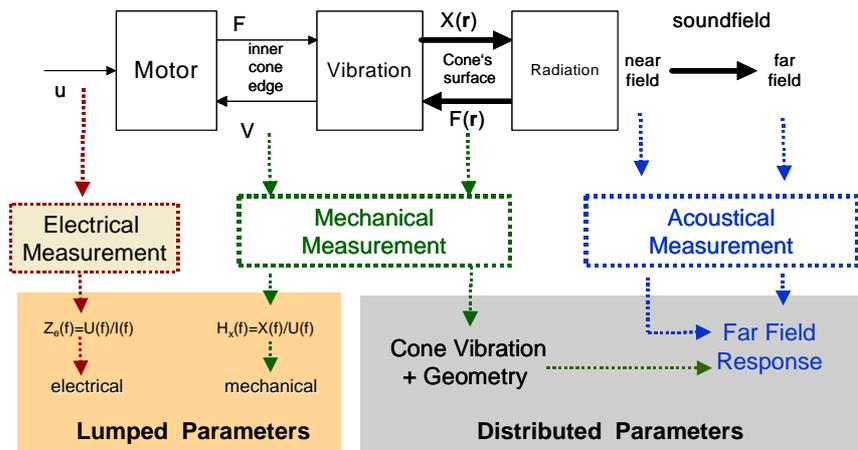
Figure 3: *Modeling the small signal performance of loudspeaker systems by using lumped and distributed parameters*

Traditional loudspeaker design and evaluation of transfer behavior was restricted to electrical and acoustical measurements as shown in Fig. 3. New cost effective laser sensors based on the triangulation principle [1] provide the geometry of the radiating surface at high accuracy and the linear transfer functions between terminal voltage and displacement X(**r**) at sufficient points **r** on the surface. Fig. 4 shows for example the result of such a scanning process collecting mechanical information at about 1000 measurement points. The mechanical scanning process requires no anechoic room and may be applied to the drive unit operated in vacuum.
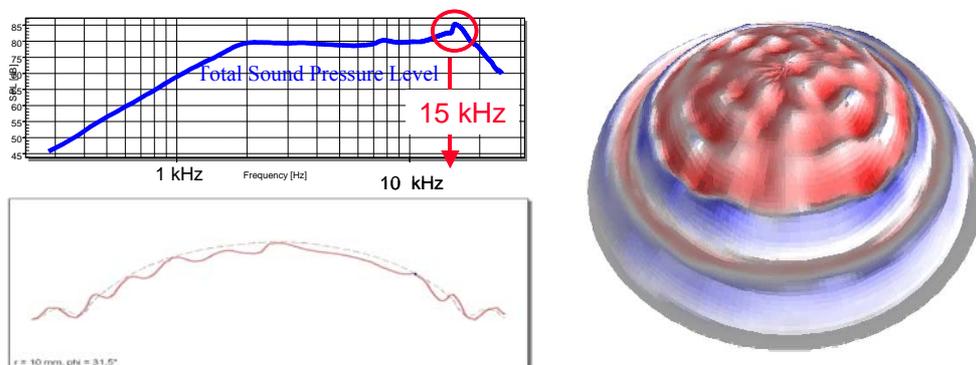

Figure 4: *A critical vibration pattern depicted as a sectional view (left down) and as 3D animation (right) of a soft dome tweeter at 15 kHz causing a peak in the sound pressure on-axis response (upper left)*
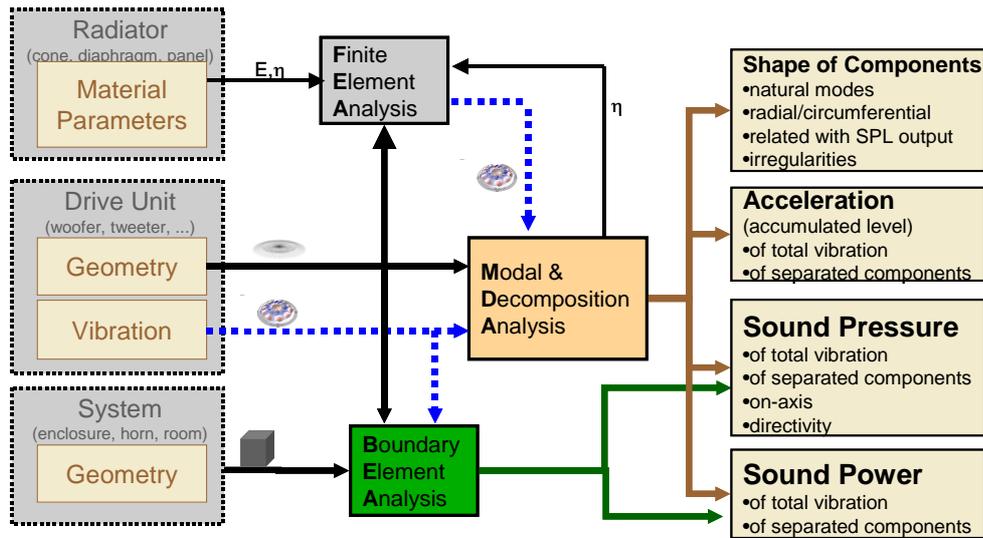
Figure 5: *Vibration and radiation analysis using distributed loudspeaker parameters (geometry and vibration of the radiator's surface) measured by laser scanning techniques*

Numerical calculation based on the scanned data provides the sound pressure on-axis or at any point in the far field giving the polar pattern of the loudspeaker as illustrated in Fig. 5. A new *Sound Pressure Related Decomposition Method* [2] shows how each part of the cone contributes to the sound pressure output in constructive or destructive way. This reveals acoustical cancellations effects, critical rocking modes and undesired circumferential modes. A *Modal Analysis* applied to the mechanical data simplifies the mechanical analysis and provides the modal loss factor η and other material parameters which are important input parameter for a *Finite Element Analysis* to investigate the design choices in greater detail. A *Boundary Element Analysis* may also consider the particular shape of the enclosure, horn or room boundaries to predict the sound field at high accuracy.
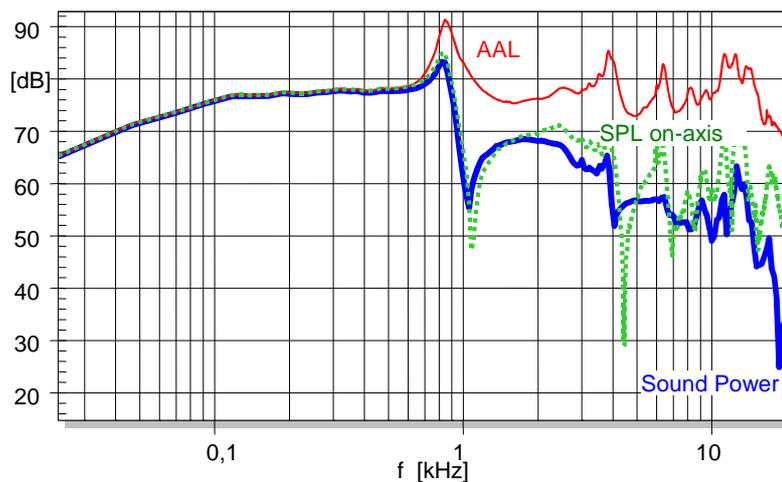

Figure 6: *The most important loudspeaker characteristics in the small signal domain: Accumulated acceleration level (AAL) as thin line describes the mechanical vibration of the radiator's surface and is directly comparable with the on-axis sound pressure level (SPL) as dotted line and the total acoustical sound power response depicted as thick line.*

Three curves calculated from the mechanical scanning data give the most condensed but almost comprehensive description of loudspeaker's small signal performance: The on-axis sound pressure response predicted in 1 m distance in the far field is depicted as a dotted line in Fig. 6. The thick line represent the sound power response of the loudspeaker and the thin line on the top shows with the accumulated acceleration level (AAL). The AAL corresponds with the total mechanical energy neglecting the phase information but normalized in such a way to be comparable with the acoustical output. It may be interpreted as the maximal acoustical sound pressure level while neglecting any acoustical cancellation. Therefore the AAL and SPL curves are identical at low frequencies (in Fig. 6 up to 800 Hz) where the loudspeaker cone vibrates in the rigid body mode and all points on the cone contribute to the sound pressure output constructively. However, at distinct frequencies such as 1.1, 4.4 and 7 kHz there are significant dips in the SPL output which are not found in the AAL. The difference between AAL and SPL curve describes the acoustical cancellation effect quantitatively. The AAL response comprises characteristic peaks which occur at the natural frequencies of the higher-order modes. The 3dB bandwidth of each "resonance peak" corresponds with the modal loss factor of the material used. At low frequencies the sound power response is most identical with both AAL and SPL responses because the loudspeaker dimensions are small compared to the wavelength and the radiator behave as an omni-directional source.

3. Regular Nonlinear Distortion

Table 2 gives an overview on the physical causes of regular nonlinear distortion affecting the loudspeaker's large signal performance [3]. The dominant nonlinearities are in the motor and suspension part of the electro-dynamical transducer because the voice coil displacement is relatively large compared to the dimensions of the coil-gap configuration and size of the corrugation rolls in the suspension (spider, surround). In micro-speakers, headphones and compression drivers the air flow in the gap may generate a nonlinear dependency of the mechanical resistance $R_{ms}(v)$ on velocity v. In vented-box loudspeaker systems there is a similar mechanism causing a nonlinear flow resistance $R_{ap}(v_p)$. High local displacement at the surround and a particular regions on the cone activate nonlinearities in the modal vibration. A typical nonlinearity related to the sound radiation is the Doppler Effect where the high excursion of the bass signal changes the position of the cone and causing variation in the propagation time affecting high frequency components radiated from the radiator at the same time. In horn compression drivers the high sound pressure causes a gradual steeping of the waveform while the sound wave is traveling from the throat to the mouth of the horn.

Causes of Nonlinear Distortion	Measurements	Characteristics
Nonlinear force factor Bl(x) and inductance $L_e(x)$, $L_e(i)$ of motor assembly (voice coil, iron path, magnet)	Voltage and current at loudspeaker terminals, sound pressure in the near field of the driver	Nonlinear parameters and large signal parameters (e.g. voice coil offset)
Nonlinear stiffness $K_{ms}(x)$ of mechanical suspension (surround and Spider)		Nonlinear symptoms for particular stimuli
Nonlinear losses $R_{ms}(v)$ of mechanical and acoustical system		IMD, X_{DC}, MTD, THD, Compression
Nonlinear flow resistance $R_{ap}(v_p)$ of the air in the port of a vented system	Sound pressure inside the vented enclosure	Compression of fundamental component at port resonance
Partial vibration of the radiator's surface	Sound pressure in near	THD, IMD, MTD

(surround, cone, diaphragm, dust cap)	or far field	
Doppler effect	Sound pressure in far field	IMD, MTD
Nonlinear sound propagation (wave steepening) in horns		IMD, THD, MTD

Table 2: *Overview on meaningful measurements for assessing the regular nonlinear signal distortion generated in loudspeaker systems and identifying their physical causes.*

The effect of the dominant nonlinearities can be investigated by the lumped parameter model shown in Fig. 7. Contrary to a linear model some elements have not a constant parameter but depend via a nonlinear function on voice coil displacement x, velocity v, current i, sound pressure in box enclosure p_{box} or other state variables.
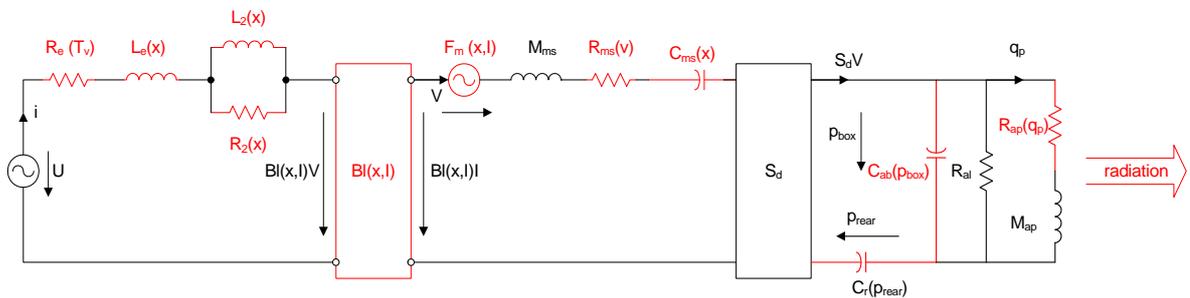

Figure 7: *Lumped parameter model of a vented-box loudspeaker system considering the dominant nonlinearities in the electrical, mechanical and acoustical domain.*

The shape of the nonlinear parameter characteristics are directly related to the geometry and properties of the material. Fig. 8 shows the nonlinear stiffness $K_{ms}(x)$ of the total suspension as the solid thick curve in the right diagram increasing at positive and negative displacements. This is very typical for any spider and surround when the shape of the corrugation rolls is deformed at high excursions.
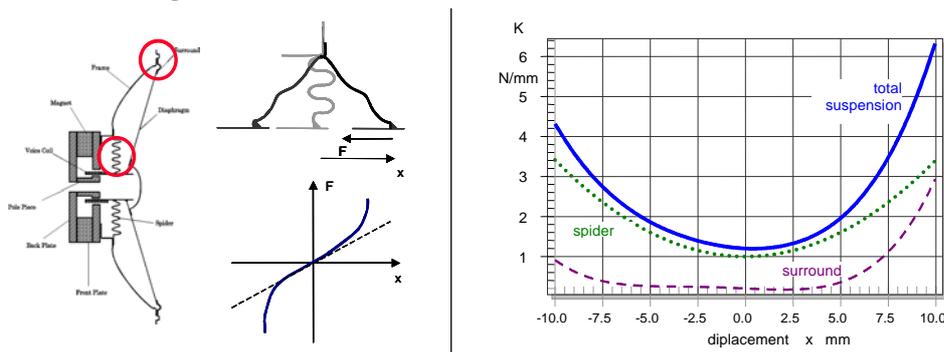

Figure 8: *Nonlinear stiffness characteristic K(x) versus displacement x of the mechanical suspension (surround and spider) dynamically measured by modern system identification using the electrical signals at loudspeaker terminals.*

The solid curve in Fig 8 also reveals an asymmetry in the stiffness characteristic which is caused by the asymmetrical shape of the surround which is more stiff and less compliant for positive than negative excursion. This asymmetry is an undesired property which causes not only 2nd- and higher-order distortion but generates a dc displacement moving the coil to the

softer side of the suspension. Nonlinearities may also cause an instability of the motor at frequencies above resonance.

The large signal performance is predictable and there is close relationship via the nonlinear parameters to the design.
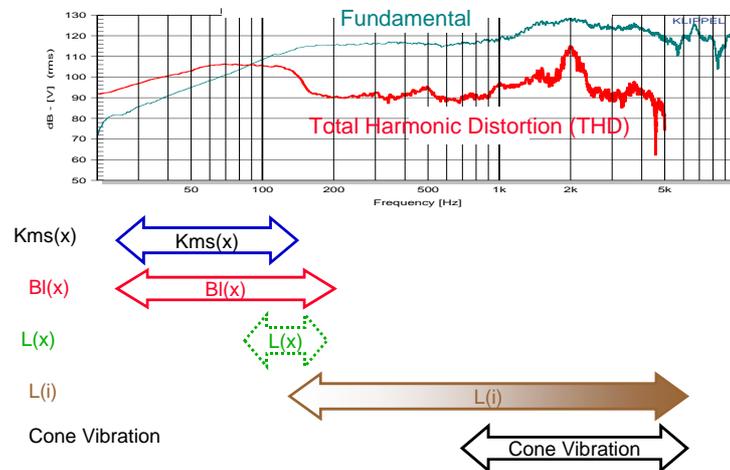

Figure 9: *Relationship between the dominant loudspeaker nonlinearities (causes) and the total harmonic distortion (nonlinear symptom) generated by a single-tone swept continuously versus frequency*

The generation of nonlinear distortion and other symptoms depends on the properties of the stimulus. A single tone generates new spectral components at multiples of the fundamental frequency which can easily be measured by conventional harmonic distortion measurements. Fig. 9 shows the response of the total harmonic distortion (THD) and relationship to the physical causes. The high level of the harmonic distortion below 150 Hz is caused by voice coil displacement x activating the stiffness $K_{ms}(x)$ or force factor nonlinearity $Bl(x)$. The displacement varying inductance $L(x)$ can only generate low values of THD in a narrow frequency range just above resonance (150-200 Hz). The inductance nonlinearity $L(i)$ varying with current i may contribute to the THD at higher frequencies. The distinct peak in THD at 2 kHz is caused by a nonlinear vibration of the cone and surround after break-up.
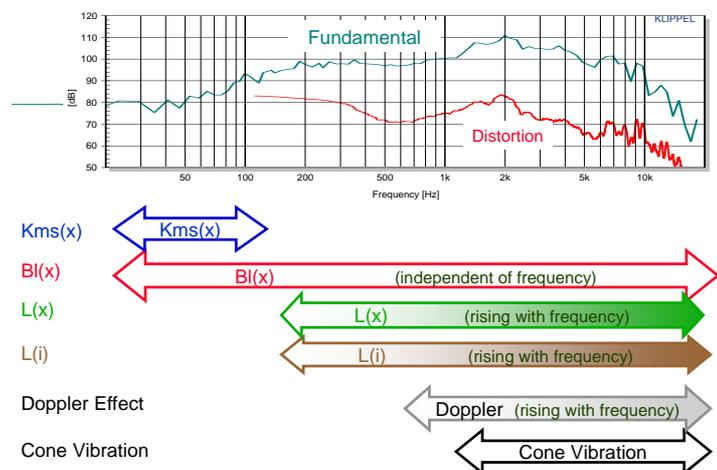
Figure 10: *Relationship between the dominant loudspeaker nonlinearities (causes) and the nonlinear distortion generated by a sparse multi-tone stimulus*

Unfortunately, harmonic distortion measurement gives not a comprehensive picture of the large signal performance of loudspeaker systems. At least a second tone is required to generate intermodulation products which occur at difference and sum frequencies in all possible combinations of the excitation frequencies. Increasing the number of fundamental components in multi-tone stimulus will generate more and more intermodulation components spreading over the complete audio band. Contrary to the THD response in Fig. 9 the nonlinear force factor $Bl(x)$ and the inductance $L(x)$ THD generates significant intermodulation distortion at higher frequencies as illustrated in Fig. 10. Thus, harmonic distortion measurements using a single test tone are not sufficient for assessing loudspeakers comprehensively and predicting the large signal performance for complex stimuli like music.

4. Impact on Perceived Sound Quality

The reproduced sound quality as perceived by a listener is one of most important criterion for the preference of an audio product. Systematic subjective evaluation requires a double-blind test strategy and psychometrical tools for assessing the sensations reliably and quantitatively. Such tests are time-consuming and expensive and the results depend on the particular listening condition (room, program material) and the training of the listeners.

Thus it is desirable to predict those subjective sensations based on objective measurements and perceptive modeling considering the interactions between stimulus, loudspeaker, room, ear and listener's training and expectations. There are two alternative approaches using different sources. One is based on personal listening experience, vague speculations or even myths. This reflects the heritage of accumulated knowledge which is difficult or impossible to verify by science. Exploiting this expertise is beneficial as long as it is combined with a critical attitude and some common sense. Wrong conceptions will die eventually and the falsification of those ideas are interesting research topics which accelerate this clarification.

The other approach is based on facts accumulated by psycho-acoustical research modeling the basic processing in the ear. Unfortunately, there are still many open questions how to apply the results of those fundamental experiments to sound reproduction of natural audio signals.
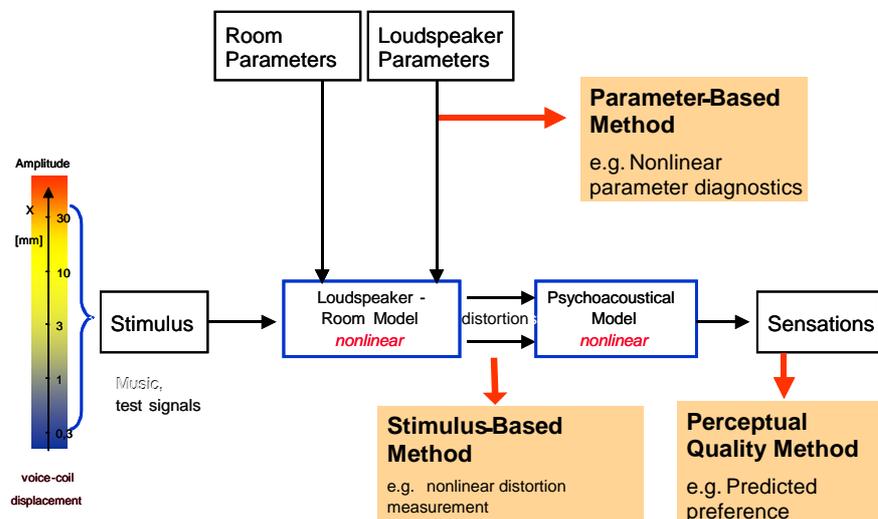
Figure 11: *Objective methods for assessing the sound quality of loudspeaker systems.*

Fig. 11 gives an overview on the current objective methods on assessing the sound quality of loudspeaker systems. The *parameter-based method* relies on loudspeaker characteristics such as lumped and distributed parameters which are independent of the stimulus. The interpretation of harmonic distortion and other nonlinear distortion belongs to the *stimulus-based method* which considers the properties of a particular stimulus, position of the listening position and the influence of the acoustical environment. The linear and nonlinear distortions separated from undistorted stimulus are the input of the following psycho-acoustical model considering generating basic perceptual attributes (loudness, sharpness, roughness) and overall judgments describing the pleasantness of the sound and preference considering the ideal conceptions of the listener [10].

The psycho-acoustical model performs a binaural nonlinear processing where a significant part of the distortion component is masked by other signal components. Below the main mechanisms are summarized which reduce the audibility of signal distortion:

- Spectral components within third-octave bandwidth contribute to the same excitation level above 400 Hz → smoothing of the amplitude frequency response, the shape of a resonance (gain, Q factor) has a minor influence on audibility as long as the excitation within the critical band is constant [6],

- Spectral components below 100 Hz contribute to the excitation level of one critical band → sufficient bass sensation can be generated by higher frequencies (60 …100 Hz) when the very low frequency components (20 …40 Hz) are attenuated by the cut-off frequency of the loudspeaker,

- 1 dB variation in the excitation level within a critical band becomes audible,

- Spectral masking excites adjacent bands → dips in the frequency response are less audible than peaks, nonlinear distortion components are masked by fundamental components [7],

- Temporal masking → the rms-value (rather than the peak value) determines the audibility of the regular nonlinear distortion,

- Hearing threshold → bass components are not audible if the listening level is too low, a small difference in the level of low frequency components may cause a significant difference in perceived bass sensation,

- Monaural processing is not very sensitive for phase shift of signal components processed in separate critical bands → Phase distortion corresponding with a group delay variation of 0.4 …2 ms within a critical band change the timbre and roughness of the sound,

- Binaural processing [5] is sensitive for interaural level differences (1…2 dB) and time delay (50 μs) → latency and group delay response should be identical in the symmetrical channels of a multi-way system to avoid lateralization of the perceived sound image,

- Precedence effect [5] maintains the primary image as long as the lateral reflections are sufficient low or the time delay is small, → strong reflections after 80 ms are unpleasant and are perceived as echo,

- Audible lateral reflections may generated a preferred sensation of spaciousness and a broadening of the primary image [13], the optimal delay and level depend on the properties audio signal (20 ms delay for speech or 40 ms for music and reflections having the same level as the direct sound) → early reflections as found in relatively small rooms improve sound quality, artificial generation of lateral reflections may be desired in an anechoic environment or small rooms (cars),

- Adaptation [6] to the acoustical environment causes a variation of the ideal conceptions versus time → the listener becomes less sensitive to linear distortion caused by room and loudspeaker after some time,

- Intermodulation distortion [4] are detected by the ear not only by exploiting spectral but also temporal clues, amplitude modulation is much more audible than frequency modulation and is perceived as fluctuation (modulating bass tone f_l < 20 Hz) or roughness at higher frequencies (20 Hz < f_b < 400 Hz) or separated spectral components (> 400 Hz) → low amplitude intemodulation distortion at 1 … 3 % caused by nonlinear force factor Bl(x) and inductance L(x) is detected as an unnatural roughness.

4.1.Auralization Techniques

Although the perceptive modeling gives valuable insight into fundamental psycho-acoustical mechanisms and basic sound attributes, it is at the current state not very accurate in the predicting of the overall assessment of the perceived sound quality and in the preference of an audio product. The ideal conceptions of a listener highly depends on training, listening habits, fashion, cultural factors and artistic properties of the program material. Some linear and nonlinear distortion are clearly audible but may be acceptable for a particular application and program material (popular music) or may even be perceived as interesting effect (artificial bass enhancement). The reliable evaluation of those criteria requires systematic listening tests using modern auralization techniques [8, 9].
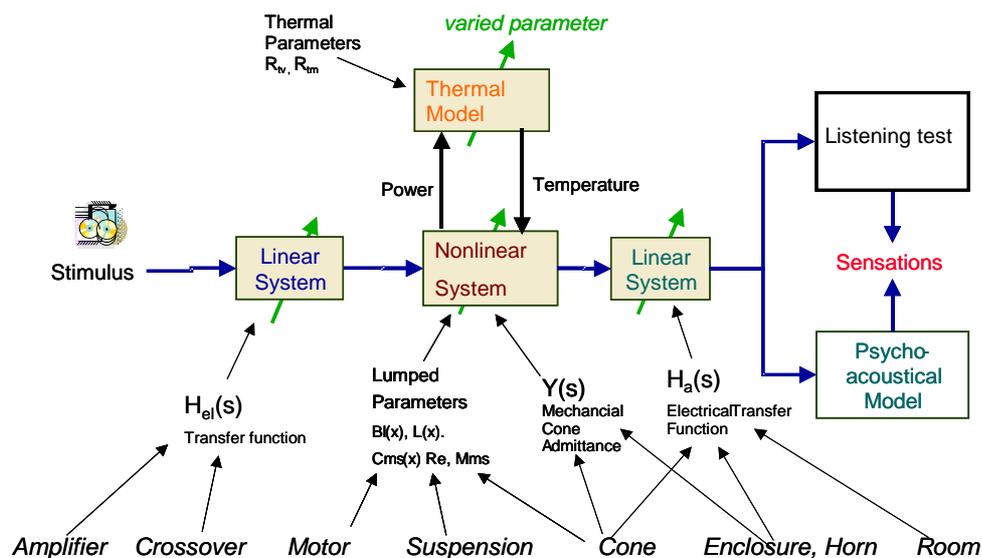.

Figure 12: *Simulation and auralization of loudspeaker distortion in reproduced audio signal based on linear and nonlinear modeling and using natural audio signals (music, speech) or artificial test signals.*

Fig. 12 shows a digital signal processing system based on loudspeaker modeling to generate a virtual audio system. This model has a sandwich structure where a nonlinear system modeling the dominant nonlinearities in the electrodynamical transducer is embedded by linear systems. The first linear system corresponds with the electrical signal path from the source to the loudspeaker terminals while the second linear systems models the signal path in the mechanical and acoustical domain where the amplitude is relatively small and the sound propagation is sufficiently linear. This technique is a convenient tool for investigating design choices before a first prototype is made and combines subjective and objective evaluation.

5. Conclusions

Linear and nonlinear distortion are unavoidable in current electro-acoustical transducers using a moving coil assembly driving diaphragms, cones and other radiators. The regular distortion are deterministic and can be predicted by using linear and nonlinear models and identified loudspeaker parameters in an early design stage. Finding acceptable limits for those regular distortion is an important part in defining the target performance at the begin of loudspeaker development. Subjective evaluation is required to assess the audibility and the impact on perceived sound quality. Some distortion which are audible might be still acceptable or even desirable in some applications. Systematic listening test, nonlinear auralization and objective assessment based on an perceptual model are useful tools to assess regular distortion.

6. References

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