Non-linear Digital Audio Processor for dedicated loudspeaker systems

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In this paper we describe a digital processor, which operates the audio signal to compensate loudspeaker non-linearities.
A DSP is used to implement a two-dimensional FIR, in which amplitude dependent SPL measurements are reverted and mapped.

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Summary

In this paper we describe a digital processor, which operates the audio signal in order to compensate loudspeaker non-linearities. A DSP is used to implement a two-dimensional Finite Impulse Response filter (FIR), which select the proper elaboration from the current amplitude of the input signal. For each amplitude the filter is choosen by inverting measurements of the Sound Pressure Level (SPL) of a dedicated loudspeaker system. Results obtained measuring a commercial woofer are reported.

The proposed audio processor is coupled with the loudspeaker and achieves a remarkable reduction of distortion, by adjusting the audio signal so as to compensate for the non-idealities in the transducer.

Voice coil loudspeakers, the most used audio transduction devices, produce an acoustic field in the air as a result of the action of an electric signal. The current driven by the audio amplifier flows in the coil, moving in constant magnetic field, and produces an axial force on the cone, which is tied to the coil (א)

In an ideal system the relationship between input (electric variables), and output (acoustic variables) would be a linear one. Unfortunately, the magnetic field is not constant as the cone is displaced from its equilibrium position and the reaction force arising from the cone suspension is hysteretic with the displacement. Moreover other factor affect significantly the linearity of the transduction action, among them noteworthy are the increase of the coil temperature which produces a variation in its resistance; the variation of the coil inductance changes with the coil displacement; the non perfect stiffness of the paper walls of the cone at high acoustic frequencies, which induce a mechanical phenomenon usually called ``break-up'' arises, whereby some cone regions move out of phase with respect to the other cone regions.

All these effects result in a deterioration of the emitted sound that can become even intolerable. In summary, the input signal does not generate a strictly proportional acoustic signal. Therefore every effort is spent to make the transduction chain as much linear as possible, in terms of amplitude and phase. Since the desired linearity cannot be achieved only by proper choice of the raw materials and of the manufacturing process, a supplementary action must be undertaken. Suitable pre-elaboration of the input signals seems to be the easiest and most promising approach. An audio signal processor can be inserted before the power amplifier and the loudspeaker, with the purpose of adjusting the signal so that the overall transducer chain is linear. For simplicity in the following considerations the power amplifier will be considered strictly linear.

The presence of the audio processor ensures maximum reliability, highest quality in audio performance and high power conversion efficiency. Reliability, as the processor controls the current flowing in the coil, avoiding overdrive conditions and the resulting damage, due to mechanical and thermal stress. Sound quality is improved, since it is possible to equalize the system and to compensate non-linearities. Efficiency is achieved, as passive filters between the power amplifier and the loudspeaker are no longer necessary, and a better power matching can be ensured.

The audio processor proposed in this work operates by inverting the SPL of the considered loudspeaker and mapping it into a Digital Signal Processor (DSP), so that the overall transduction chain DSP and loudspeaker result in a linear transfer function in the effective bandwidth.

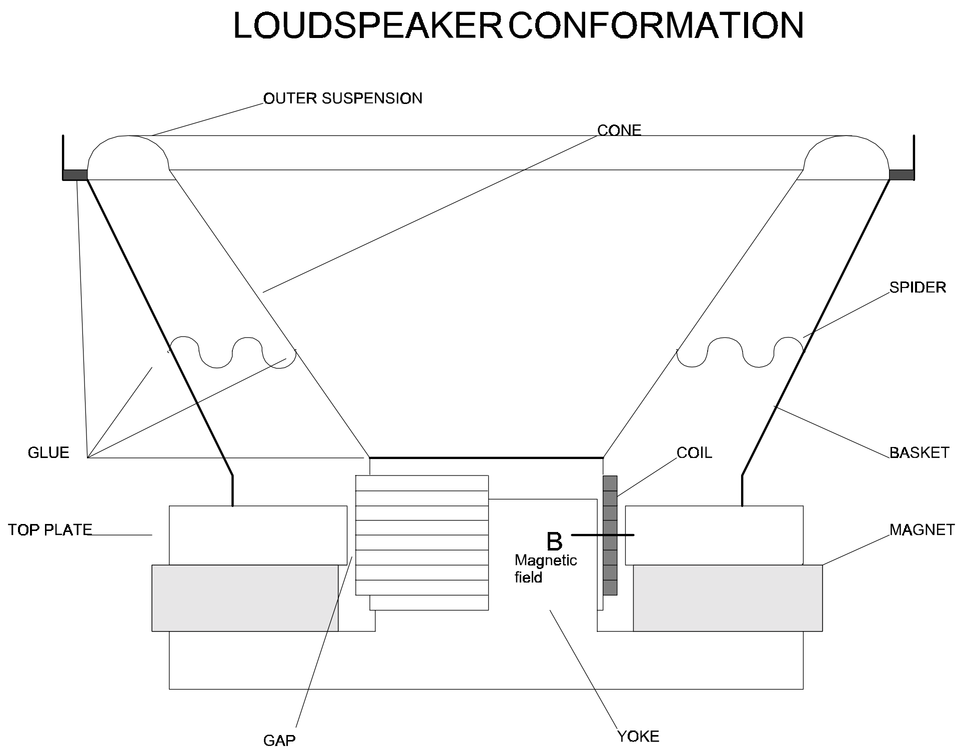

Figure 1 Basic loudspeaker conformation

Digital Pre-elaboration of loudspeaker system

Preliminary audio processor implementations aiming at a reduction of distortion are already described in literature for both closed-loop control approach [Black_1977], [Adams_1983] and open loop approach [Klippel_1992b],[Klippel_1996],[Bellini_1998]. In [Klippel_1996] closed loop simulations and measurements are used only to tune design parameters, while in [Bellini_1998] an analog fuzzy processor is used to pre-elaborate the driving signal. Open loop control approaches are possible only thanks to the availability of accurate loudspeaker models. The field is extensively studied in literature [Kaizer_1987], [Birt_1990], [Klippel_1990], [Klippel_1992a], but to the best of our knowledge, the available models rely on measurement results rather than on first principles and construction parameters. Until now loudspeakers are modeled, following the well-known Small-Thiele approach [Thiele_1971], [Small_1972].

In this paper an indirect modeling approach is considered. In fact even the designed audio processor relies on several SPL measurements of the considered loudspeaker system, which were performed using different amplitude for the driving signal. Since the transducer is a non-linear one, they are different in shape and in level. Therefore in the DSP a two-dimensional non-linear filter is mapped, where the samples are operated by a FIR whose taps are synthesized by inverting the measured SPL, corresponding to the current amplitude of the driving signal. Moreover the non linearity of the magnetic field is considered, by increasing the driving signal, where the Total Harmonic Distortion (THD) is high. From these considerations it stems that good performances are obtained especially at low frequencies.

It should be considered that, for a product line including several different loudspeakers, it will be necessary to develop an audio processor family, each member being dedicated to a specific loudspeaker model. In this context, reconfigurability and accuracy are key features of the design. The digital implementation allows to achieve reconfigurability and accuracy. An automatic design flow was developed which produces, starting from measurements, the tap of the FIR to be employed and finally the assembly code to map the desired filter into the DSP. It allows a fast prototyping action for the linearization of different kind of loudspeaker, since the manufacturer only needs to perform standard SPL and impedance measurements on his products, and then use them as inputs for his design flow. Feedback with measurements of the final chain (audio processor and loudspeaker) was used to tune the best-matching anti-distortion filters.

For the experiments a commercial double coil sub-woofer with a voice coil diameter of 25 mm was used. It is normally used feeding the left and the right audio channels, respectively, into the two coils. It has an operating frequency range of [20Hz, 2kHz], and a resonance frequency of 55Hz ±9Hz. Preliminary results demonstrate an improvement in audio reproduction quality both from the objective and the subjective point of view. Listening tests were performed to validate subjective audio quality. Measurements of the THD using a TMS320C54xx Development Starter Kit by Texas Instrument were performed in an anechoic room, with a commercial double-coil sub-woofer, stimulated by a pure sinusoidal tone in the operating frequency range [20Hz, 2kHz]. They demonstrate a remarkable linearization of the reproduced SPL, and a reduction of THD at low frequencies, ב.

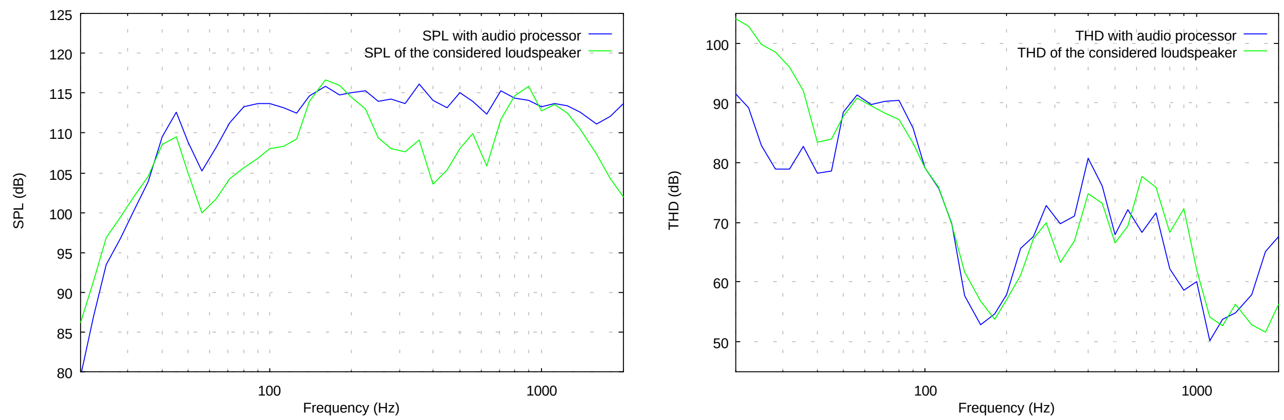

Figure 2 Comparison between the SPL (a) and the THD (b) of the considered loudspeaker with and without the audio processor.

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