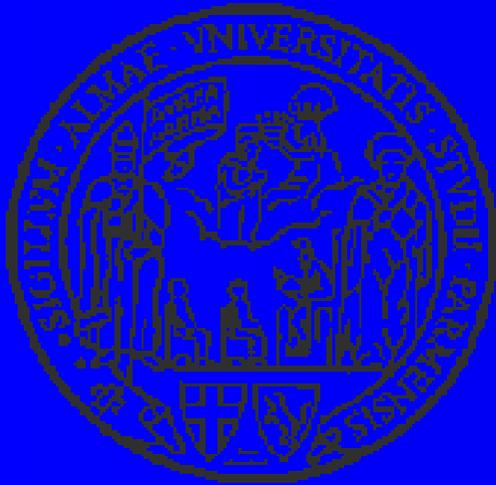


Inverse numerical filters for linearisation of loudspeaker's response

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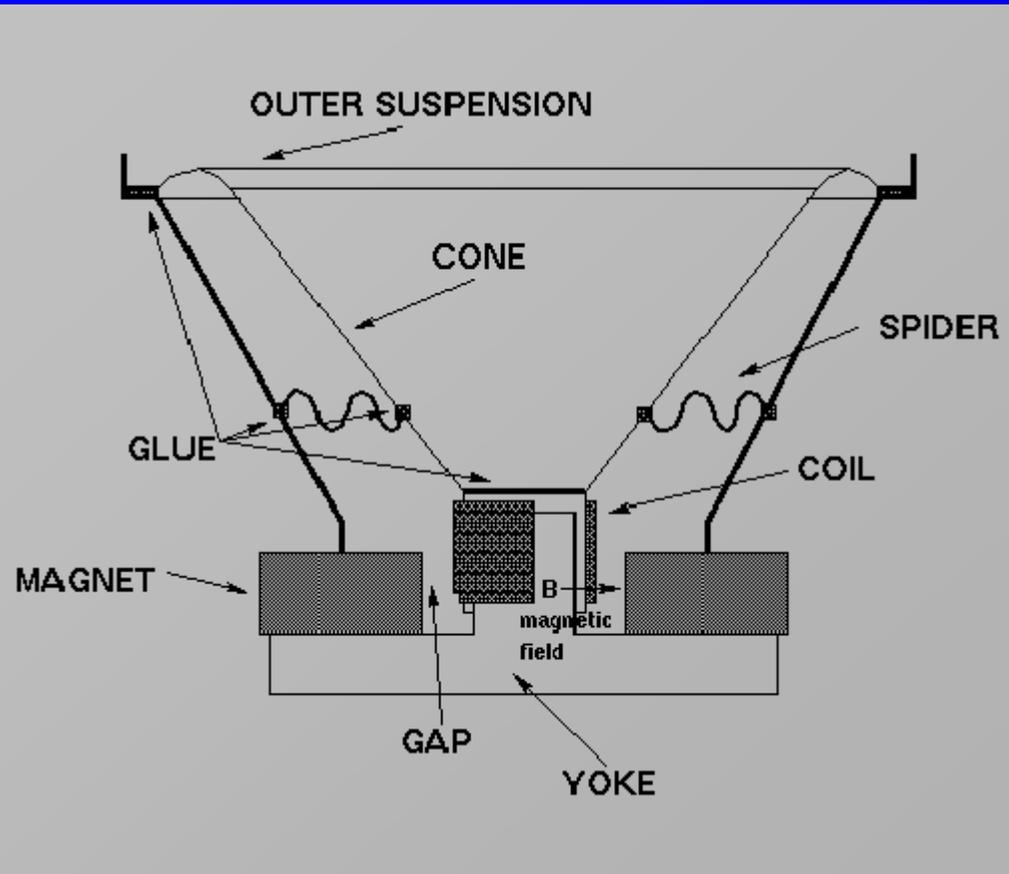




Outline

- Loudspeaker physical conformation;
- Loudspeaker non-linear modeling;
- Digital audio Processor for distortion compensation;
- DSP implementation;
- Measured distortion results;

Basic loudspeaker conformation

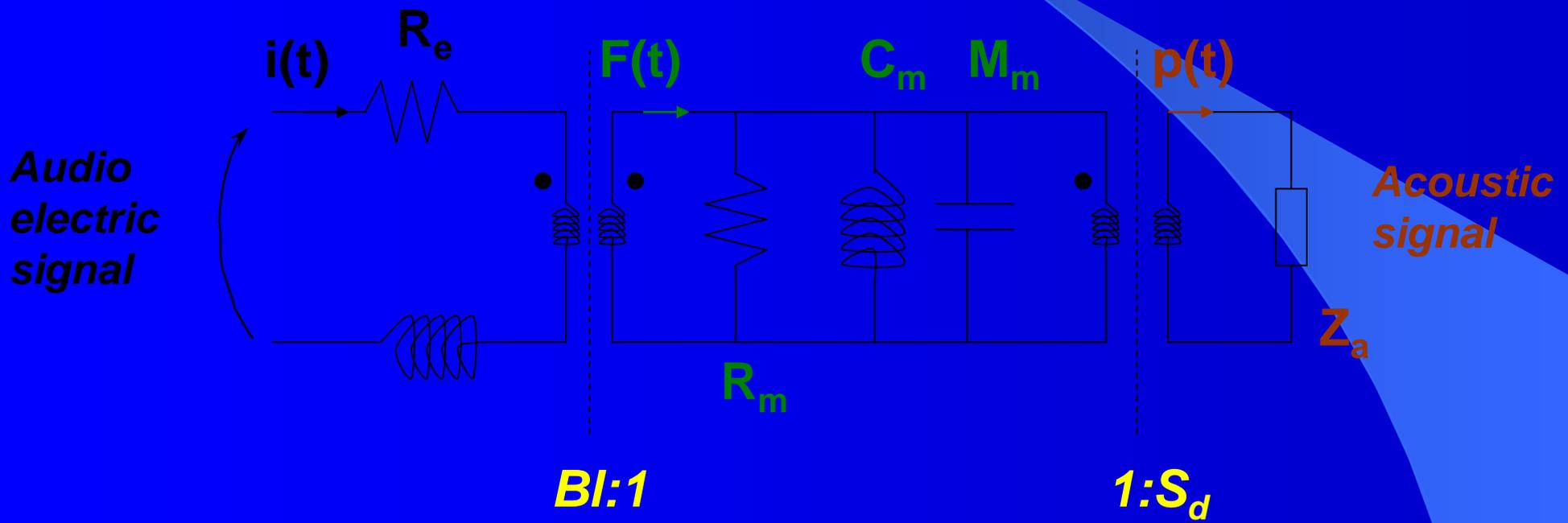


- Electric- mechano-acoustic transducers:
 - $p(t) = k e(t)$
- Non-linear behavior:
 - The magnetic induction B is not constant with displacement
 - non ideal suspension stiffness
 - $R = R(T)$

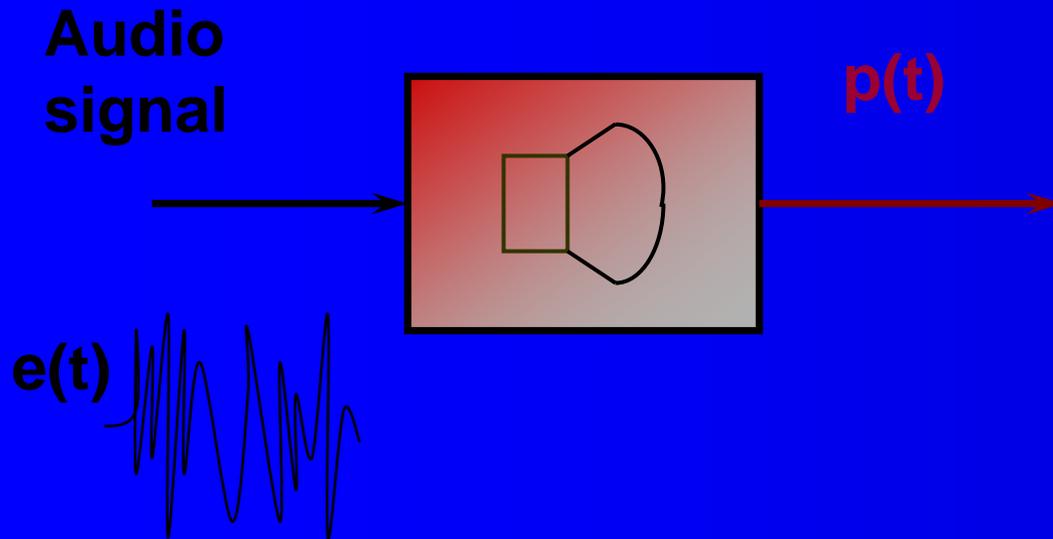


Angelo Farina

Loudspeaker modeling



Loudspeaker modeling

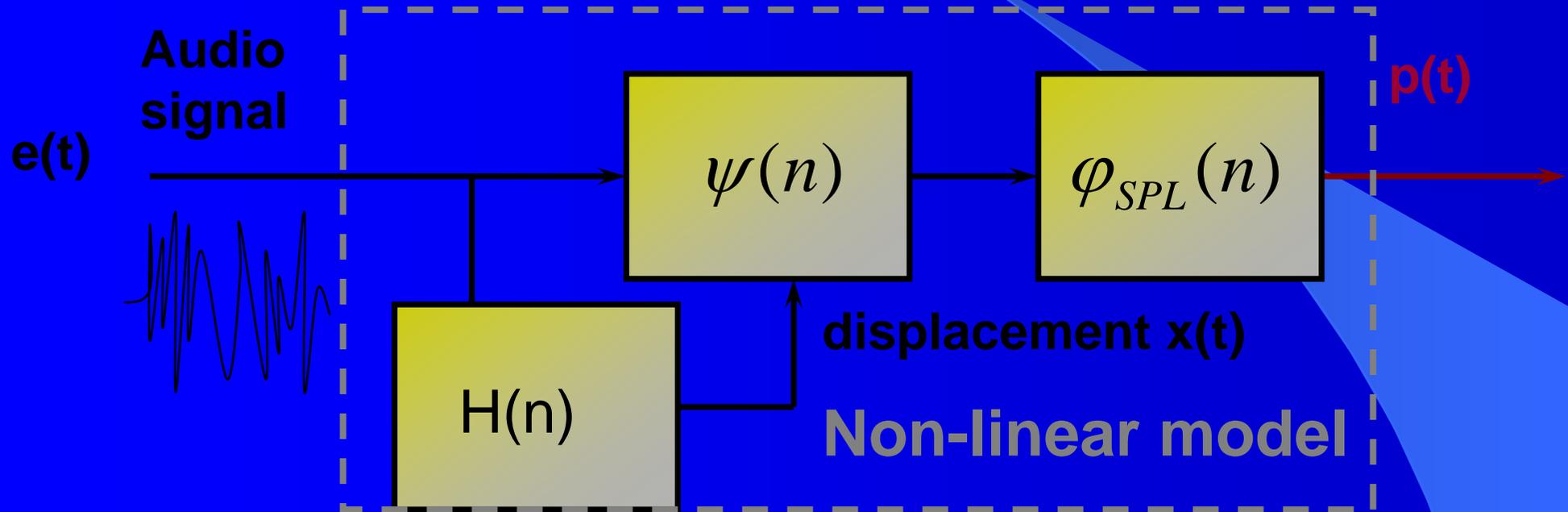


- Non-linear loudspeaker I/O relation is linearized around fixed displacements x_n
- ψ_{x_n} accounts only for non-linear behavior

$$\varphi_{\hat{x}_n}(s) = \frac{p(s)}{e(s)}$$

$$\psi_{\hat{x}_n}(s) = \frac{\varphi_{\hat{x}_n}(s)}{\varphi_0(s)}$$

Loudspeaker non-linear modeling



- $\varphi_0(s) = \frac{p(s)}{e(s)} \rightarrow \varphi_{SPL}(n)$ is similar to SPL measurements



Mathematical formulation

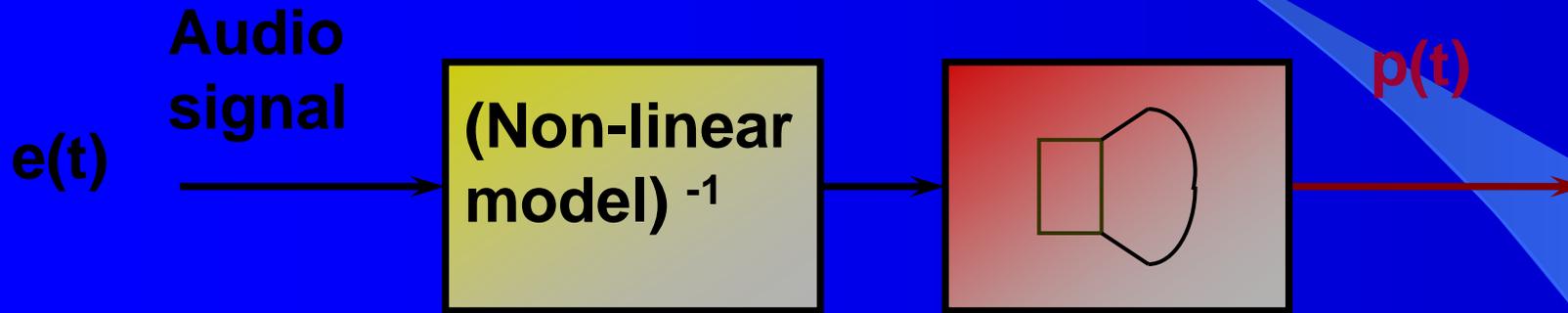
A set of displacement-dependent transfer functions is derived from the Thiele-Small formulation, allowing for the dependence of B_l , L and K from the displacement x .

$$\varphi_{\hat{x}_n}(s) = \frac{\tilde{z}_a(s)\tilde{S}_r(s)sB_l(\hat{x}_n)}{R_e + sL(\hat{x}_n)} \left[K(\hat{x}_n) + \frac{B_l^2(\hat{x}_n)s}{R + sL(\hat{x}_n)} + \tilde{J}(s) \right]^{-1}$$

n different values of x are considered, and around each of them the variation of the parameters is considered linear. A separate inverse filter is computed for each of the n not-linear responses, and each sample is convolved with the proper inverse filter, which is chosen by first computing the estimated instantaneous displacement x .



Audio processor

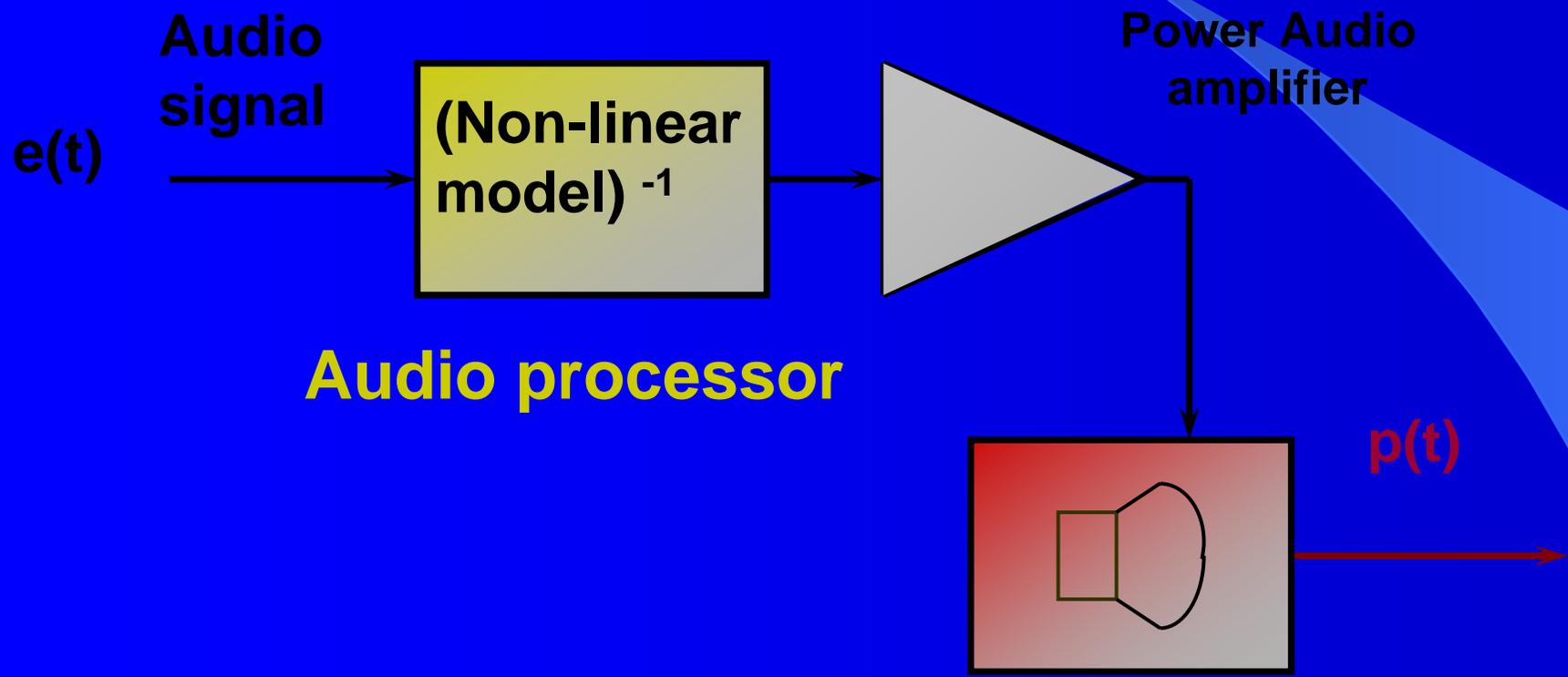


Audio processor

- The more critical part is the synthesis of the $H(n)$ filter, which makes it possible to estimate the instantaneous displacement



DSP implementation



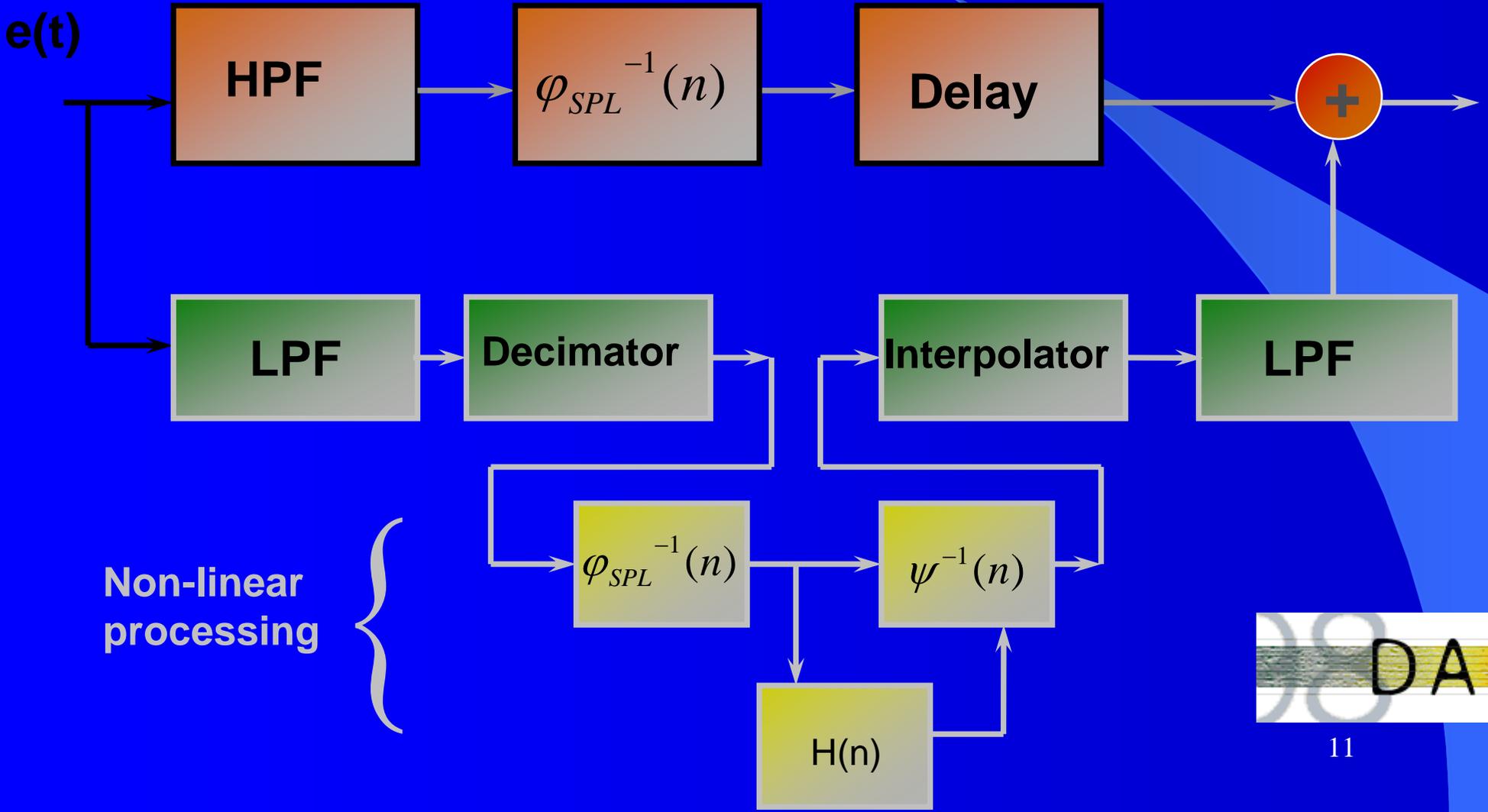


DSP implementation

- Audio processor implemented with a 320C54x DSK;
- 40 MIPS, 10Kword Dual Access RAM;
- Sampling frequency 23148 Hz;
- Converter resolution 14 bit.

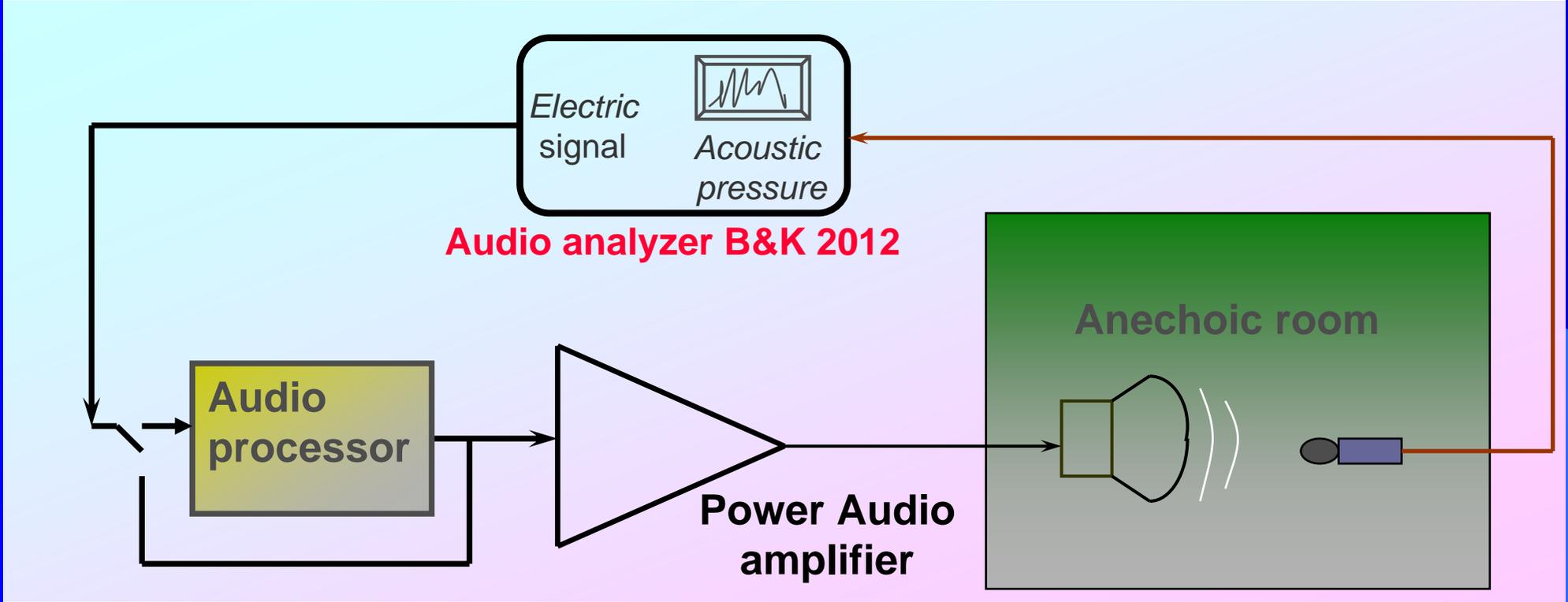


Multirate Audio processor



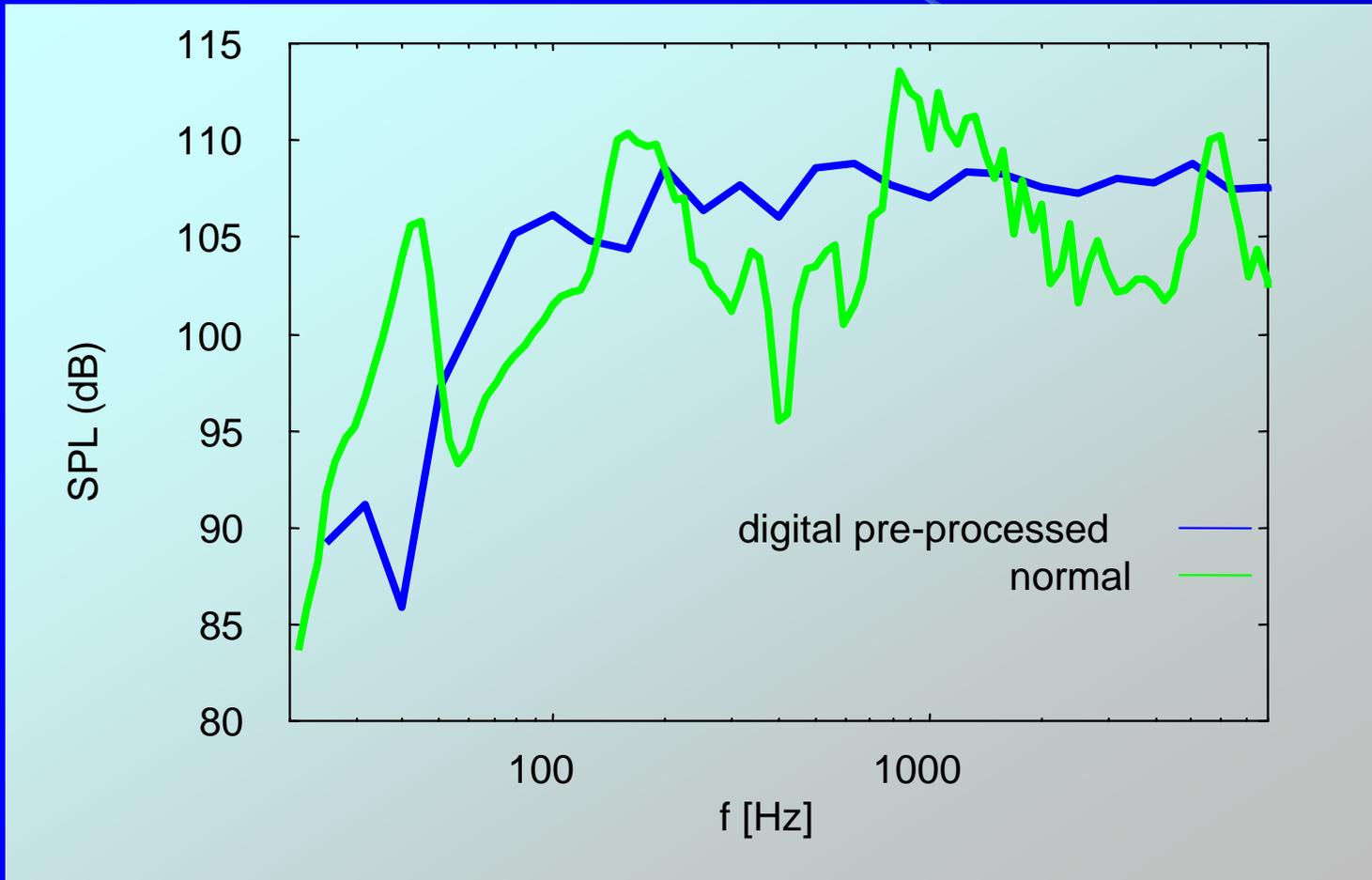


Audio processor measurements



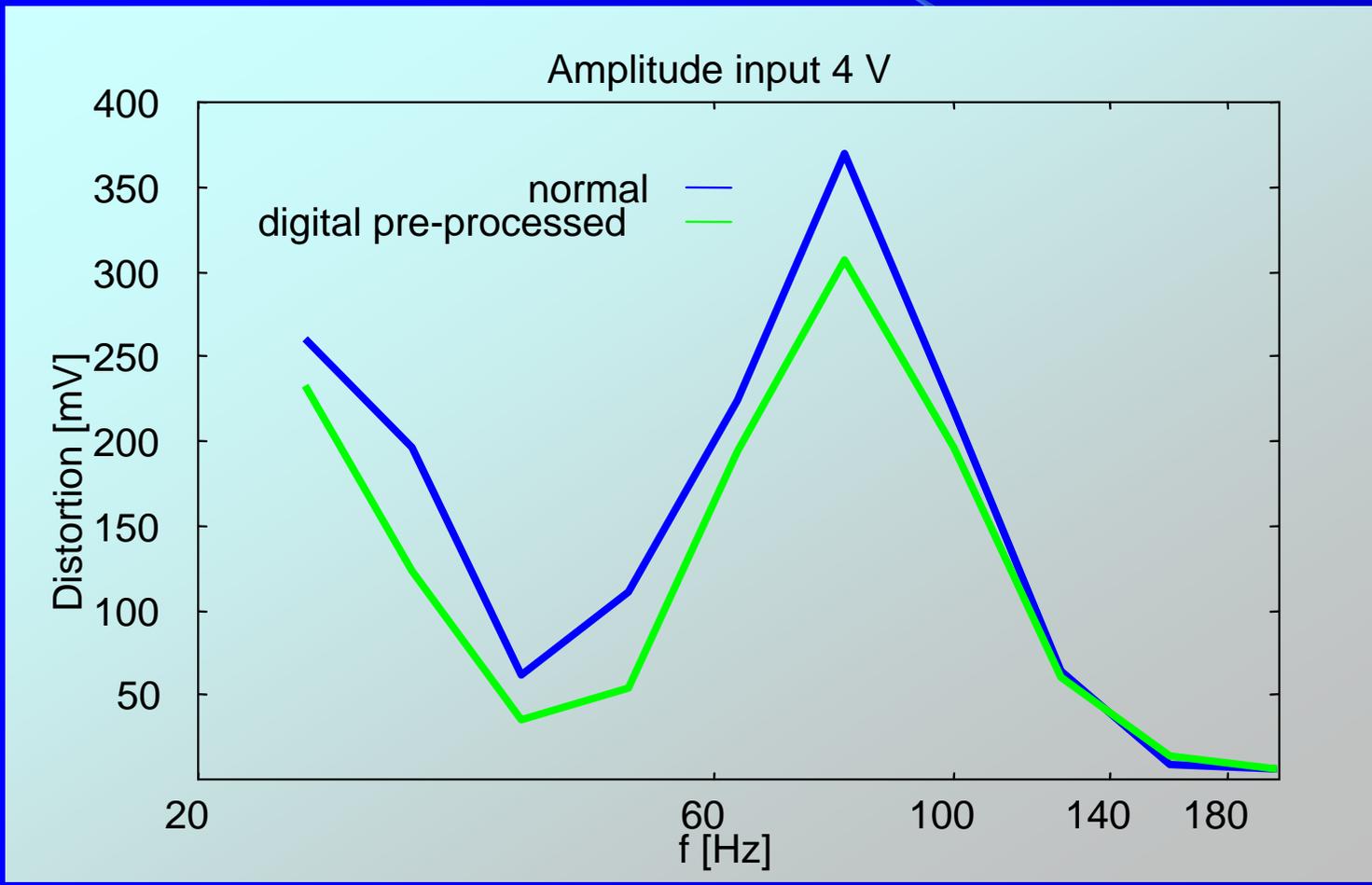


Response (SPL) Results



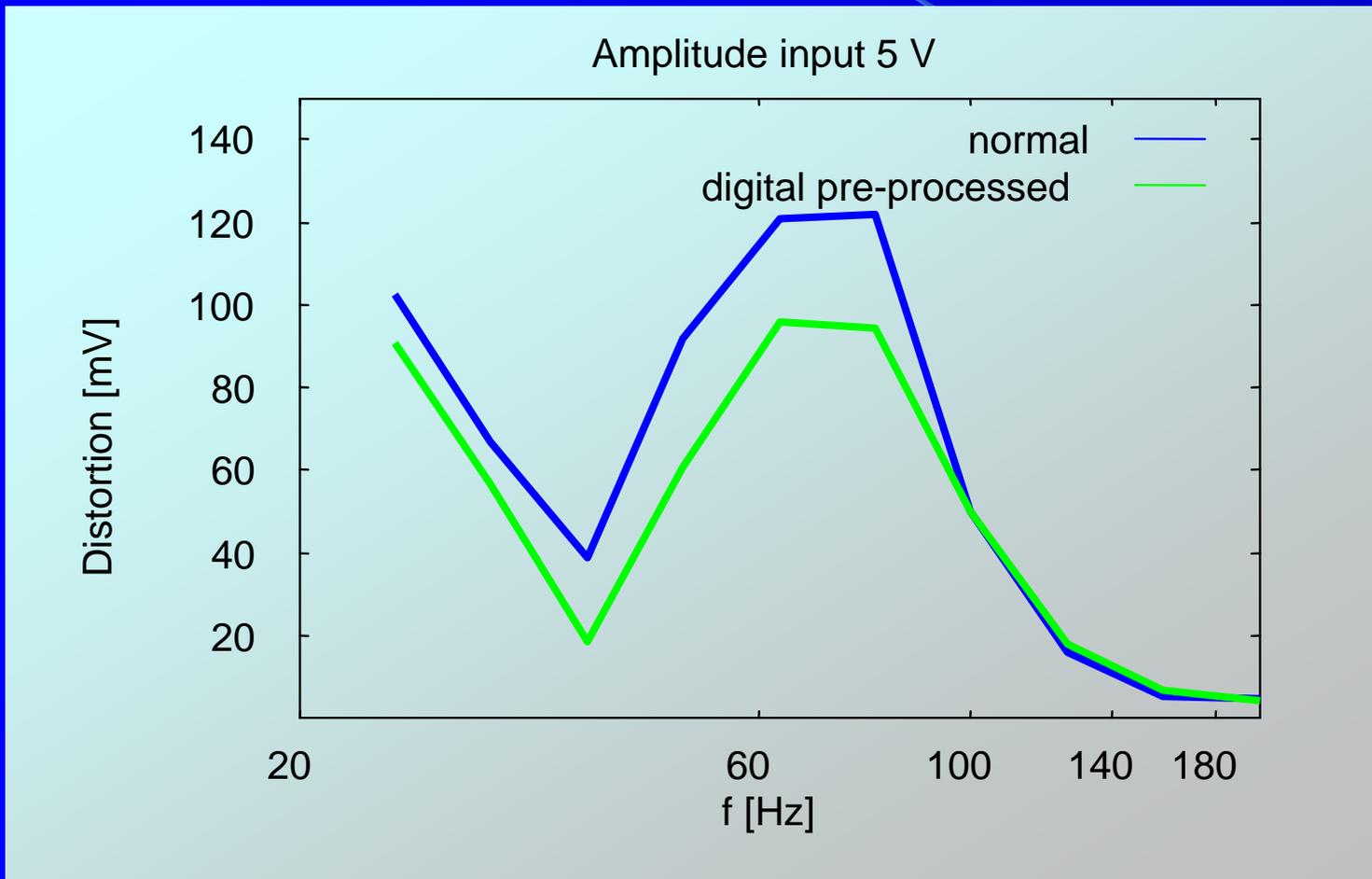


Distortion (THD) Results





Distortion (THD) Results





Conclusions

- Definition of a non-linear model for low frequency loudspeaker systems;
- Design of a parametric audio processor for the compensation of non-linear distortion of loudspeaker;
- Implementation of the audio processor with a low cost commercial DSP;
- Measured reduction of Distortion with the insertion of the audio processor;



The future

- New Hardware (Analog Devices SHARC on the EZ-KIT stand-alone board)
- Inverse filters computed with the Kirkeby regularization technique
- Full-band linearization by means of “warping” instead of dual-band processing
- The inverse filter will include also the acoustics behaviour of the car compartment