# Sound acoustics

Research, Development, Implementation

# Quality enhancement of audio systems based on digital ELAComp filter (Electro Acoustic Compensation)



## 1. Problem today

Electro acoustic components and systems are found in all kinds of products like TV, PC, headphones, headsets, various telecommunication systems and hearing aids.

In most cases the quality of such systems is influenced disadvantageously by the product design and hence is often the weakest link in the signal processing chain.

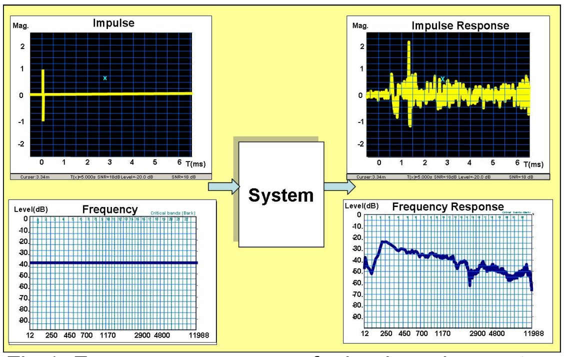


Fig 1: Frequency response of a loudspeaker system with dull sounding signal reproduction

The playback sounds unarticulated, dull or shrill leading to decreased intelligibility particularly for speech and so leads frequently to an increased echo problem in telecommunication systems.

## 2. State of the art

In order to compensate above mentioned disadvantages pre-filters (bass and treble) are introduced in the signal path of the loudspeaker. However, this is not solving the core problem at all.

Equalizers can not be tuned to all frequency dependent nonlinearities and are not suitable for correct acoustic frequency compensation.

Perfect frequency compensation can be achieved with the aid of a FIR filter by the convolution of the signal with the impulse response, which results from the deviation to the desired frequency response. Tests with this approach show however, that acoustic resonance frequencies are eliminated only with an extreme long impulse response window and hence an efficient real time processing is not possible.

IIR filters are difficult to tune due to the resonance behavior of the filter poles and the variable group delay. Tests with different filter design principles (least square, frequency sampling…) did not show any satisfactory results.

## 3. ELAComp a new filter for clean sound quality

ELAComp is a new filter principle with high frequency resolution and without latency making the reproduction of an impulse at the output of an unknown system possible. Hence a nearly perfect equalization of the frequency response is achievable (Fig. 2).

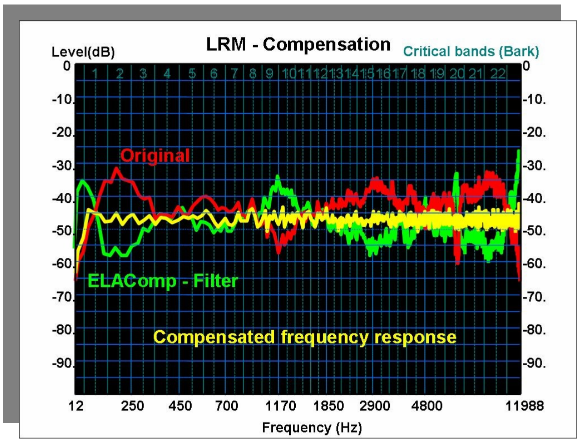


Fig 2: ELAComp filter for a clean sounding signal reproduction

The filter can be adapted directly with the measured transfer function of the system under test without iterative computations leading to a fast and precise convergence. The computational effort for each channel comprising only about 220 MAC.

A reproduction of an impulse with a rest error of about -40dB is possible, if the compensation does not exceed the limits

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of the available margins. Table 1 shows the comparison between ELAComp and conventional filters.

| | ELAComp | FIR | IIR |
|---|---|---|---|
| **Convergence** | ++ | + | - |
| **Impulse response** | precise | precise | diffuse |
| **Group delay** | constant | constant | variable |
| **Latency (ms)** | 0 | 10 | variable |
| **Computational effort (MAC)** | 220 | 2400 | 500 |

Table 1: ELAComp versus conventional filters

### ELAComp - Tools

The computation of the ELAComp filter coefficients is supported by ELACDesign, a program for Windows operating systems.

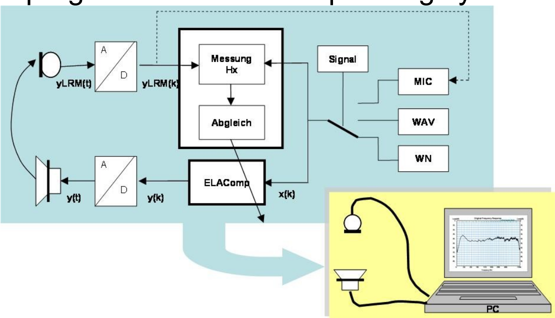


Fig 3: ELACDesign for the evaluation of the possible product enhancement

ELACDesign computes the product dependent filter coefficients within a few minutes for measurement and evaluation procedure in real time.
The playback of standard conform recordings (*.wav) with and without ELAComp makes subjective listening comparison tests possible. Hence it is possible to prove the possible quality enhancement of an unknown system with less effort immediately after filter computation.

### Results with ELAComp - filters

Tests with different electro acoustic components mainly low cost consumer products were carried out at 8, 24 and 48 kHz sampling rate (SR) within following system arrangements.

- **Loudspeaker – Microphone (LM)**
  Compensation of quality degrading loudspeaker cabinet designs.
- **Loudspeaker – Room-Microphone (LRM)**
  Compensation of quality degrading arrangements in rooms including the compensation of loudspeaker systems.
- **Room – Microphone (RM)**
  Compensation of the room acoustics at the microphone.

Each system under test showed big unevenness in the frequency response The maximum level difference within the indicated 3 dB bandwidth amounted to ± 20 dB around the average level measured over several critical bands. Using the ELAComp filter led to a huge improvement in the frequency response showing only about 1 dB deviation based on the same averaging method

Objective tests were carried out with the HASQUE® principle. Simulations with test recordings showed an enhancement of up to 2 on the five point MOS scale.

Considering communication systems, where the loudspeaker signal is fed back via the room microphone path to the far end partner, the echo return loss (ERL) could be increased with the ELAComp filter by up to 12 dB maintaining the same loudness.

Subjective listening tests proved that fundamental perceptible quality enhancement is achievable with the new ELAComp filter principle. Speech and music signals sound more natural and clean with enhanced intelligibility.

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