Abstract

Usually the goal of loudspeaker design is to achieve the best sound quality. To achieve that, one of the requirements is the flatness of the frequency response. The best way to do it is the proper design of the loudspeaker. However, if we buy a product, it's obvious we can't change the inner design of the equipment, but we can make the response flat by digital equalization. It means that we filter the signal prior to sending it to the loudspeaker. The frequency response of the filter compensates the errors of the loudspeaker response. For that, a DSP can be used to filter the signals.

When compensating the frequency response we aim to reduce the ripples of the transfer function measured at the listening position. We also have to pay attention not to make it worse in other points of the room. Our main goal is to make the frequency response as flat as possible in a larger area. For that, we compensate only the main characteristics of the frequency response because the little details are position-dependent. This can be achieved by designing the filter using the fractional octave smoothed version of the frequency response.

This thesis presents the steps of loudspeaker equalization using MATLAB for measurement and filter design. After choosing the best signal for measurement and the best filter type for this application, I apply a fractional octave smoothing to the frequency response. Furthermore I define the frequency response of the target considering the frequency band limits of the loudspeaker. I design FIR equalizer filters in frequency- and time-domain using the smoothed version of the measured frequency response. Then I compare the designed filters by measuring their errors compared to the target function. Finally I implement the chosen filter in an Analog Devices SHARC DSP and check the results objectively by measurements and subjectively as well.