

Bass/treble - advanced

1 From bass/treble to an equalizer

In the previous hands-on you programmed a digital version of a bass/treble knob. This allowed some amplification/attenuation of the low/high frequencies, just as on an old amplifier. You might think that this is not enough, so let's get a bit more advanced and construct a digital version of an equalizer!

1.1 Five knobs to turn

Implement a function that constructs a FIR filter mimicking five different bands (subbands) of an equalizer, each covering an equal bandwidth on a linear frequency scale. The function takes the gains (which can be negative or positive) for each of the subbands and returns the filter coefficients of the resulting (overall) filter. Choose the order of the filter yourself and explain your choice.

- Plot the frequency response of the filter with gains that decrease in steps of 5 dB from the lowest to the highest subbands
- Zero-pad your impulse response to twice/ten times the length and plot again the frequency response. Explain what you see.
- Pass a white noise signal through your filter and plot the spectra of the original and the filtered signal on top of each other.
- Process your favourite piece of music through the filter and listen to it! It's fantastic, isn't it?

2 The grandparents and DSP

It was suggested by one of the feedback persons to have some "real" questions in the assignments. So here they come. Please write your answer in technical terms and in a concise way. Assume you would want to explain the issue to a colleague whos DSP knowledge might be a bit rusty:

1. Assume your grandmother is bored during christmas eve and is generating a sweep in MATLAB (sampling frequency of 5000 Hz), starting at 50 Hz and reaching up to 5000 Hz. Now she plays back the sweep in via your stereo system in your living room. Despite being impressed by your DSP-grandmother, your parents are a bit puzzled by the result. Explain a) what you expect to hear, b) why your parents are puzzled and c) what the reason behind the observed phenomenon is.
2. Now your grandfather woke up and also wants to play with DSP. But since the computer is occupied by your grandmother, he takes pen and paper and wants to impress people with his math skills. He shows that the stock market curve in the newspaper is smoothed by the application of averaging 3 subsequent points on the graph. He forgot what the time interval between two sampling points are, but figured out a way to have a generalized description. He found that some fluctuations would be totally suppressed by this averaging technique and explains this using his drawing of the frequency response of this 3-point filter. He also

managed to compute the analytical form of the frequency response and further showed that this filter has linear phase and can even have zeros phase under some conditions. Derive the frequency response of the filter used by your grandfather and show the linear/zero phase of that filter. Calculate which frequency would be totally suppressed if the update time between two points on the stock market would be one second. Explain how this frequency response can be expressed without knowing the exact sampling frequency.