Sheet 12 – Digital signal processing

Please write all commands in the MATLAB editor into separate m-files (one for each function or script) and save it in a folder that you specifically dedicate to this workshop. If you don't know how a command is being used type "`help [commandname]`" into the command window. Comment each code line briefly to document what it is doing.

Exercise 1:

a) Load the impulse response `'imp_response_church_48k.wav'` and the other audio files with their respective sampling frequencies into MATLAB.
b) Use the function `resample()`, in order to bring all signals to having the same sampling frequency of 48 kHz.
c) Plot one of the audio signals of your choice (1) in the time domain and (2) in the frequency domain. Pay attention to do the spectral plot with an appropriate frequency axis.
d) Listen to the audio file.

Exercise 2:

The impulse response `'imp_response_church_48k.wav'` characterizes the acoustic transmission from a source to the receiver inside a church.

Convolve one audio signal of your choice with this impulse response to create an audio signal at the location of the receiver inside the church. How do time signal, spectrum and sound change?

Exercise 3 (optional):

Create a Butterworth low pass filter of 4^{th} order with a cutoff frequency of 600 Hz at the same sampling frequency as before (48 kHz). Filter one (clean) audio signal with this filter. How do time signal, spectrum and sound change now?