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Fakultät für Informatik
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Assignment 3

„Multimedia“

Deadline 30.05.2016, F033.

Description of assignment
In this assignment the fast Fourier transform (FFT) is to be used to encode audio data. As for the LPC method the audio data is decomposed into blocks. Each block is transformed to the frequency domain using the `fft` command. After that the frequencies with smallest amplitudes are removed. The resulting data vector is then transformed back to the time domain using `ifft` and the output signal is combined from the sequence of the resulting vectors. Because many low-amplitude frequencies are masked, the differences are audible as slight distortions only, even if many frequencies are removed.

For the solution use the following templates and audio files:
1. `run_fft.m`: Template for the transformation.
2. `*.wav`: Audio files.

Hints:
- `http://www-home.htwg-konstanz.de/~umlauf/Skript/Matlab.en.pdf`
- To read and write audio files use `wavread` and `wavwrite`.
- To decompose the audio data into blocks and to combine these blocks back to an output signal, index vectors can be used. Using `index=1:block_length` generates an index vector and `x(index)` yields access to the first elements in vector `x`. This can also be used in assignments, e.g. `x(index)=0` sets all values to zero in vector `x`, that are indexed by `index`. The index vector is incremented by `index=index+block_length`.
- Audio output can be triggered by `sound` and `soundsc` (sampling frequency 12000 and 16000).
- The command `specgram` is used to visualize audio data. Comparing audio input with your audio output using `specgram` shows the removed frequencies.
- To search for frequencies with small amplitudes you can use the commands `abs` and `sort`.

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