

Matlab GUI for WFB spectral analysis

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Abstract

In the case of the sound signals analysis we usually use logarithmic scale on the frequency axis. Computing of a sound signal spectrum by the Fourier transform does not bring ideal results in this case. WFB spectral analysis is a new method which combines wavelet filter bank with the Fourier transform. This method brings good frequency resolution on the low frequencies together with a fast response on the high frequencies. This paper consists of a short introduction to the WFB spectral analysis and a description of the Matlab GUI for the WFB spectral analysis.

Introduction

Human ear can hear sound signals with frequencies from 20 Hz. Maximal audible frequency is approximately 16 kHz depending on the age of a subject and other factors. Human ear frequency sensitivity is not the same at low and at high frequencies. Human ear is much more sensitive for frequency changes at low frequencies. This phenomenon is described in the theory of Critical bands. Introduction to this theory can be found in [2, Chapter 6].

Because of this phenomenon in the case of sound signals analysis we usually use logarithmic scale on the frequency axis. Computing of a sound signal spectrum by the Fourier transform does not bring ideal results in this case. Frequency resolution on the low frequencies is too poor and resolution on the high frequencies is too detailed on the other hand. For the better frequency resolution we have to use more samples that is not effective.

Problem of the FFT using for the sound signal frequency analysis solves new method called WFB spectral analysis.

1 WFB spectral analysis

Description of the new method named the WFB spectral analysis is in this chapter. Block diagram of the WFB spectral analysis is shown in Fig. 1. Detail description of each system block is performed in following chapters.

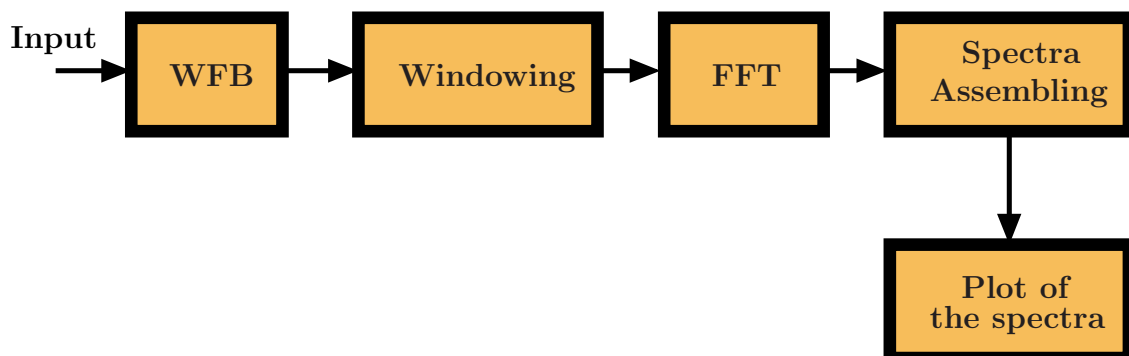


Figure 1: Block diagram of the WFB spectral analysis.

1.1 Wavelet Filter Bank

In the WFB spectral analysis is an input sound signal decomposed by a wavelet filter bank. This wavelet filter bank consists of 7 one-stage decomposition blocks. As shown in Fig. 2 an input signal is divided into two separate parts in each one-stage block. The first part is filtered by a high pass filter and then decimated. This output is called a detail. The second part of the input signal is filtered by a low pass filter and then decimated too. This second output is called an approximation.

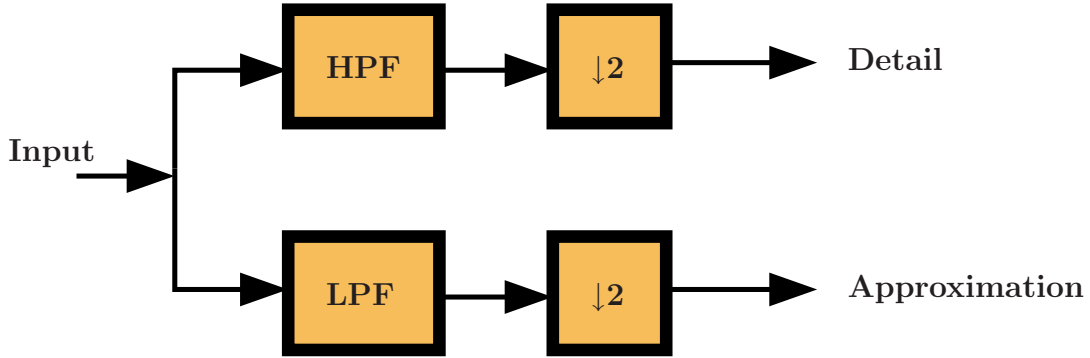


Figure 2: Detail of a one-stage decomposition. Approximation and detail is at the output.

Such design of the one-stage decomposition does not increase amount of a data (because of the decimation), signal is critically sampled. For N input samples is received $round(\frac{N}{2})$ samples of an approximation and $round(\frac{N}{2})$ samples of a detail. Decimation of the detail cause mirroring of the spectral components which are actually situated in the second Nyquist zone.

The approximation of the first one-stage block is re-decomposed by the second one-stage block, while the detail is not. This one-stage decomposition of the previous one-stage approximation is made seven times. Seven details (one from each one-stage block) and an approximation from the last seventh block is received on the output of the filter bank. Detail diagram of the WFB spectral analysis wavelet filter bank with designated output frequency bands is shown in Fig. 3. An inspiration for the designed wavelet filter bank was [3].

Main advantage of such procedure is that eight output signals representing eight different frequency bands can be processed separately according to the specific needs of each frequency band.

Disadvantage of this procedure is a possibility of aliasing. As the signal is critically sampled in the output frequency bands and frequency characteristics of the high pass and low pass filters are not ideal aliasing may occur on the edges of the bands. New spectral components may arise because of the aliasing.

For more information about wavelet filter banks see [1].

1.2 Windowing

Short time Fourier transform is used in the WFB spectral analysis. When this transform (implemented by the FFT algorithm) is used the signal has to be cut into the blocks of N time samples. Number N of the samples in one block is usually power of two. System block named Windowing performs this sequencing of the signal into the blocks in all frequency bands.

The most simple window function is a rectangular function. Unfortunately it has very bad spectral characteristic so it is used occasionally. There are several other window functions, the hamming window function is currently used in the WFB spectral analysis.

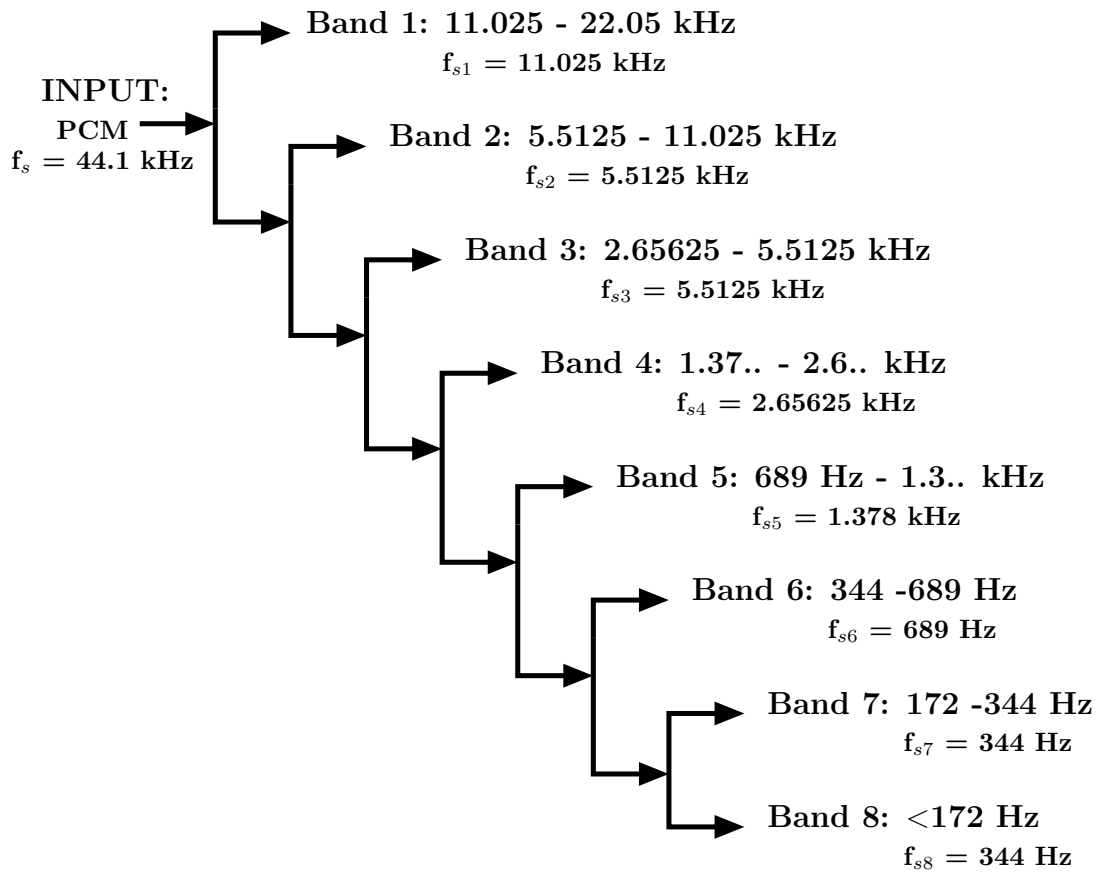


Figure 3: Used wavelet filter bank with the appropriate frequency bands and the sampling frequencies.

Windowing is performed in each frequency band separately so it is possible to use different window functions across the bands.

High frequency components (more than 11 kHz) from the first frequency band are windowed by a very short window function. Therefore output spectrum refresh can be high. Low frequency components (less than 172 Hz) from the last frequency band are windowed by a very long window function so frequency resolution can be high.

1.3 FFT

FFT block performs short time Fourier transform of the current block of samples. FFT is computed in each frequency band separately. Only magnitude in decibels is currently displayed so only magnitude part is on the output of the FFT block.

1.4 Spectra Assembling

System block Spectra Assembling collect data from all 8 FFT blocks. Keep in mind, that output magnitude spectrum of the first 7 blocks is mirrored because of the decimation in the Wavelet filter bank block while the last one is not mirrored because it is an approximation not a detail.

Block takes the first half of each spectrum (up to the half of the sampling frequency), collect them together and computes vector of correspondent frequencies.

1.5 Plot of the spectra

Block Plot of the spectra draw the computed spectrum. Block use `semilogx` function so frequency axis has the logarithmic scale, the magnitude is displayed in decibels.

2 Comparison of the classic FFT and WFB analysis

Classic frequency analysis made by FFT and frequency analysis performed by WFB are compared in Fig. 4. Both analyses were made for 256 output frequency samples. Analysis made directly by Fourier transformation has even distribution of the frequency samples so when the result of such an analysis is plotted with the logarithmic scale of the frequency axis, the resolution at the low frequencies is bad and occurrence of the samples at the high frequencies is very dense. Impulse signals may be also masked when using common FFT analysis.

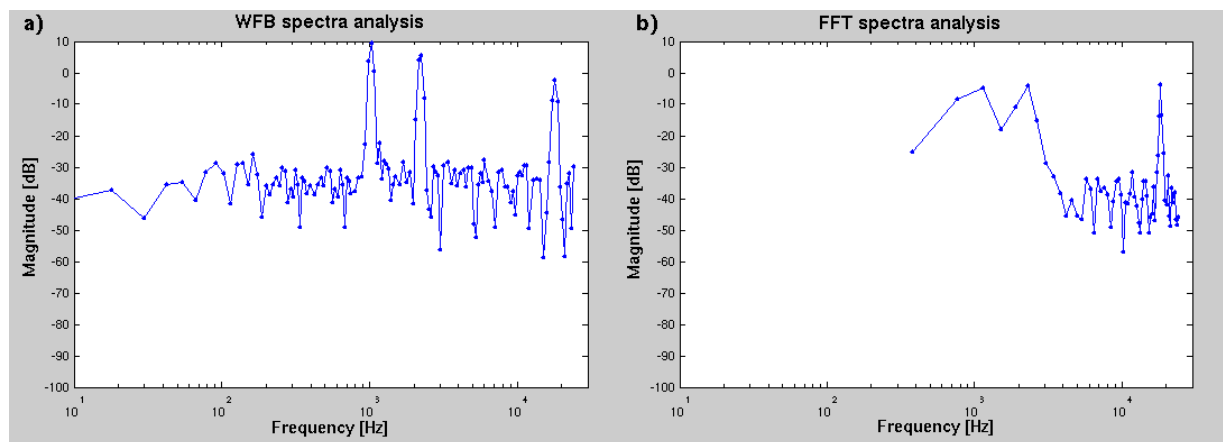


Figure 4: Comparison of the classic FFT b) and WFB a) spectral analysis. Both analyses were made for 256 output frequency samples.

WFB spectral analysis does not have even distribution of the frequency samples therefore when the result of such an analysis is plotted with the logarithmic scale of the frequency axis the resolution stays good even at the low frequencies.

3 Matlab GUI for the WFB spectral analysis

For the WFB spectral analysis was designed a special GUI. The GUI is shown in Fig. 5. It consist of two sliders, three buttons and a plot area where graphs are displayed. The first slider is used for a sliding in the time. Current version of the designed system can analyse pre-processed audio files only. Real-time version is being developed at the moment. The second slider is used for the amplitude scaling. By the pressing of the first button is possible to switch to the time mode. In this mode time course is displayed in plot area. Second button is for the switching to the FFT mode. In this mode common FFT analysis is performed. Third button is for the new WFB analysis. Due to the difficult real-time computing of the WFB sound signal analysis an input signal is pre-processed so the response of the result display is very fast.

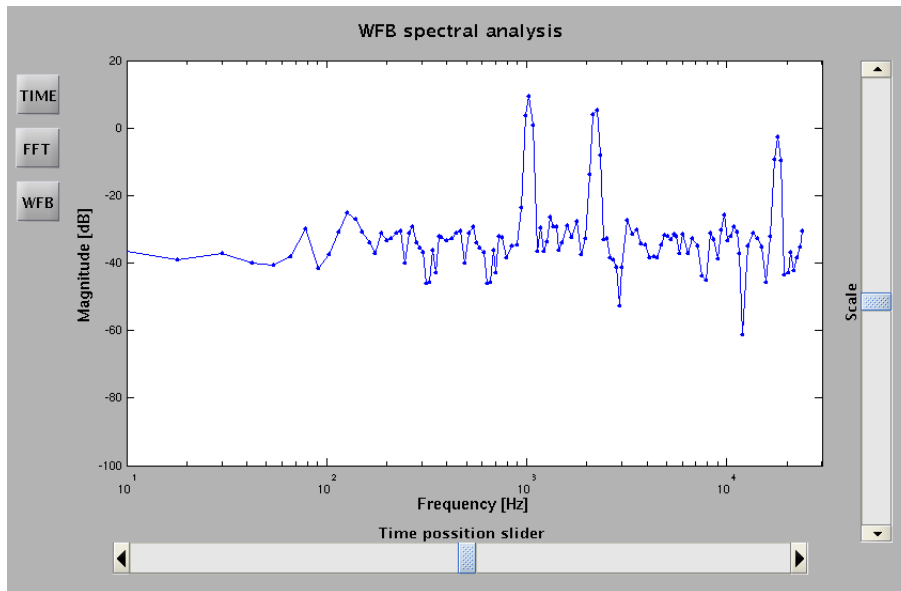


Figure 5: Designed Matlab GUI for WFB sound signal spectral analysis.

Conclusion

New method for the sound signal analysis was implemented in the Matlab. For the possibility of comparison was implemented also common frequency analysis based on the Fourier transform. Simple Matlab GUI was designed for the easier frequency analysis of the sound signals. Time course, FFT spectral analysis and WFB spectral analysis of the sound signals can be performed by the designed system.

Design of the whole system was made with a useful help of the book [4].

References

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