THE MATLAB TOOLBOX FOR SYNTHESIS/ANALYSIS UNIFORM AND NON-UNIFORM POLYPHASE FILTER BANKS*)

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Abstract. This article presents the software for the design and performance evaluation of polyphase multirate filter banks. As the MATLAB toolbox, it can be easily applied and integrated with existing code. In addition to well known uniform solutions, there were elaborated less popular non-uniform systems based on all-pass transformation. We believe that this is the first implementation of novel all-pass transformed cosine modulated filter bank.

1. Genesis of the toolbox

Undoubtedly the MATLAB environment is the essential tool for the DSP people. In fact, with permanently developed signal processing toolbox, it became standard for experiments, simulations and prototyping. However there is a lack of the procedures for even most classical polyphase filter banks [1, 2]. As the subband decomposition [3] is the base technique for many modern algorithms (e.g. signal compression or enhancement), the filter banks realizing it represent very important area of contemporary DSP. So they should be appropriately supported.

The considered toolbox was created with the intention of filling up these shortcomings. It arose as a result of MSc thesis and then was developed to assist every-day research work.

2. The toolbox contents

The main part of the toolbox is the set of the procedures implementing different filter bank structures. The functions appear in the complementary pairs – one is for the analysis stage, and the second represents corresponding synthesis filter bank.

There are implemented the following systems:

- uniform critically sampled DFT modulated filter bank,
- uniform oversampled DFT filter bank,
- uniform critically sampled cosine modulated filter bank,
- non-uniform all-pass transformed DFT filter bank,
- non-uniform all-pass transformed cosine modulated filter bank

Associated GUI allows interactively working with the filter banks. First, the structure can be selected and its parameters fixed. Than the coefficients of prototype FIR filter are automatically generated by preferred algorithm (several windowing techniques and optimization minimizing Chebyshev or least-square error). Next, one of several predefined sequences or data loaded from external WAV-file can be passed through the joint analysis / synthesis system. As input signal, as channel and output sequences can then be observed both in time and frequency domain. If it was passed the unit impulse, then responses of the system can be analyzed. There is also possibility to listen both input and output signals played as sounds.

3. The details of novel cosine modulated filter bank

The works on the toolbox bring the idea of novel non-uniform filter bank, mixing the concept of all-pass transformation with well known cosine modulation scheme.

The all-pass transformation [4] is a simple method to obtain non-uniform filter bank. It consists in replacing all the delays in base polyphase structure by identical, causal and stable, all-pass filters

$$z^{-1} \quad \Rightarrow \quad A(z)=\frac{z^{-1}-a}{1-az^{-1}} \qquad |a|<1 \tag{1}$$

Such a filter has unity magnitude response and nonlinear phase

$$A\left(e^{j\omega}\right)=e^{j\phi(\omega)} \qquad \phi(\omega)=-\omega+2\arctan\frac{a\sin\omega}{a\cos\omega-1} \tag{2}$$

The all-pass transfer function represents conformal bilinear mapping of unity circle to itself [5]. As the frequency scale of the system is deformed, its frequency response becomes warped. The character of the deformation depends only on the value of coefficient a and is explicitly given by the plot of the negative phase $-\phi(\omega)$. This is explained in Fig. 1.

This approach has been used successfully to DFT filter banks. The systems approximating critical band, according to psychoacoustic Bark scale, found application in speech enhancement [6]. However, the experiments showed that one of the main drawbacks is the complexity of channel sequences, and, in turn, high computational load of in-band algorithms dealing with complex numbers.

We ascertained that the above trouble can be solved by application of the all-pass transformation to the classical pseudo QMF system.

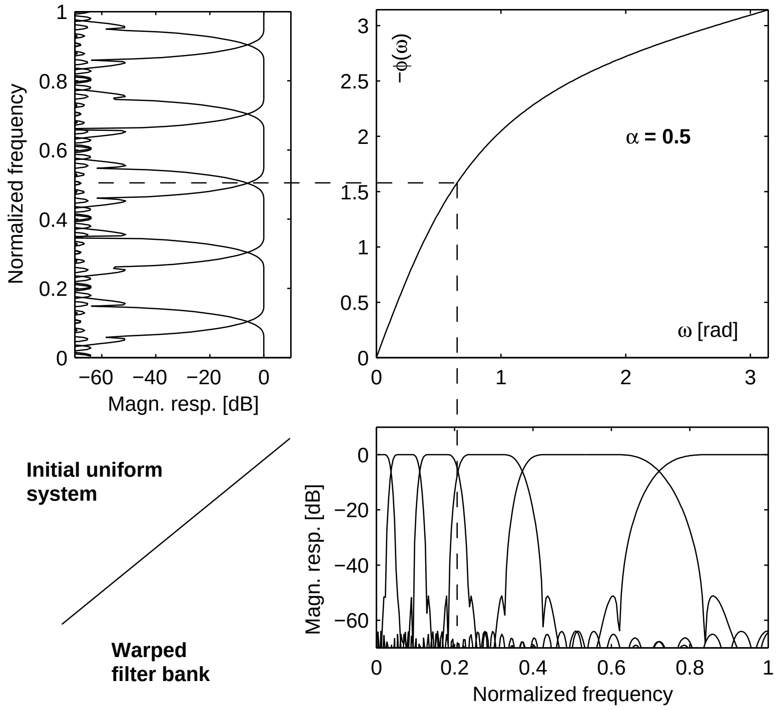

Fig. 1 Filter bank frequency response warping as the effect of all-pass transformation

Band filters in K-channel pseudo QMF system are cosine modulated versions

$$\begin{aligned} h_k(n) &= 2p(n)\cos\left(\frac{\pi}{K}\left(k+\frac{1}{2}\right)\left(n-\frac{N}{2}\right)+\theta_k\right) \quad \text{synthesis} \qquad & k &= 0..K\text{-}1 \\ & & n &= 0..N \\ f_k(n) &= 2p(n)\cos\left(\frac{\pi}{K}\left(k+\frac{1}{2}\right)\left(n-\frac{N}{2}\right)-\theta_k\right) \quad \text{analysis} \qquad & \theta_k &= (-1)^k\frac{\pi}{4} \end{aligned} \tag{3}$$

of one prototype $p(n)$, supposed to be even order real-coefficient FIR filter with cut-off frequency $\pi/2K$. As cosine is equivalent to the sum of two conjugate exponents, this modulation can be efficiently realized with the help of FFT and frequency responses are expressed as

$$H_k(e^{j\omega}) = e^{j\theta_k}W^{(k+0.5)N/2}P\left(e^{j\omega}W^{k+0.5}\right)+e^{-j\theta_k}W^{-(k+0.5)N/2}P\left(e^{j\omega}W^{-(k+0.5)}\right) \qquad W=e^{-j\frac{\pi}{K}} \tag{4}$$

The all-pass transformation of such a system results in the structure of Fig. 2, having warped responses of the form

$$H_k(e^{-j\phi(\omega)}) = e^{j\theta_k}W^{(k+0.5)N/2}P\left(e^{-j\phi(\omega)}W^{k+0.5}\right)+e^{-j\theta_k}W^{-(k+0.5)N/2}P\left(e^{-j\phi(\omega)}W^{-(k+0.5)}\right) \qquad W=e^{-j\frac{\pi}{K}} \tag{5}$$

Owing to cosine modulation, the channel signals are purely real. This comes at the cost of higher complexity of subband analysis / synthesis, but the algorithms operating in the channels simplify.

The system maintains common advantages of its DFT former. Design is easy as it depends on only one FIR prototype. The flexibility of bandwidth shaping allows good approximation of the critical band of hearing [7]. One can also consider real-time dynamic system reconfiguration by simple change in warping coefficient.

4. Summary and future works

The presented toolbox should be a valuable tool in DSP oriented education, science and engineering, where multirate filter banks are studied independently or applied in complex subband processing systems, solving actual problems. Some further improvements may be appropriate – the most important are: implementation of recently developed, uniform oversampled and biorthogonal filter banks [8], extending design methods to perfect reconstruction case, and successive introduction of the structures, which are less common and practical, but interesting from scientific point of view.

On the other hand, developed all-pass transformed filter bank with cosine modulation can find some applications, when flexible non-uniform bandwidths are needed and reality of the channel signals is preferred.

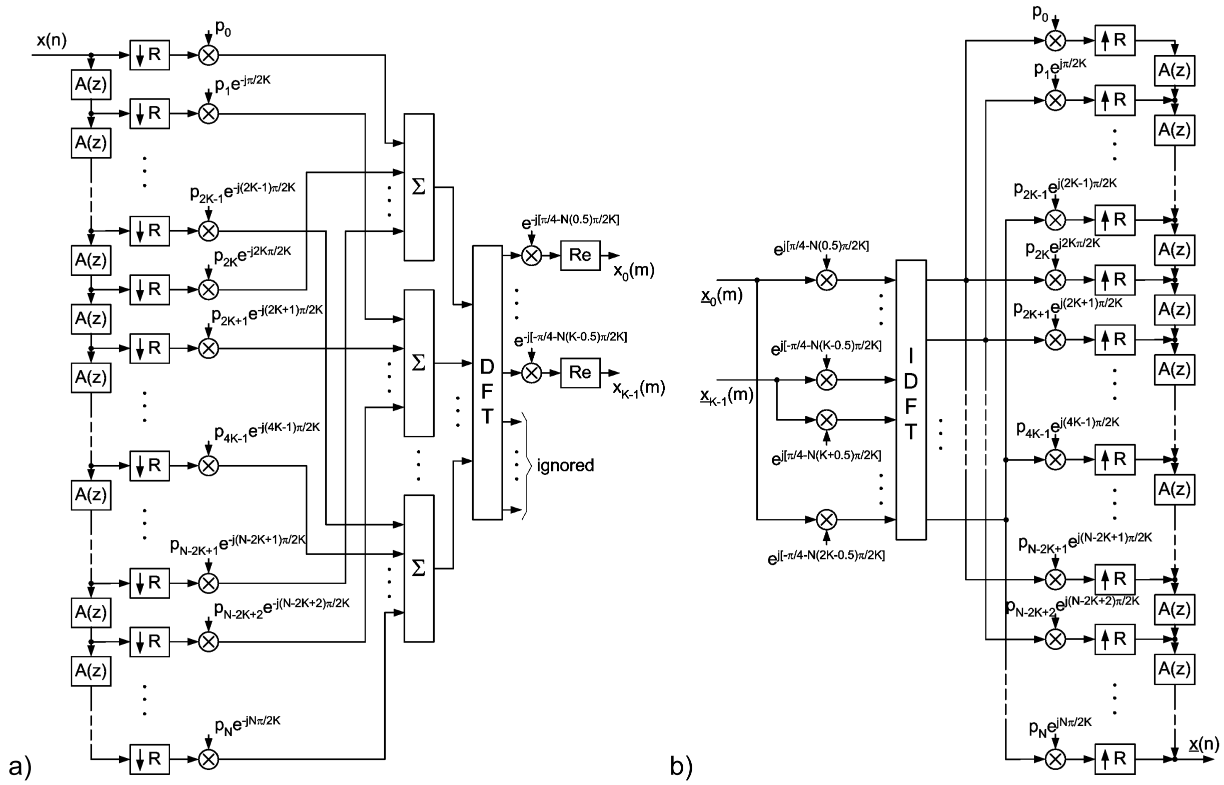

Fig. 2 The structure of the analysis (a) and synthesis (b) non-uniform cosine modulated filter bank

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