Efficient implementation of QMF filter bank for power harmonic analysis on digital signal processor

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Abstract— *In this paper the efficiency of the polyphase decomposition form against the direct decomposition form of quadrature mirror filter bank for power harmonic analysis is investigated. For that purpose, the both forms of decomposition in 3-levels were implemented on digital signal processor TMS320C6713 using SIMULINK. The obtained results were presented on the conference "EUROCON 2015". In addition to that, the polyphase and direct forms of decomposition in 5-levels were implemented on the same processor, directly in Code Composer Studio. Experimental results from both investigations show increased efficiency when polyphase decomposition form is used. Also, the influence of the input finite length signal extension on the accuracy of calculated RMS result is investigated, for the aim of reliable harmonic analysis. The investigation has shown that the periodic signal extension gives best results in case of stationary input signals and can be used for both direct and polyphase implementations.*

Keywords—*discrete wavelet transform, wavelet packet transform, input signal extension, polyphase implementation, harmonic analysis*

I. Introduction

As a result of the widespread use of nonlinear power electronic devices significant disturbances in Power systems are caused, such as reactive power burden, transient oscillations, voltage dips and swells, harmonic distortion, inter-harmonics and etc. The accuracy of Power system harmonic analysis is essential for evaluating the overall power quality. The Discrete Fourier Transform (DFT) is proposed in the IEC Standard 61000-4-7 [1] for the measurement of harmonics and inter-harmonics in power supply systems as the processing tool for harmonic analysis. Time-variations of individual harmonics are analyzed by short-time Fourier transform (STFT) which provides both time and frequency based views of a signal, but the drawback is that once a particular size is chosen for the time window, that window is the same for all frequencies. The results of transient harmonic analysis are therefore not satisfactory.

In recent years, use of Discrete Wavelet Transform (DWT) as a powerful signal processing method is receiving increased attention for Power Quality (PQ) analysis. Many wavelet based algorithms for harmonic analysis in power systems are proposed [2]-[7]. Reported results are competitive with the results obtained using the harmonic-group concept proposed by the IEC for different measurement conditions, showing the potential of the wavelet analysis as an alternative processing tool for the harmonic estimation in power systems. Additionally, the wavelet's dilation and translation property gives time and frequency information accurately. Apart from it, the process of shifting enables the analysis of waveforms containing non-stationary disturbance events. These unique properties are making DWT best suited for PQ analysis. The form of Dyadic Discrete Wavelet Transform (DDWT), which is a set of low-pass and high-pass wavelet filters, is being most frequently applied, in point of fact its more advanced version – Wavelet Packet Transform (WPT). WPT can decompose the frequency spectrums of a signal into uniform bands and thus it improves the accuracy of harmonic detection [2], [7]-[10].

Vaidyanathan with 24 coefficients (v24), Daubechies with 20 coefficients (db20) and Coiflet 5 with 30 coefficients (coif5) are proposed in the literature as the most adequate wavelet filters for harmonic analysis [6]. In [11] and [12] is shown that Johnston's filters with 32 coefficients are also suitable for harmonic analysis. This is due to their better selectivity compared to well-known wavelet filters. One of the main advantages of these filters is that they are with linear phase, and more efficient WPT form can be implemented using polyphase representation of the decimation filters. The efficiency of the polyphase implementation and the accuracy of the RMS calculations are the main objective in this paper.

II. WAVELT TRANSFORM FOR HARMONIC ANALYSIS

A. *Discrete wavelet transform (DWT)*

The discrete wavelet transform is given with the equation:

$$DWT[m,n] = \frac{1}{\sqrt{a_0^m}} \sum_{k=-\infty}^{\infty} f[k]\psi\left[\frac{k - nb_0 a_0^m}{a_0^m}\right] \qquad (1)$$

where function ψ is the base function or the mother wavelet, and *b* and *a* are dilatation and translation parameters, respectively. With the choice $a_0 = 2$ and $b_0 = 1$, a dyadic orthonormal wavelet transform is obtained [13], [14] and can be easily and quickly implemented by filter bank techniques known as Multi-Resolution Analysis (MRA). The filter bank is used to decompose the signal into various levels using alow-pass filter with a transfer function $H_0(z)$ and a high-pass filter with a transfer function $H_1(z)$, as shown in Fig.1. The basic idea of the MRA is that of the successive approximation, together with that of “added detail”. The low frequency part (approximation signal) is split again into two parts of high and low frequencies. Depending on the application and on the size of the input signal, process could be repeated several times. As result, logarithmic decomposition of frequency spectra of the input signal is obtained.

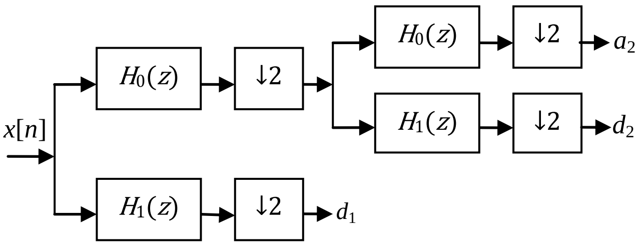

Fig. 1 DWT decomposition over 2 levels

Obtained approximation and detailed wavelet coefficients have non uniform frequency bands and cannot be used for measurement of RMS values of different harmonic components. This limitation can be overcome with the use of WPT.

B. *Wavelet Packet Transform (WPT)*

The Wavelet packet transform (WPT) is a generalization of DWT. The difference is that in the WPT signal decomposition, both the approximation and detail coefficients are further decomposed at each level and as the result a uniform frequency decomposition of the input signal is obtained. The number of output bands for *L*–level decomposition is 2^L. An example for decomposition over 2 levels is shown in Fig. 2. With selection of an adequate sampling frequency and level of decomposition these uniform frequency bands can be used for harmonics measurement of the input signal [3], [4].

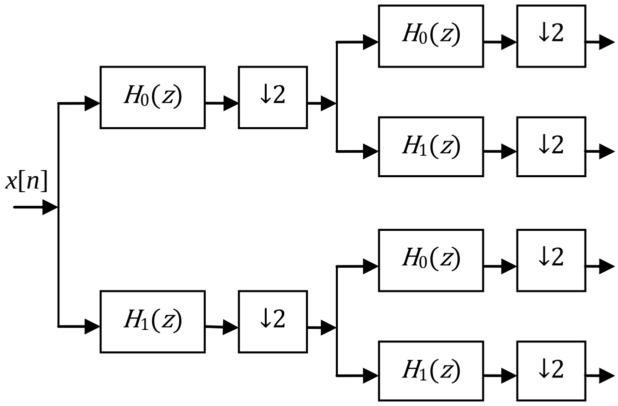

Fig. 2 WPT decomposition over 2 levels

The RMS of voltage can be computed directly from the wavelet packet coefficients [3], [4]:

$$\begin{aligned} V_{rms} &= \sqrt{\frac{1}{T}\int_0^T v(t)^2} \cong \sqrt{\frac{1}{2^N}\sum_{n=0}^{2^N-1} v[n]^2} \\ &= \sqrt{\frac{1}{2^N}\sum_{i=0}^{2^j-1}\sum_{k=0}^{2^{N-j}-1}\left(d_j^i[k]\right)^2} = \sqrt{\sum_{i=0}^{2^j-1}(V_j^i)^2} \end{aligned} \qquad (2)$$

Where $V_j^i = \sqrt{\frac{1}{2^N}\sum_{k=0}^{2^{N-j}-1}\left(d_j^i[k]\right)^2}$.

In the equation V_j^i is the RMS value of frequency band at node *i* and $d_k^i[k]$ are the wavelet coefficients at node *i*.

III. POLYPHASE IMPLEMENTATION OF TWO-CHANNEL QUADRATURE MIRROR FILTER BANK

A. *Two-channel quadrature mirror filter bank*

The basic structure of a two-band filter bank is shown in Fig. 3. $H_0(z)$ and $H_1(z)$ designate the low-pass and high-pass analysis filters, respectively, and $F_0(z)$ and $F_1(z)$ designate the low-pass and high-pass synthesis filters, respectively.

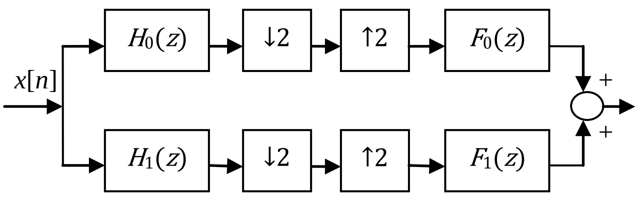

Fig. 3 Two band filter bank

The input-output relation of a two-band filter bank is given with the following equation:

$$\hat{X}(z) = \frac{1}{2}\left[H_0(z)F_0(z) + H_1(z)F_1(z)\right]X(z) + \\ + \frac{1}{2}\left[H_0(-z)F_0(z) + H_1(-z)F_1(z)\right]X(-z) \tag{3}$$

The first term describes the transmission of the signal through the system, while the second term represents aliasing error due to the change of sampling rate in the filter bank. The simplest way to cancel the aliasing is by selecting the filters in the analysis stage as

$$H_1(z) = H_0(-z) \tag{4}$$

and by selecting the synthesis filter as

$$F_0(z) = H_0(z),\ F_1(z) = -H_1(z)\,. \tag{5}$$

Since the mirror-image symmetry about the frequency $\omega = \pi/2$ exists between $H_1(z) = H_0(-z)$, these filters are known as quadrature mirror filters (QMF) [13]. The two-channel QMF bank structure is known as critically sampled filter bank as decimation, and interpolation factors are equal to number of bands. Therefore all the four filters are completely determined by the low-pass analysis filter $H_0(z)$ only. By using (4) and (5), the expression for the alias free reconstructed signal can be written as:

$$\hat{X}(z) = \frac{1}{2}\left[H_0^2(z) - H_1^2(z)\right]X(z) = \\ = \frac{1}{2}\left[H_0^2(z) - H_0^2(-z)\right]X(z) = H(z)X(z) \tag{6}$$

In this way, the whole filter bank is completely defined by a single low-pass filter $H_0(z)$. Additionally, if $H_0(z)$ is FIR and has linear phase, then overall transfer function $H(z)$ will have linear phase, so phase distortion is eliminated. This means that $h_0[n]$ is symmetric. That is, $h_0[n] = h_0[N - n]$ for a filter of order N. QMF bank with linear phase filters can be implemented efficiently by using polyphase decomposition.

B. *Noble identities*

When we try to use polyphase representation different types of cascade interconnection between filter and decimators/interpolators are needed. If the filter transfer function $H(z)$ is rational then it can be redrawn as it is shown in Fig. 4 and Fig. 5. These structures are known as Noble identities [13] and are very useful in theory and implementation of multirate systems.

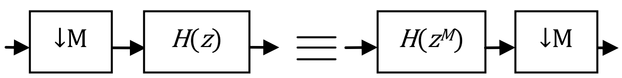

Fig. 4 Noble identity for decimation

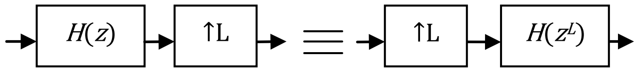

Fig. 5 Noble identity for interpolation

C. *Polyphase representation*

The polyphase representation is an important advancement in multirate signal processing which leads to computationally efficient implementation of decimation and interpolation filters, as well as filter banks. The basic idea is given in addition:

Let $H(z)$ be a symmetric filter with a linear phase.

$$H(z) = \sum_{n=-\infty}^{\infty} h[n]z^{-n} \tag{7}$$

By separating the even numbered coefficients of $h[n]$ from the odd ones, $H(z)$ can be written as

$$H(z) = \sum_{n=-\infty}^{\infty} h[2n]z^{-2n} + z^{-1}\sum_{n=-\infty}^{\infty} h[2n+1]z^{-2n} \tag{8}$$

Defining

$$E_0(z) = \sum_{n=-\infty}^{\infty} h[2n]z^{-n} \\ E_1(z) = \sum_{n=-\infty}^{\infty} h[2n+1]z^{-n} \tag{9}$$

$H(z)$ can be written as:

$$H(z) = E_0(z^2) + z^{-1}E_1(z^2) \tag{10}$$

Consider the Noble identities for decimation filter shown in Fig. 4 with $M = 2$ and using the equation (10), Fig. 6(*a*) can be redrawn as in Fig. 6(*b*). This implementation is more efficient than a direct implementation of $H(z)$. The use of polyphase decomposition enables rearranging the computations of the filtering operation.

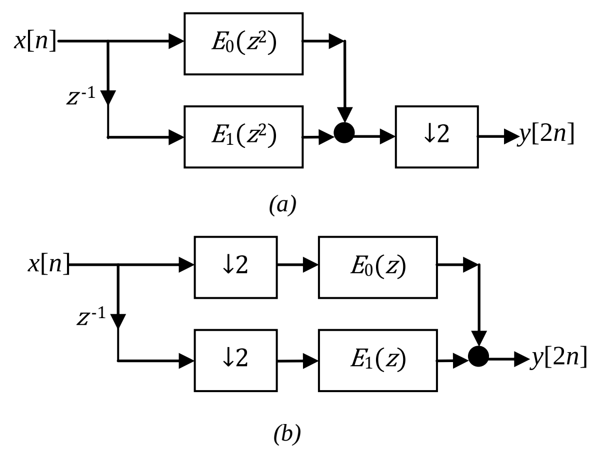

Fig. 6 Polyphase decomposition

If direct form implementation is used for decimation filter, then only the even numbered output samples are computed and that requires (N+1) multiplications per unit time (MPUs) and N additions per unit time (APUs). However, during the computation of odd numbered output samples, the structure is simply resting. If polyphase implementation is used, then the computation of O/P samples requires only (N+1)/2 MPUs and N/2 APUs [13]. Thus polyphase representation of decimation filter reduces the computational complexity of the filter bank. For complete implementation of two-channel QMF bank using polyphase framework, a total of only N MPUs and N APUs (for one sample of the input signal) are required [13] where N is the length of prototype low-pass filter $H_0(z)$.

IV. Experiments and Results

In this paper algorithms for harmonic analysis based on direct decomposition form and polyphase decomposition form are implemented. We used 50Hz input waveforms sampled at fs_1=1600Hz and fs_2=6400Hz. Several tests were performed on different stationary signals. Here we present results obtained for an arbitrary chosen signals that contain all odd harmonics up to the 15th harmonic. The signals are the following:

$$v(t) = \text{sqrt}(2)\,[\,250.0209\sin(\omega_0 t) + 4.1251(3\omega_0 t - 90^0) + 0.3671\sin(5\omega_0 t + 179^0) + 3.2021\sin(7\omega_0 t\ 114^0) + 8.5781\sin(9\omega_0 t - 173^0) + 1.2571\sin(11\omega_0 t - 124^0) + 1.9895\sin(13\omega_0 t - 28^0) + 1.1064\sin(15\omega_0 t + 177^0)]$$

$$i(t) = \text{sqrt}(2)[3.6558\sin(\omega_0 t) + 2.7805\sin(3\omega_0 t - 173^0) + 1.7790\sin(5\,\omega_0 t - 1^0) + 0.8140\sin(7\omega_0 t - 178^0) + 0.0381\sin(9\omega_0 t - 92^0) + 0.2494\sin(11\omega_0 t - 9^0) + 0.2537\sin(13\omega_0 t + 178^0) + 0.1109\sin(15\omega_0 t + 5^0)]$$

with $\omega_0 = 2\pi f$, f = 50 Hz.. They are shown on Fig. 7 and Fig. 8 respectively.

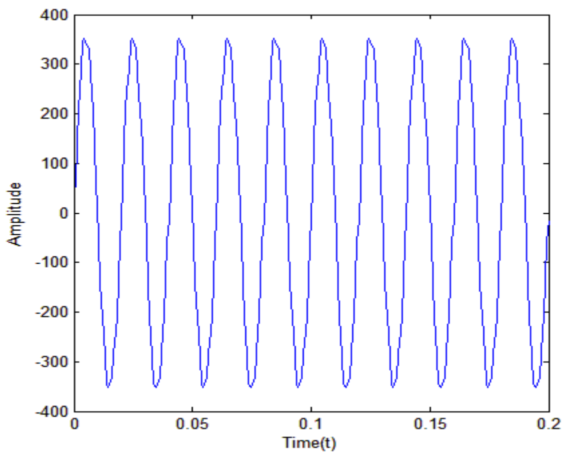

Fig. 7 Voltage signal

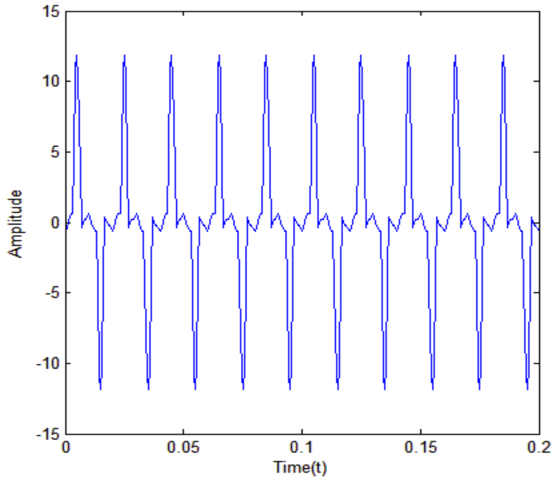

Fig. 8 Current signal

Additionally to well-known wavelet filters for harmonics measurement (db20, coif5, and v24) the Johnston's filters with 32 coefficients (J32 C, J32 D,J32 E) are used [12], [15]. Coefficients for different Johnston's filters together with the design parameters are given in [16]. Letter C is a notation for all Johnston's filters that have normalized transition band of 0.0625, D refers to normalized transition band of 0.043 and E refers to 0.023.

A. Input signal extension

In order to investigate the influence of the input signal extension on the accuracy of wavelet transform results Wavelet packet transform-based algorithm given in [4] is implemented for the two given stationary waveforms sampled at 1600Hz. The implementation is made in MATLAB. Extension types that were applied on the both sides of the signal (bilateral padding) are the following:

- Periodic extension ('per') - signal is periodically extended outside the original support;
- Symmetric whole-point extension ('symw') - symmetric edge replication starting from the second edge value;
- Zero-padding extension ('zpd') - signal is zero outside the original support.
- Symmetric half-point extension ('sym') – symmetric edge replication starting from the first edge value.

The both types of symmetric extension can be used only if the corresponding filter is symmetric and has a linear phase, while the periodic and zero-padding extension can be used with any type of filters. In the case of image border extension it was shown that symmetric extension performs better than the periodic extension. Its main advantage is that it may introduce a corner but it doesn't introduce a jump [14]. The influence of the signal extension on the harmonics calculations in case of stationary signals is shown in Table I and Table II.

TABLE I. RESULTS FOR DIFFERENT TYPES OF EXTENSION FOR INPUT SIGNAL V(T)

	J32 C				J32 D				db20	
	ZPD	**PER**	SYMW	SYM	ZPD	**PER**	SYMW	SYM	ZPD	**PER**
50 Hz	249.7689	**249.9890**	250.4234	249.6009	249.1206	249.1396	**249.9993**	249.1715	231.1948	**250.0066**
150 Hz	7.9220	**4.3183**	22.9342	14.0404	7.5086	**3.8946**	22.6146	13.1259	14.4012	**4.7968**
250 Hz	1.9850	**0.3363**	6.9836	2.8495	1.9086	**0.3701**	7.0518	2.6863	6.5380	**1.1195**
350 Hz	2.3505	2.1004	3.9825	**3.6612**	**3.1608**	2.9686	4.4844	4.3236	**4.4211**	4.6170
450 Hz	**8.8612**	8.9311	9.3198	8.9902	**8.6412**	8.6998	9.1193	8.7542	7.5657	**7.9090**
550 Hz	1.2315	1.2298	1.4938	**1.2328**	**1.2601**	1.2705	1.4573	1.2491	1.0971	**1.1976**
650 Hz	1.0951	**1.1117**	1.1807	1.1948	1.0872	**1.1026**	1.1123	1.2045	1.1576	**1.1549**
750 Hz	2.0117	**1.9892**	2.1296	2.1909	2.1203	**1.9842**	2.5521	2.5661	1.8514	**1.9986**

TABLE II. RESULTS FOR DIFFERENT TYPES OF EXTENSION FOR INPUT SIGNAL I(T)

	J32 C				J32 D				db20	
	ZPD	**PER**	SYMW	SYM	ZPD	**PER**	SYMW	SYM	ZPD	**PER**
50 Hz	3.6464	**3.6576**	3.6603	3.6448	3.6491	**3.6601**	3.6652	3.6482	3.4097	**3.6837**
150 Hz	2.7708	**2.7736**	2.8535	2.7731	2.7796	**2.7823**	2.8698	2.7838	2.5521	**2.6450**
250 Hz	1.7911	**1.7835**	1.8323	1.8144	1.7656	1.7599	1.8141	**1.7852**	**1.8309**	1.9260
350 Hz	0.7856	0.7855	**0.8131**	0.7786	0.7993	**0.7994**	0.8291	0.7945	0.6945	**0.7191**
450 Hz	0.2123	**0.2098**	0.2158	0.2150	0.1542	**0.1496**	0.1597	0.1600	**0.3483**	0.3673
550 Hz	0.2488	**0.2494**	0.2517	0.2516	0.2464	0.2468	0.2510	**0.2485**	0.2275	**0.2441**
650 Hz	0.2526	**0.2533**	0.2574	0.2505	**0.2541**	0.2549	0.2597	0.2525	0.2459	**0.2548**
750 Hz	0.1107	0.1112	**0.1109**	0.1119	**0.1111**	0.1114	0.1115	0.1122	0.1042	**0.1116**

The results are obtained after applying 3-level WPT decomposition using previously mentioned filters. Johnston filters are used for all four type of input signal extension. Daubechies filter 'db20' is used only for periodic and zero-padding extension because it is asymmetric. For applying polyphase decomposition using Johnston filters only the periodic extension of the input signal can be used. That is because with separating the even numbered filter coefficients from the odd ones, the obtained filters become asymmetric. In the tables the bold-marked values are the closest to the exact values. It is evidently that in the case of stationary waveform the periodic extension gives the best results.

B. *Direct vs. polyphase implementation efficency*

It was mentioned above that the limitation of DWT for measurement of RMS value of different harmonic components can be overcome using the WPT. With the use of the Noble identities WPT can be implemented with direct or polyphase form. In this paper we investigate the efficiency of the polyphase decomposition against the direct decomposition using Johnston filters J32C, J32D and J32E for periodic input signal extension on each level. The efficiency is the same for the all of the three filters, because they have the same length. For that purpose the both forms of decomposition are implemented on the floating-point signal processor TMS320C6713 (225MHz). Towards the analysis one of the previously given signal is used. The signal consists of 10 periods with 32 samples in each period for sampling frequency f_{s1}=1600Hz and 10 periods with 128 samples in each period for sampling frequency f_{s2}=6400Hz.

In addition theoretical calculations of the efficiency and execution time of the processor are given.

a) Theoretically calculated efficency through the number of multiplications and additions

In the shown block-diagrams the following nomenclature is used:

- N - Length of the filter
- L - Length of input signal

In this case N is equal to 32 coefficients and L is equal to 320 samples that makes 10 periods from the fundamental signal. The calculation of the efficiency for the direct form, for one level of decomposition is the following:

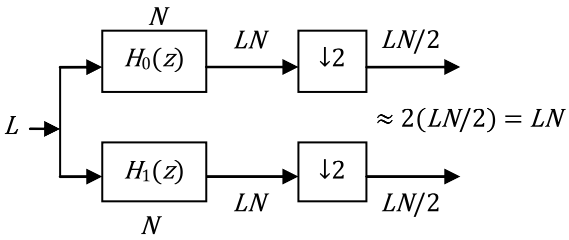

Fig. 9 Block-diagram for calculation of the number of instructions for direct decomposition form

That gives 10240 multiplication and additions per level. For 3 levels of direct decomposition their number is 40960.

The calculation of the efficiency for the polyphase form, for one level of decomposition is the following:

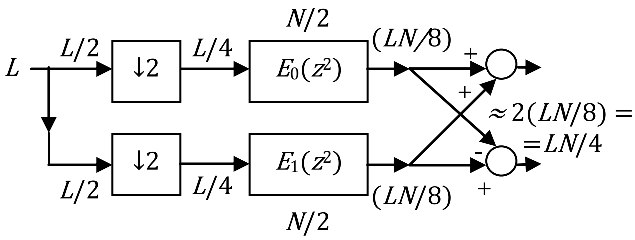

Fig. 10 Block-diagram for calculation of the number of instructions for polyphase decomposition form

That gives 2560 multiplications and additions per level. To that number are added 160 additions and subtractions (together noted as additions) needed for obtaining polyphase outputs and that gives 2560 multiplication and 2720 additions.

For 3 levels of polyphase decomposition their number is 10240 multiplications and 10880 additions.

From the obtained results is evident that the polyphase decomposition form is almost 4 times more efficient than the direct decomposition form.

b) Direct and polyphase implementation on TMS320C6713 for f_{s1}=1600Hz using SIMULINK

The implementation of direct wavelet decomposition and polyphase wavelet decomposition on signal processor for f_{s1}=1600Hz is made in blocks using SIMULINK environment [17]. In SIMULINK a function-block for direct wavelet decomposition already exists, but is not used because it works only with zero-padding extension. Hence, we have implemented two models for wavelet decomposition using subsystems so that arbitrary chosen signal extension can be used. Every level has 2^{L-1} subsystems, where L is the number of the corresponding level. They are consisted of embedded MATLAB function-blocks for extension and decimation, as well as convolution blocks. In this model a periodic extension is used. In Fig. 11 and Fig. 12 is shown the content of these subsystems for direct and polyphase decomposition, respectively.

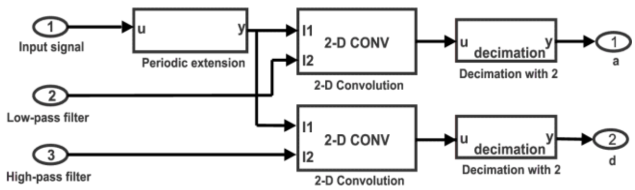

Fig. 11 SIMULNIK model for direct wavelet decomposition

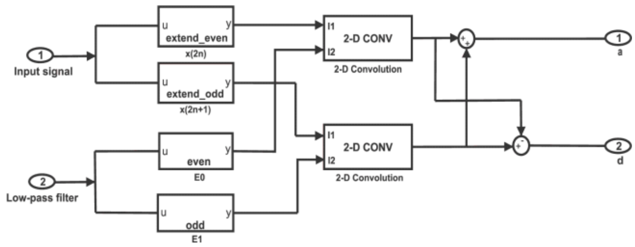

Fig. 12 SIMLUNIK model for polyphase wavelet decomposition

For the purpose of our investigation, measurement of calculation times and instruction cycles for all of the 3 levels of decomposition is made. The results when Johnston filters with 32 coefficients are used in case of polyphase and direct implementation are given in Table III.

TABLE III. EXECUTION TIME OF WPT COMPUTATIONS FOR POLYPHASE DECOMPOSITION AND 320 SAMPLES FROM THE INPUT SIGNAL

J 32 D,C,E				
	Direct		**Polyphase**	
Level	**Cycles**	**Time [ms]**	**Cycles**	**Time [ms]**
1	183.690	0,82	63.833	0,28
2	427.950	1,90	106.245	0,47
3	618.975	2,75	200.363	0,81

For better comparison of the efficiency in addition in Table IV are given the results when Daubechies filter with 20 coefficients (db20) and Vaidyanathan filter with 24 coefficients (v24) are used.

TABLE IV. EXECUTION TIME OF WPT COMPUTATIONS FOR DIRECT DECOMPOSITION AND 320 SAMPLES FROM THE INPUT SIGNAL

	DB20		**V24**	
Level	**Cycles**	**Time [ms]**	**Cycles**	**Time [ms]**
1	81.743	0,36	110.250	0,49
2	163.733	0,73	239.400	1,06
3	267.075	1,19	381.600	1,70

The only drawback of this implementation is that the use of SIMULINK allows only 3-levels of decomposition. That is a result from the fact that during the generation of the code for each level of decomposition new variables are declared, which brings to memory overload. Hence, for decomposition in more than 3-levels, the same algorithm was implemented directly in Code Composer Studio.

c) Direct and polyphase implementation on TMS320C6713 for f_{s1}=1600Hz in Code Composer Studio

For implementation of direct and polyphase decomposition in 3-levels directly in Code Composer Studio, Johnston filters with 32 coefficients are used. In addition, Table V, are given the results for instruction cycles and calculation times when periodic input signal extension is used.

TABLE V. EXECUTION TIME OF WPT COMPUTATIONS FOR POLYPHASE DECOMPOSITION AND 320 SAMPLES FROM THE INPUT SIGNAL

J 32 D,C,E				
	Direct		**Polyphase**	
Level	**Cycles**	**Time [ms]**	**Cycles**	**Time [ms]**
1	189.643	0,85	77.632	0,35
2	466.157	2,08	143.734	0,64
3	745.411	3,32	209.281	0,93

In the following table, Table VI, are given the results when Daubechies filter with 20 coefficients (db20) and Vaidyanathan filter with 24 coefficients (v24) are used.

TABLE VI. EXECUTION TIME OF WPT COMPUTATIONS FOR DIRECT DECOMPOSITION AND 320 SAMPLES FROM THE INPUT SIGNAL

	DB20		**V24**	
Level	**Cycles**	**Time [ms]**	**Cycles**	**Time [ms]**
1	146.371	0,65	163.471	0,73
2	365.954	1,63	404.373	1,80
3	592.349	2,64	647.598	2,88

It is evident that the calculation times and the number of instruction cycles are smaller when SIMULINK implementation is used. That is due to declaring new variables in the SIMULINK model, which in the new code is solved by using several “for cycles”. Thus, values are transferred from one variable to another which increases the number of instruction cycles and results in calculation time increasing.

d) Direct and polyphase implementation on TMS320C6713 for f_{s2}=6400Hz in Code Composer Studio

The implementation of direct wavelet decomposition and polyphase wavelet decomposition on the signal processor for sampling frequency f_{s2}=6400Hz is made directly in Code Composer Studio. For periodic input signal extension and 5-levels of WPT decomposition the calculation times and instruction cycles given in Table VII are obtained.

TABLE VII. EXECUTION TIME OF WPT COMPUTATIONS FOR POLYPHASE DECOMPOSITION AND 1280 SAMPLES FROM THE INPUT SIGNAL

J 32 D,C,E				
	Direct		**Polyphase**	
Level	**Cycles**	**Time [ms]**	**Cycles**	**Time [ms]**
1	773.137	3,44	290.604	1,30
2	1.872.665	8,33	566.900	2,52
3	2.934.431	13,05	842.425	3,75
4	4.012.013	17,84	1.111.997	4,95
5	5.102.583	22,68	1.384.204	6,16

In Table VIII are given the results when Daubechies filter with 20 coefficients (db20) and Vaidyanathan filter with 24 coefficients (v24) are used.

TABLE VIII. EXECUTION TIME OF WPT COMPUTATIONS FOR DIRECT DECOMPOSITION AND 1280 SAMPLES FROM THE INPUT SIGNAL

	DB20		**V24**	
Level	**Cycles**	**Time [ms]**	**Cycles**	**Time [ms]**
1	555.940	2,47	628.013	2,80
2	1.425.108	6,34	1.568.994	6,98
3	2.309.925	10,27	2.526.459	11,23
4	3.182.848	14,15	3.469.338	15,42
5	4.065.357	18,07	4.427.770	19,68

The difference in time when polyphase decomposition is used is more than obvious in both implementations. The ratio between direct and polyphase results is close to the one that was theoretically obtained. The duration of the execution time for the fifth level of polyphase decomposition, implemented by applying "Johnston's" filters with 32 coefficients, is even shorter than the second level of direct decomposition implemented by applying Johnston's filters with 32 coefficients as well as the other proposed filters. With that the efficiency of the polyphase implementation over the direct implementation is shown.

V. CONCLUSION

In this work, at first the influence of the different signal extension was investigated. Experimental investigation performed with the use of Johnston’s J32 C, J32 D and J32 D filters showed that the periodic extension is the most adequate in case of stationary signals. According to that conclusion both direct and polyphase forms of wavelet packet were implemented on a signal processor for two different sampling frequencies applied on the input signal. For the purpose of the efficiency investigation, measurement of calculation times and instruction cycles of the processor were made. Thereby the efficiency of QMF bank with linear phase filters, using polyphase decomposition, for the aim of harmonic analysis was shown.

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