Multi-Resolution PR NMDFBs for Programmable Variable Bandwidth Filter in Wideband Digital Transceivers

fredharris[a], *Behrokh Farzad*[b], *ElettraVenosa*[c], *Xiaofei Chen*[d]
San Diego State University
San Diego, USA
[a]fharris@mail.sdsu.edu, [b]farzad@rohan.sdsu.edu. [c]elettra.venosa@unina2.it, [d]chenxiaofei_sdsu@yahoo.com

***Abstract*—This paper describes a novel application of perfect reconstruction (PR) non-maximally decimated filter banks (NMDFBs). PR non-maximally decimated analysis/synthesis chains are used for performing resampling, channelization and filtering in very wideband software defined radios. Digital wideband filtering is currently limited by the hardware clock speed which constrains the maximum signal bandwidth that can be processed digitally. Polyphase channelizers decrease the sample rate of the input signals and parallelize the wideband filtering process at cost of a single filter. In this paper we show how to assemble multiple-tier PR analysis/synthesis channelizer chains for implementing, in a very efficient way, variable bandwidth digital filters and multi-resolution systems operating on very wideband signals. When an internal tier analysis channelizer is applied to a single output channel of a first external tier analysis channelizer it decomposes this channel in multiple narrower sub-channels and, by applying appropriate complex processing elements to those sub-channels we can modify the designed filter bandwidth as desired.**

***Index Terms*—Non-maximally decimated filter bank, perfect reconstruction, polyphase channelizer, digital receiver, variable bandwidth filter.**

I. INTRODUCTION

Wideband signal processing capability is strongly required in software radios (SRs). Currently, the hardware capabilities set the limits of the input bandwidth that can be processed digitally. Data must be sampled and processed at least at two samples per symbol. Analog-to-digital converters (ADCs), could offer sample rates which are of the order of a few Gsps. However, at such high speeds even the simplest standard digital filtering task can saturate the hardware processing limits and this prevents the implementation of wideband digital systems. We might be able to sample the analog signal appropriately but, currently, we are unable to process it. PR NMDFBs offer the capability of processing/filtering wideband signals while keeping the hardware operating at lower processing rate [1,2], [4].

A PR NMDFB chain is composed by an analysis filter bank (AFB) and a synthesis filter bank (SFB) whose composite response of their low pass prototype filter (LPPF) is a Nyquist pulse. By applying intermediate processing elements, which are complex vectors, between AFB and the SFB any spectral shape (filter frequency response) can be well approximated [4] in the channelized domain and many fundamental tasks of a digital receiver can be efficiently performed.

In this paper we focus on the capability of PR NMDFBs to synthesize variable bandwidth filters. Generally, to approximate a wideband low-pass filter frequency response in the channelized domain the intermediate processing elements between the AFB and the SFB will have to be mostly zeroes (in the stop-band) and ones (in the pass-band). The wideband filtering capability derives from the possibility of enabling and disabling the analysis channelizer channels as desired. The filter pass-band is defined by the number of enabled channels which can be dynamically changed as desired. When a single intermediate processing element is set to zero the corresponding analysis output channel is nullified (filtered out) and it does not proceed further in the processing chain. Once the first channelizer has been designed the width and the number of the channels acting on the input bandwidth is fixed. The filter pass-band is given by the number of channels which are enabled over the total number of channels available (number of paths) and it can change only of an integer multiple of the channel width. If, for example, the two external channels, which correspond to the filter transition bands are disabled then the designed filter bandwidth becomes two channels narrower.

If we apply two smaller PR analysis/synthesis channelizers to the analysis output channels which correspond to the filter transition bands, then the designed filter bandwidth acquires the capability of being adjusted much finely since it can now be reduced by integer multiples of the second tier sub-channels. The smaller tier analysis/synthesis chains could be applied to the output of any first tier channelizer channels. This is the strategy used for designing multi-resolution digital wideband systems.

This paper is organized in five main sections. In section 2 basics on non-maximally decimated up and down converter channelizers are provided. In section 3 the proposed architecture is presented. Section 4 provides simulation results demonstrating the correct functionality of the proposed architecture. Section 5 provides the conclusions and suggestions for future developments.

II. Background

Figure 1 depicts the high level block diagram of a non-maximally decimated analysis channelizer. This engine performs the sample rate change from the input rate f_S to the output rate $2f_S$ /M and it offers the advantage to avoid the spectral folding at the channel band edge which is critical in the wideband signal scenario. The authors derived the form of the M-to-2 down sampler channelizer in an earlier paper [10].

The input data buffer performs the correct data loading of the M/2 input samples into the M-path filter while the circular output buffer performs the time alignment of the shifting time origin of the input samples in the M-path filter with the non-shifting time origin of the IFFT phase rotators. The M/2 time sample shift of the input time series redefines the time origin and causes sinusoids with an odd number of cycles in the length M array to alternate sign on successive shifts. The alternating sign occurs because the odd indexed frequency bins alias to the half sample rate while the even indexed frequency bins alias to DC. Rather than reversing the phase of alternate output samples from the odd indexed bins we perform an M/2 point circular shift of alternate M length vectors before presenting them to the IFFT. The circular shift applies the correct phase alignment to all frequencies simultaneously. Of course, there is not a circular shift of the buffer but rather an alternate data load into the IFFT input buffer.

Figure 2 depicts the high level block diagram of a non-maximally decimated synthesis channelizer. This engine performs the dual tasks of the analysis down converter performing the sample rate change from the input rate 2fs to the output rate Mfs. The choice to use a 2-to-M up sampler offers the advantage that it permits the input signal to be oversampled by 2 and avoids the difficulty of having the sample rate precisely match two sided bandwidth of the input signals as well as permitting a shorter length prototype filter due to an allowable wider transition bandwidth. The authors derived the form of the 2-to-M up sampler channelizer in an earlier paper [10].

The M-point IFFT applies the complex phase rotation to the separate baseband input signals as well as performs the initial 1-to-M up sampling of the input samples. The circular buffer, following the IFFT, performs the correct data offset of the two M/2 point halves of the IFFT output vector to maintain phase alignment with the channelizer output vector. The complex sinusoids output by the IFFT always defines its time origin as the initial sample of its output vector. The output of the polyphase filter exhibits a time origin that shifts due to the M/2 time sample shift embedded in the output commutator. The M/2 time sample shift of the output time series causes sinusoids with an odd number of cycles in the length M array to alternate sign on successive shifts. The alternating sign is the reason that the odd indexed frequency bins up convert to a frequency k+N/2 rather than frequency index k. Rather than reverse phase alternate input samples to the odd indexed IFFT bins we perform an M/2 point circular shift of alternate M-length vectors from the IFFT. The circular shift applies the correct phase alignment to all frequencies simultaneously. In the real implementation, there is not a circular shift of the buffer but rather an alternate data load from the IFFT output buffer into the polyphase filter.

When those engines are cascaded (analysis/synthesis) and complex processing elements are placed between them, we acquire the capability of overcoming the hardware clock limits. In fact, at the output of the analysis channelizer the input sample rate is reduced and we have access to all the channels simultaneously. Here the required signal processing is applied via intermediate processing elements that change gain and phase of each channel. After the desired filtering task has been performed, the channels are recomposed and the time series is upsampled, as required, by the synthesis channelizer.

To maintain the perfect reconstruction property, the composite response of the analysis and synthesis filter bank has to be a Nyquist pulse.

III. Proposed Architecture

Figure 3 describes the high level block diagram of the proposed architecture which is composed of two-tier PR channelizers. Details on the external and internal analysis and synthesis channelizers have been depicted in figures 1 and 2. The external tier is a 128 paths PR analysis/synthesis chain which performs 128:2 down sampling of the input time series.

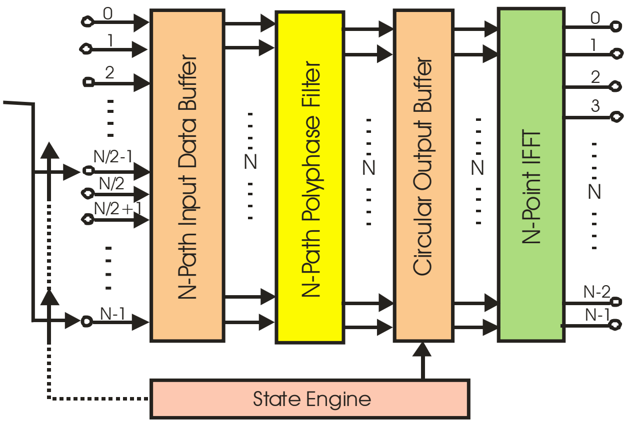

Figure 1: High Level Block Diagram of a Non-Maximally Decimated Polyphase Analysis (Down Converter) Channelizer.

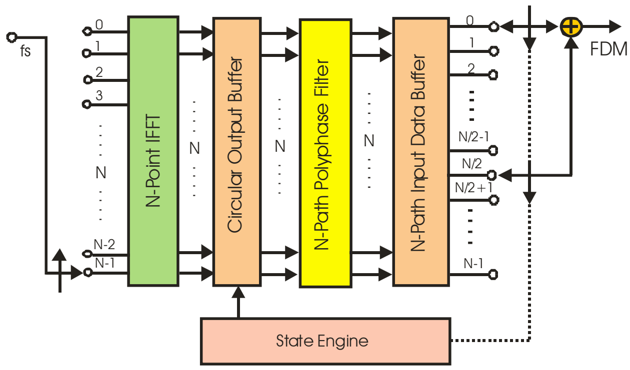

Figure 2: High Level Block Diagram of a Non-Maximally Decimated Polyphase Analysis (Down Converter) Channelizer.

The low pass prototype filter in the analysis is a windowed sinc function while the synthesis low-pass filter is designed with the Remez algorithm and its pass-band covers entirely the analysis low-pass prototype filter. This is a standard strategy for achieving the PR property. As mentioned before, the perfect reconstruction property is verified as long as the composite response of the analysis/synthesis low-pass prototype filters is a Nyquist pulse. And this is guaranteed when the synthesis prototype filter does not affect the spectral shape of the analysis prototype filter.

Two smaller PR chains are applied at the output of the external tier analysis channelizer. These will act on the two channels corresponding to the wideband filter transition bands which in our case are channels +/- 25 of the external tier analysis channelizer. For convenience, the number of paths that we selected for those engines is 10. However a different choice could have been made. The size of those smaller channelizers determines how finely we want to be able to modify the filter bandwidth. When a larger number of path is selected more channels are available to us for obtaining the desired bandwidth. However when the channelizer size increases also its workload increases slightly. The selected small analysis channelizers perform a decimation 10:2 of the input time series and the low pass prototype filters are selected by using the same strategy as in the external PR chain. The analysis low pass prototype is a windowed sinc while the synthesis low pass prototype is a standard digital filter designed with the Remez algorithm.

Notice that the internal tier channelizers don't necessarily have to be used for reducing or increasing the filter bandwidth. Once they are applied to the external channels, corresponding to the band edge of the wideband filter, it is possible to apply any gain and desired phase to the sub-channels via intermediate processing elements, thus the second tier channelizers could be used to shape differently the transition bandwidth of the wideband filter. As an example, they could be used for sharpening the transition bands of the wideband filter instead of reducing its bandwidth. The internal tier channelizers provide us the capability of modifying the wideband filter in any desired way while keeping the workload at a minimum and a reasonable latency.

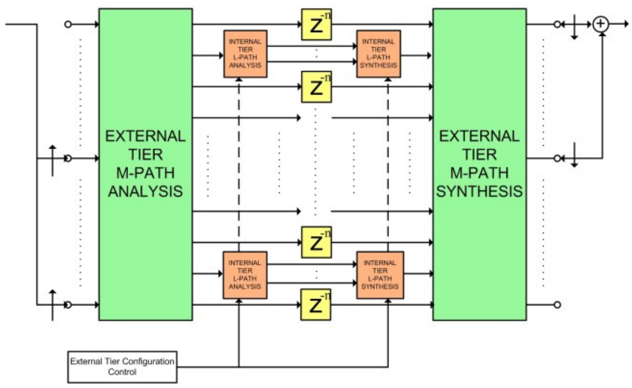

Figure 3: High Level Block Diagram of Proposed Two Tiers Channelizer Based Variable Bandwidth Filter.

Notice also that the internal tier channelizers could be applied at the output of any channel of the external tier channelizer. This strategy is usually used for building multi-resolution systems in scenarios with wide input spectra containing narrow-band signals of interest. In these cases the external channelizer channelizes the entire input spectrum. An energy detector establishes which channels contain signals and which do not and then the internal tier channelizers gets applied to the channels of interest.

The same resolution could be obtained with a single tier PR channelizer chain having a number of paths which is equal to the product of the number of paths of the two tiers. However this will maximize the delay of the system. Also it is not smart to pay for high resolution in spectral areas in which we don't need it. Remember that increasing the number of channelizer paths increases the amount of hardware resources used.

IV. SIMULATION RESULTS

In this section we show some simulation results demonstrating the correct functionality of the proposed architecture.

We designed an external tier PR channelizer which has 128 paths. The desired filter pass-band has been designed to cover less than half available spectrum (its transition bandwidths correspond to channels -/+ 15). This is a very reasonable choice for communication systems. The designed Nyquist filter in the analysis channelizer is a windowed sinc function and it is 1791 taps long. We include an additional zero in the front and partition this filter on the available paths to achieve exactly 14 taps per path. The synthesis filter for the corresponding synthesis channelizer is designed by using Remez algorithm and it is again 1791 taps long. An additional zero tap is included here before reshaping the filter over 128 paths. The pass-band of this filter has been designed to cover completely the analysis low-pass prototype filter. As mentioned before, this allows to preserve the Nyquist shape of the composite filter response and to satisfy the perfect reconstruction requirements.

For the smaller 10 paths channelizer the same strategy is used. The length of the low-pass prototype filters is 139 taps for both the analysis and synthesis engines. One zero tap is appended to the filter length and 14 taps per path are obtained. The frequency response of both the analysis and synthesis filters are plotted in Figure 4 on a normalized frequency axes. The red curve shows the frequency response of the synthesis low-pass filter while the blue curve is the frequency response of the analysis low pass prototype filter.

Figure 5 shows how the pass-band of the wideband filter can be changed with the PR channelizers we designed. In each subplot, the red spectrum shows how much we can expand or reduce the original band which is depicted in blue. This selectable bandwidth is programmable by the user. If more bandwidth enlargement or reduction is needed, the second tier channelizers could be designed to have a larger number of channels. In case in which no fine tuning is needed we might just decide to enlarge the width of the 10 paths channelizer. The subplots in Figure 5 show the original band being reduced by one sub-channel at time. Of course we can select to reduce the original bandwidth by multiple sub-channels in a single step. For visualizing the results better, all the subplots in Figure 5 show a zoomed version of the transition band of the variable wideband filter over the positive frequency axis.

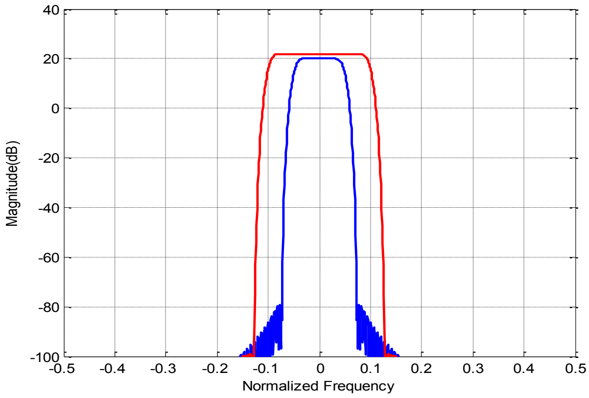

Figure 4: Spectra of Analysis Low-Pass Prototype Filter (in Blue) and Synthesis Low-Pass Prototype Filter (in Red).

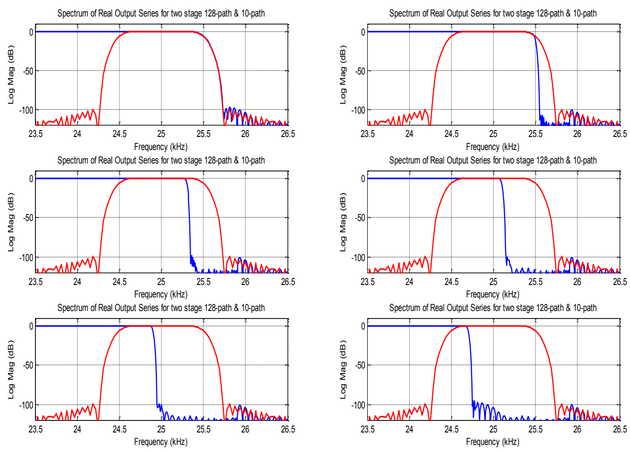

Figure 5: Frequency Responses (Positive Frequencies), Zoomed over the Transition Bands, of the Reduced Bandwidth Filter.

Figure 6 shows, in the upper subplot, the time response of the real part of the filter after its bandwidth has been reduced by one sub-channel while the lower subplot shows the frequency response of the same filter with a zoomed view on its transition bands.

The zoom simplifies visualization of the bandwidth reduction process.

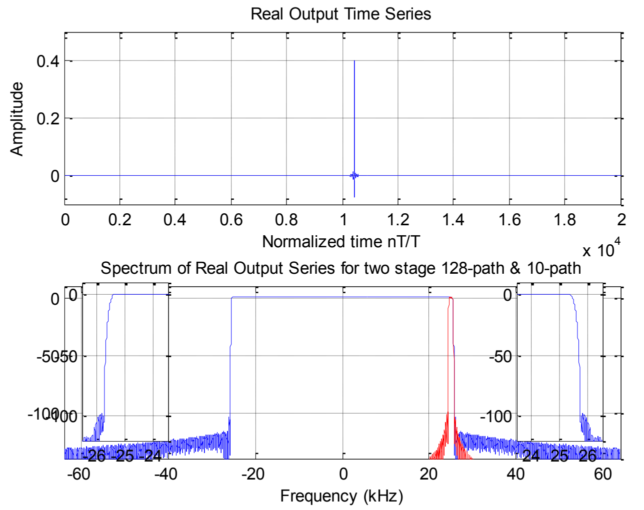

Figure 6: Time Response (Upper Subplot) and Frequency response (Lower Subplot) of the Filter Bandwidth Reduced by One Sub-Channel.

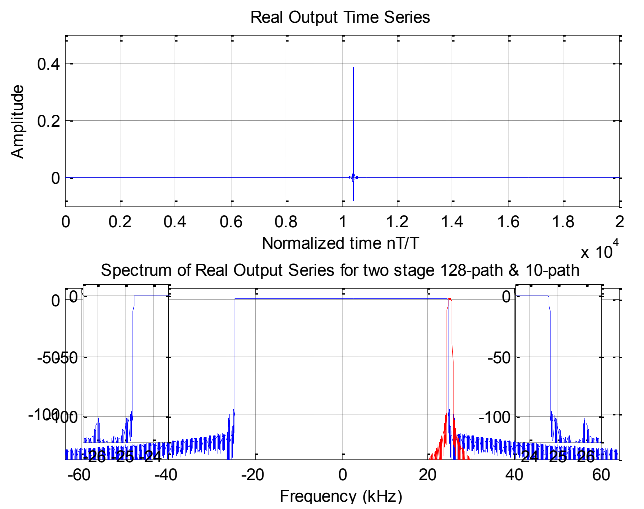

Figure 7: Time Response (Upper Subplot) and Frequency response (Lower Subplot) of the Filter Bandwidth Reduced by Ten Sub-Channels.

Figure 7 shows the same subplots as Figure 6 when the original filter bandwidth has been reduced by all the ten sub-channels available. In particular, in the upper subplot we can see the time response of the reduced filter bandwidth while in the lower subplot its frequency response is shown. As it can be seen from the zoomed plots superimposed on the same figure, and comparing Figures 6 and 7, we have been able to reduce the filter bandwidth by about one kHz.

V. CONCLUSIONS

In this paper we have presented an efficient architecture which utilizes PR non-maximally decimated polyphase filter banks to build variable bandwidth filters. The proposed architecture can be embedded in the new generation of very wideband digital receivers which are also PR NMDFB based [4]. When the internal tier channelizers are applied at specific locations in the filter pass-band, this same architecture can be used to build efficient multi-resolution wideband software defined radios. We recall here that having two tier of channelizers instead of a single channelizer with a larger number of channels contributes to reducing the latency of the system while increasing its efficacy.

REFERENCES

[1] Harris, F.; Dick, C.; Chen, X.; Venosa, E., "Wideband 160-channel polyphase filter bank cable TV channeliser," *Signal Processing, IET* , vol. 5, no. 3, pp. 325,332, June 2011.

[2] f.j. harris, E. Venosa, X. Chen and B.D. Rao, "Polyphase Analysis Filter Bank Down-Converts Unequal Channel Bandwidths with Arbitrary Center Frequencies," *Analog Integr Circ Sig Process*, vol. 71, no. 3, pp. 481-494, 2012.

[3] T. Karp and N. J. Fliege, "Modified DFT filter banks with perfect reconstruction", *IEEE Trans. Circuits Syst. II, Anal. Digital Signal Process.*, vol. 46, no. 11, pp. 1404 -1414, Dec. 1999.

[4] Chen, X.; Harris, F.J.; Venosa, E.; Rao, B.D., "Non-Maximally Decimated Analysis/Synthesis Filter Banks: Applications in Wideband Digital Filtering," *Signal Processing, IEEE Transactions on* , vol. 62, no. 4, pp. 852, 867, Feb. 15, 2014.

[5] f.j. harris, "*Multirate Signal Processing for Communication systems*", Prentice Hall, Upper Saddle River, New Jersey 07458, 2004.

[6] Vaidyanathan, P.P., "Theory and design of M-channel maximally decimated quadrature mirror filters with arbitrary M, having the perfect-reconstruction property," *Acoustics, Speech and Signal Processing, IEEE Transactions on* , vol. 35, no. 4, pp. 476, 492, Apr 1987.

[7] Vetterli, M.; Le Gall, D., "Perfect reconstruction FIR filter banks: some properties and factorizations," *Acoustics, Speech and Signal Processing, IEEE Transactions on* , vol. 37, no. 7, pp. 1057, 1071, Jul 1989.

[8] Jain, V.K.; Crochiere, R.E., "A novel approach to the design of analysis/Synthesis filter banks," *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP '83.* , vol. 8, no. 3, pp. 228, 231, Apr 1983.

[9] E. Venosa, f. harris and F. Palmieri, *On Software Radio: Sampling Rate Selection, Design and Synchronization*, Springer Science + Business Media, LLC 233 Spring Street, New York, NY 10013, USA.

[10] f. harris, C. Dick, X. Chen, and E. Venosa, "M-Path Channelizer with Arbitrary Center Frequency Assignments", WPMC 2010, Oct 11, 2010 - Oct 14, 2010, Recife, Brazil.