

Multi-rate Analysis and Synthesis Systems incorporating Modifications using a Highly Configurable WOLA Coprocessor

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Filterbank (multi-rate) analysis and synthesis strategies prove advantageous in many signal processing areas operating as a divide and conquer strategy tackling difficult problems into an equivalent series of much simpler problems. For example, large convolutional systems encountered in applications such as echo cancellation and feedback cancellation may require a large number of filter taps. Using the filterbank technique, it may equivalently be implemented as a parallel combination of much shorter subband filters. When properly designed, the filterbank subband signals are minimally overlapping in frequency yielding signals that are approximately orthogonal to each other. Lately, digital filterbank techniques, with their great precision, have enabled many strategies to be implemented that were difficult or impractical with analog structures. Accordingly, much theory has been developed including the so-called perfect reconstruction filterbank.

An oversampled DFT filterbank using WOLA (weighted overlap-add) processing provides an extremely efficient and elegant solution. This tutorial will describe this filterbank within a dedicated ASIC and algorithmic procedures for casting many algorithms into a multi-rate framework.

Numerous advantages are obtained using the multi-rate framework:

- 1) Adaptive filtering techniques typified by the LMS algorithm are greatly affected by the eigenvalue spread problem. In short, the LMS algorithm stalls when the input signal possesses a large ratio between the maximum and the minimum eigenvalue. This happens when the signal is distinctly non-white including many useful signal classes such as speech. The subband approach significantly reduces the coloration by representing the original spectrum as a parallel combination of much whiter subband signals. The original coloration is largely captured by the inherent scaling of the subbands.
- 2) Subbands may be adapted separately. This is a result of the orthogonality. This proves to be an advantage when processing power is limited. This tradeoff does not exist in the fullband LMS system where all taps must be adapted all the time.
- 3) Enhanced Tracking ability:
Each subband may be adapted with separate convergence factors. This is useful in applications where narrowband disturbances exist. Only the subbands affected need be adapted helping to concentrate resources in these subbands and/or reducing power consumption.
- 4) Filtering complexity reduction:
The filtering operation complexity is greatly reduced by converting intensive time-domain convolutions to relatively short frequency-domain convolutions. In certain cases, the filtering in each parallel path may be reduced to multiplication by a single (possibly) complex value.

As mentioned previously, filterbanks are used in many important applications; many more than can be listed here. Typical applications are:

- 1) **Coding applications** In the encoder, the input signal is passed through an analysis filterbank after which each subband is quantized (coded) with a precision dependant on a psycho-acoustic model. This model is selected to code only the perceptually significant portions of the input signal to reduce the overall bit-rate. The synthesis filterbank in the decoder then reproduces an approximation of the input signal by means of this coded digital stream. The combination of the analysis filterbank (encoder) and the synthesis filterbank (decoder) may be designed to possess the perfect reconstruction property or the approximate reconstruction property (pseudo-QMF) depending on fidelity and delay requirements. In this application, the quantization noise may be roughly classified as additive subband noise. The synthesis filterbank performs double-duty by synthesizing the output signal while rejecting any generated quantization noise which is out-of-band.

- 2) **Adaptive filtering applications** Here, the filterbank is not intended to reconstruct the input signal directly but after modifications have been made to the analysis signal. Typically, the filterbank is being invoked to model a desired system (as in hearing aid applications) or to model an undesired or disturbing system in such a manner that the original disturbance may be cancelled (as in echo cancellation systems). These modifications may be scalar real multiplications as in hearing aid applications or may be scalar complex multiplications or vector complex multiplications in the case of echo cancellation. Since the modifications are multiplicative, different criteria for creating and using filterbanks in these applications must be developed as compared to coding applications.

The WOLA filterbank structure is highly configurable and best performance is of course only achieved with an understanding of the optimizations and tradeoffs that can be made within its structure. This tutorial will describe how these optimizations should be made for typical applications.