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Complex Wavelet Modulation Subbands for Speech Compression

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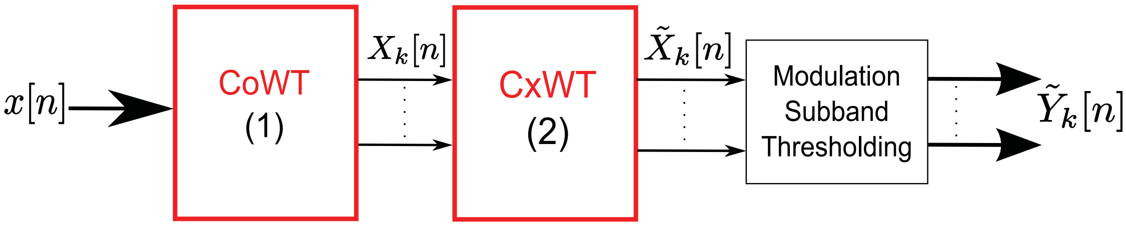
Low-frequency modulation of sound carry essential information for speech and music [1]. They must be preserved for compression. The complex modulation spectrum has already been used for audio compression and is commonly obtained by spectral analysis of the sole temporal envelopes of the subbands out of a time/frequency analysis (Modified Discrete Cosine Transform combined with a Modified Discrete Sine Transform *e.g.* in [2]). However, amplitudes and tones of speech or music tend to vary slowly over time thus the temporal envelopes are often smooth and mostly of polynomial type. Processing in this domain usually creates undesirable distortions because only the magnitudes are taken into account and the phase data is often neglected. We remedy this problem with the use of a complex wavelet transform as a more appropriate envelope and phase processing tool. Complex wavelets carry both magnitude and phase explicitly with great sparsity and preserve well polynomials. Moreover an analytic Hilbert-like transform is possible with complex wavelets implemented as an orthogonal filter bank [3].



This figure illustrates the proposed method. x is the audio signal. X_k are the time/scale subbands (or scalogram) from the Continuous Morlet Wavelet Transform (CoWT) as defined by $CoWT_\sigma(x) = \int_{-\infty}^{+\infty} \Psi_\sigma(t)x(t)dt$ with $\Psi_\sigma(t) = C_\sigma \pi^{-\frac{1}{4}} e^{-\frac{1}{2}t^2}(e^{i\sigma t} - e^{-\frac{1}{2}\sigma^2})$ and $\sigma = 10$. $\tilde{X}_k$ are the modulation scale/scale subbands from the Complex Wavelet Transform (CxWT) implemented via a non-redundant 3 band flexible orthogonal filterbank [3] with complex Daubechies wavelet filters of length 10. And finally $\tilde{Y}_k$ are the subbands after (hard or soft) thresholding and before quantization. By working in this alternative transform domain coined as “Modulation Subbands”, this transform shows very promising compression capabilities thanks to interesting sparsity properties and suggests new approaches for joint spectro-temporal analytic processing of slow frequency and phase varying audio signals.

References

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