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# Real-world Particle Filtering-based Speech Enhancement


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## Particle Filtering for Speech Enhancement

Particle Filters (PFs): sequential Monte Carlo methods, approximate solutions to state estimation problem, they can handle non-Gaussian non-linear models.

They operate by:

- Drawing candidates for the state to estimate
- Assigning scores to these candidates (based on model & measurement)
- Forming a global estimate.

Speech enhancement or de-noising can be formulated in state-space form, thus PFs have been used for this purpose, with very good reported results in AWGN.

However they have been so far:

- More computationally expensive than traditional solutions
- Not able to endure "real-world" situations (complex noise, low SNR, etc.)

## Main Objective and Constraints

The main objective is to obtain a PF-based algorithm with the following constraints:

(1) Capable of operating in complex noise conditions, not restricted to stationary noise, and/or using ideal voice activity detectors (VADs)

(2) As "light" as possible computationally

(3) Able to work for wideband speech (0-10 kHz bandwidth in this work)

(4) Resulting method must perform favourably in comparison with well-reviewed and well-established algorithms.

## Proposed Solution Overview

In the proposed solution:

- Subband domain processing (reduces required processing rate, smaller/lighter models can be used in each subband)
- Elementary subband PF units: very light, can operate reliably with less than a dozen particles each
- Flexibility and robustness in terms of noise handling, with internal and/or external noise statistics estimation possible
- Psychoacoustic/perceptual constraining can be incorporated with ease.

## Subband Domain Processing

- Use of maximally decimated filterbanks, to process minimal amount of data
- "Medium" amount of bands (32) for time-frequency resolution compromise.
- Three tested filterbank configurations
  - Multi-stage Wavelet-Packet decomposition of depth 5 (24 coeffs. each)
  - Pseudo-QMF filterbank with a Kaiser prototype window (468 coeffs.)
  - Modified Discrete Cosine Transform (MDCT) with a Kaiser-Bessel Derived window (64 coeffs.).

## Elementary Subband Particle Filter Units

Proposed subband speech and noise model:

- Speech: 1st order time-varying autoregression (TVAR),
- Noise: 0th order TVAR, i.e. a Gaussian process with time-varying standard deviation.
- Elementary subband PF units are thus very simple
- In addition, Rao-Blackwellization* can be used, the amount of particles required is then small i.e. less than a dozen particles (good robustness).

With known noise variances in each band (see below for details), a particle update is simple and consists of:

- Drawing 2 Gaussian random numbers (maintained set of means, fixed variances)
- 1-dimensional Kalman Filter update (~ 5 multiplications, 5 additions of real numbers)
- Computing the particle's weight (exponential and square root functions).

*Rao-Blackwellization: Rao-Blackwellized Particle Filters (RBPFs) are applicable when a subset of a state vector is linear-Gaussian, conditional upon other states. Optimal Kalman filtering can be used for the linear part, and PF filtering for the non-linear part.

## Noise Handling

Two solutions are proposed for noise handling:

1. Draw candidates for noise level (gain) internally at each iteration
   → Advantageous computationally (very low additional cost, i.e. one Gaussian number to draw and another variable to resample)
   → Allows for very sudden changes in noise level (sample-by-sample treatment)
2. Alternatively, external background noise Power Spectral Density (PSD) estimation (e.g. minimum statistics based estimator)
   → Estimated PSD can be "discretized" to a single point in each band
   → More robust than Voice Activity Detection

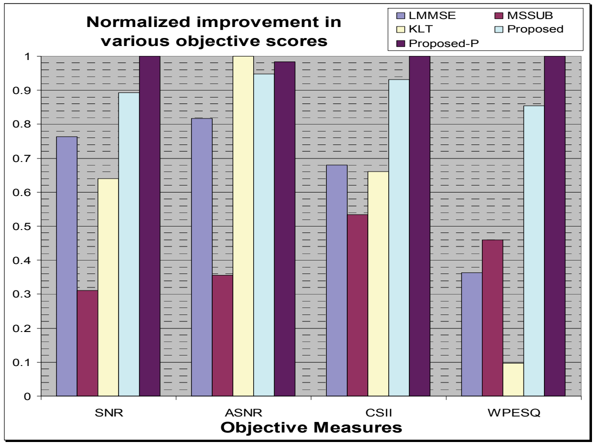
## Perceptual Constraining

If external background noise PSD estimation is available:

- Use of perceptual hearing features, in particular simultaneous (i.e. frequency) masking curves
- Simple under/overestimation of the returned noise level can then be applied, depending on some conservative rules
- For example, an overestimation would be removing more noise, but would also introduce more speech distortion. The opposite also holds.

## Performance and Simulations

- Comparisons with other schemes: averages over 12 different conditions – low, mid, high input SNR, and cafeteria, busy street, stadium and rain noise types
- Several objective measures used for evaluation: SNR, average (i.e. segmental) SNR, intelligibility index (CSII), wideband perceptual speech quality (w-PESQ)
- M-band (parallel) filterbanks were found to provide better results (better frequency selectivity, less inter-band leakage effects)
- Performance of internal noise PSD estimation approach close to performance with external dedicated noise estimation, but faster (less complex)




***LMMSE:*** *Log-Spectral Amplitude Estimator* ***MSSUB:*** *Multi-band Spectral Subtraction*

***KLT:*** *Subspace method* ***Proposed:*** *proposed method (-P is with perceptual constraints)*

## Audio Demonstrations

Corresponding audio demonstrations are available on-line, for anyone to subjectively assess the quality of enhanced speech signals:

http://www.site.uottawa.ca/bouchard/papers/cip2010.zip