

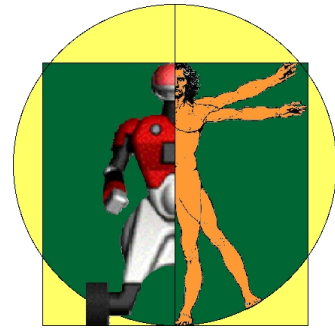


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INSTITUT DES MATÉRIAUX
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LABORIUS



Microphone Array Post-Filter for Separation of Simultaneous Non-Stationary Sources

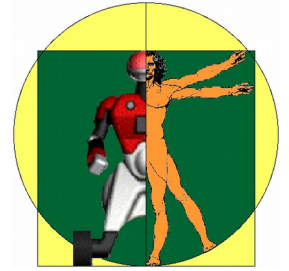
Jean-Marc Valin, Jean Rouat, François Michaud

Department of Electrical Engineering and Computer Engineering
Université de Sherbrooke, Québec, Canada
Jean-Marc.Valin@USherbrooke.ca



Fondation canadienne pour l'innovation



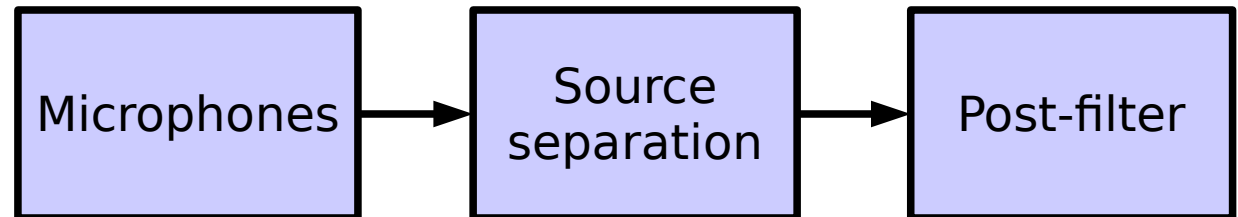


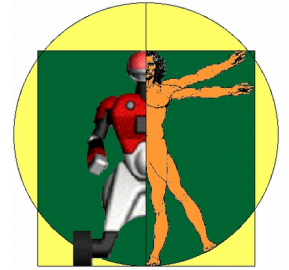
Motivations

The context: sound source separation

The problem: beamforming and similar techniques provide limited noise reduction

The solution: use a post-filter to further reduce noise and interference





Approach

Linear source separation

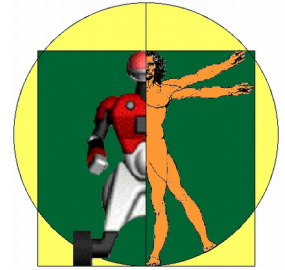
Geometric Source Separation (Parra) is used
Works for any linear separation algorithm

Post-filter

Frequency-domain processing

Based on the optimal Ephraim and Malah estimator

Gain modification according to probability of speech presence (Cohen)



Contribution

Multiple sources of interest

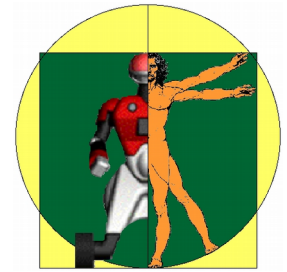
Generalize post-filters to separation of multiple sources

Non-stationary noise

Decouple background noise (stationary) and directional interference (transient)

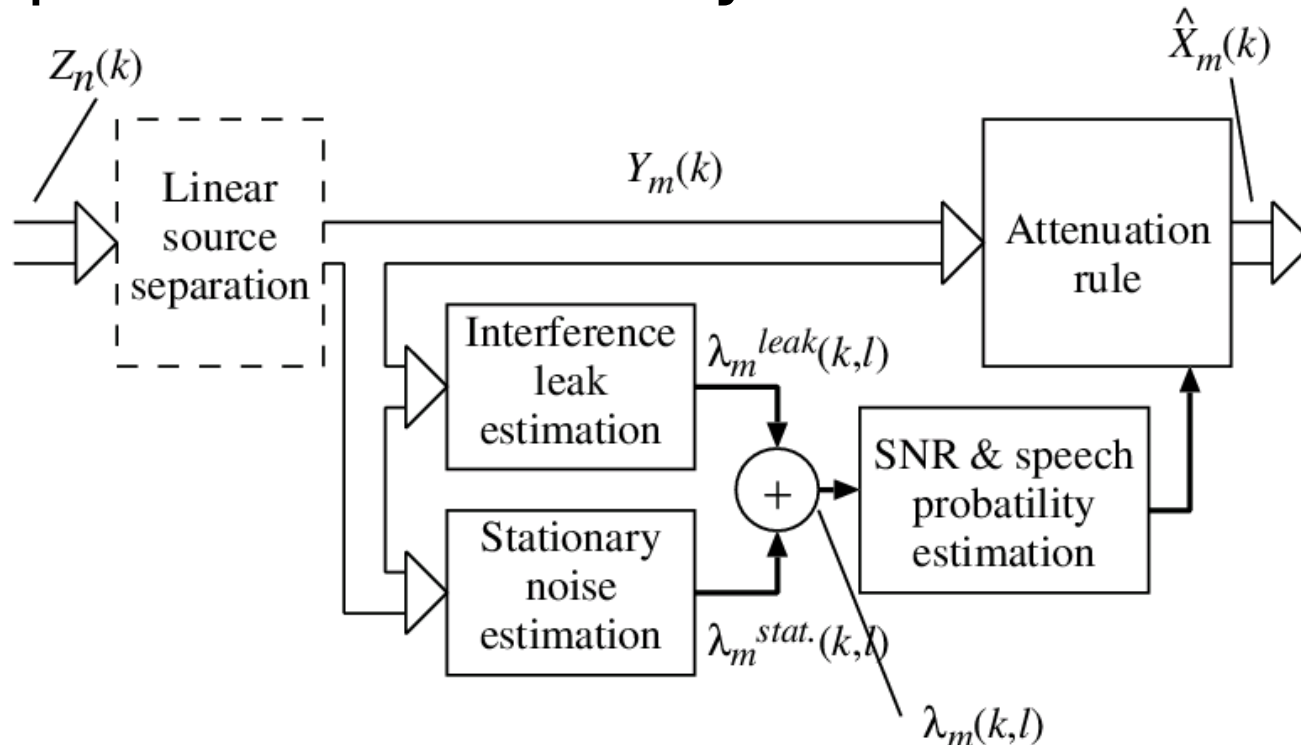
Fast estimation of interference

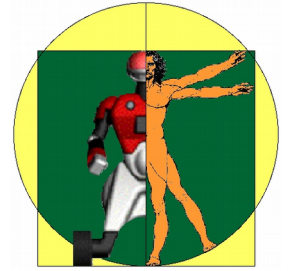
Direct estimation from initial separation



Post-Filter Overview

Noise estimate as the sum of two components (stationary + transient)





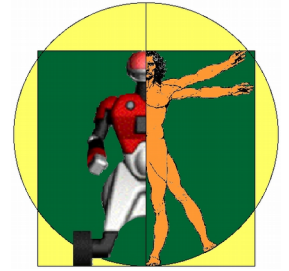
Background Noise Estimation

Minima-Controlled Recursive Average (Cohen)

Applied for each source of interest

Initial estimate provided directly from the microphones

$$\lambda_m^{stat.}(k, \ell_0) = \frac{1}{N^2} \sum_{n=0}^{N-1} \sigma_{x_n}^2(k)$$



Interference Estimation

Source separation leaks

Incomplete adaptation

Inaccuracy in localization

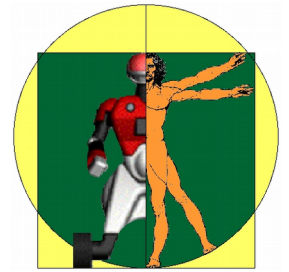
Reverberation

Imperfect microphones

Estimation from other separated sources

$$\lambda_m^{leak}(k, l) = \eta \sum_{i=0, i \neq m}^{M-1} S_i(k, l)$$

$$S_m(k, l) = \alpha_s S_m(k, l - 1) + (1 - \alpha_s) Y_m(k, l)$$



Suppression Rule

Loudness-domain optimal estimator

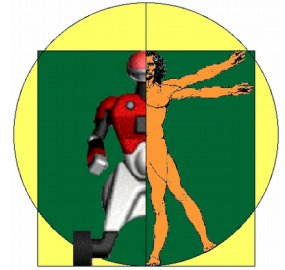
$$\hat{X}_m(k, l) = G_m(k, l)Y_m(k, l)$$

Assuming speech is present:

$$G_{H_1}(k) = \frac{\sqrt{v(k)}}{\gamma(k)} \left[\Gamma \left(1 + \frac{\alpha}{2} \right) M \left(-\frac{\alpha}{2}; 1; -v(k) \right) \right]^{\frac{1}{\alpha}}$$

$$\gamma(k) \triangleq |Y(k)|^2 / \lambda(k) \quad \xi(k) \triangleq E \left[|X(k)|^2 \right] / \lambda(k)$$

$$v(k) \triangleq \gamma(k)\xi(k) / (\xi(k) + 1)$$



Speech Presence Uncertainty

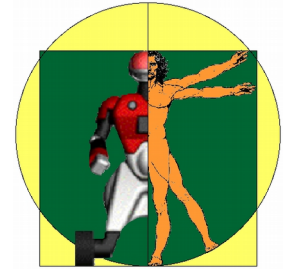
Optimal gain modification for loudness-
domain

$$G(k) = [p(k)G_{H_1}^\alpha(k) + (1 - p(k))G_{min}^\alpha]^\frac{1}{\alpha}$$

Setting $G_{min} = 0$ leads to

$$G(k) = p^2(k)G_{H_1}(k)$$

Unlike log-domain estimator, no arbitrary
limit on attenuation



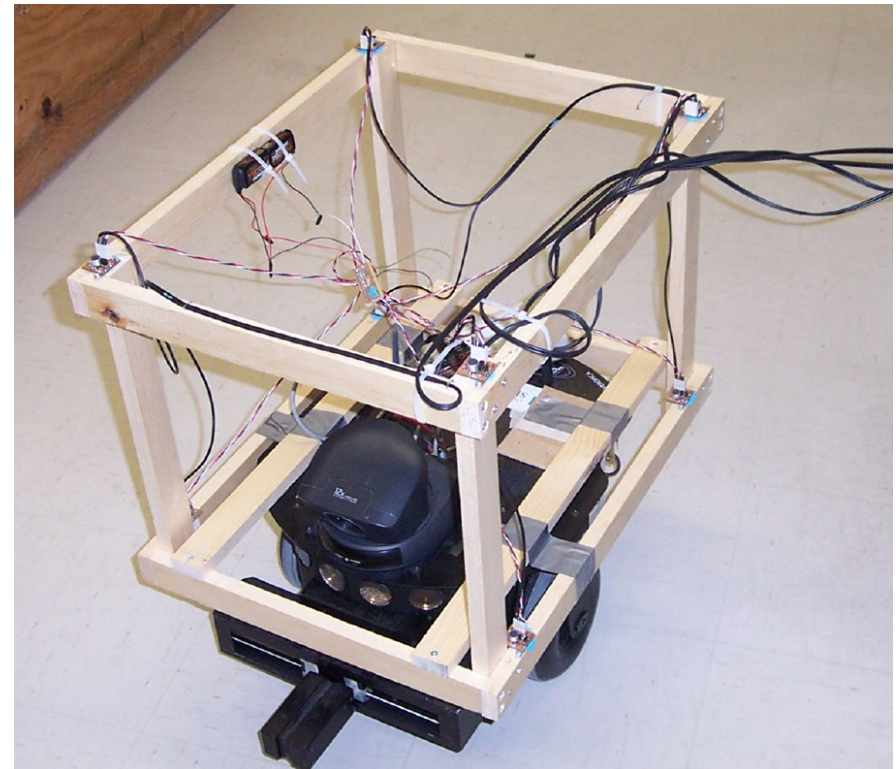
Experimental Setup

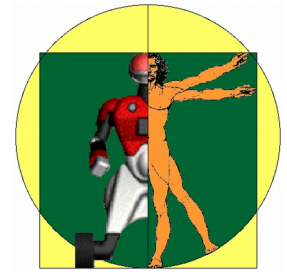
Array of 8 inexpensive microphones on a
mobile robot

Automatic
localization

Noisy conditions








Moderate
reverberation

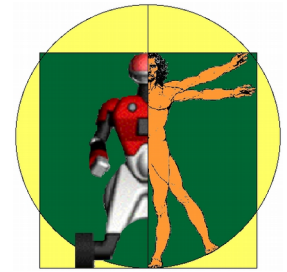




Results (Signal-to-Noise Ratio)

Three voices recorded separately so clean signal is available

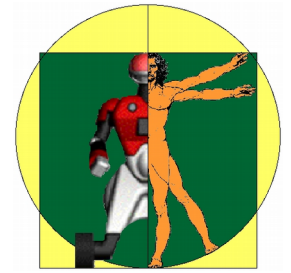
SNR (dB)	female 1	female 2	male 1
Microphone input 	-1.8	-3.7	-5.2
GSS only	9.0 	6.0 	3.7 
GSS+single channel	9.9	6.9	4.5
GSS+proposed	12.1 	9.5 	9.4 



Results (Log-Spectral Distortion)

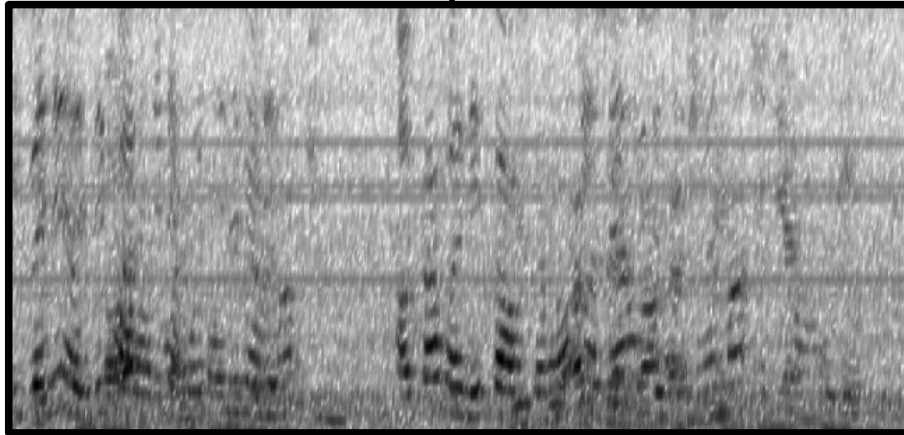
$$LSD = \frac{1}{L} \sum_{l=0}^{L-1} \left[\frac{1}{K} \sum_{k=0}^{K-1} \left(20 \log_{10} \frac{|X(k, l)| + \epsilon}{|\hat{X}(k, l)| + \epsilon} \right)^2 \right]^{\frac{1}{2}}$$

LSD (dB)	female 1	female 2	male 1
Microphone input	17.5	15.9	14.8
GSS only	15.0	14.2	14.2
GSS+single channel	9.7	9.5	10.4
GSS+proposed	6.5	6.8	7.4

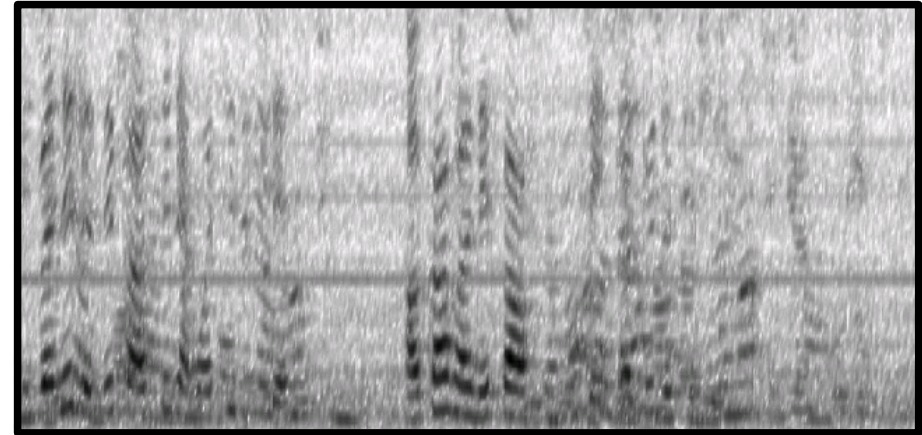


Results (spectrograms)

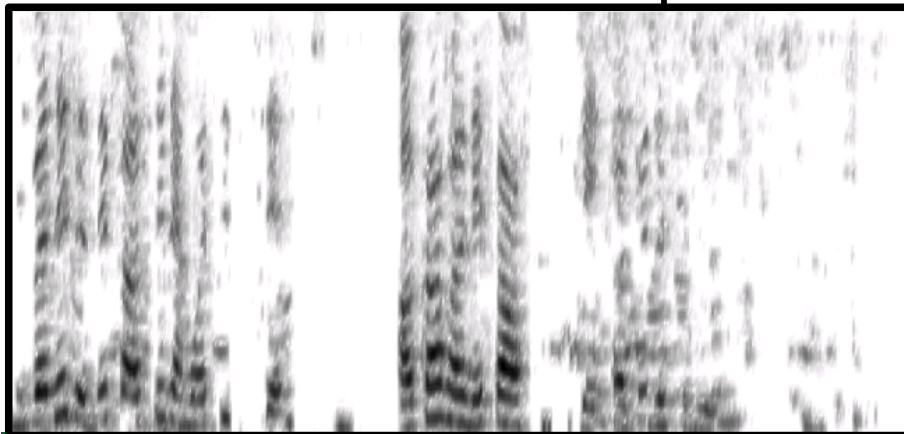
Input



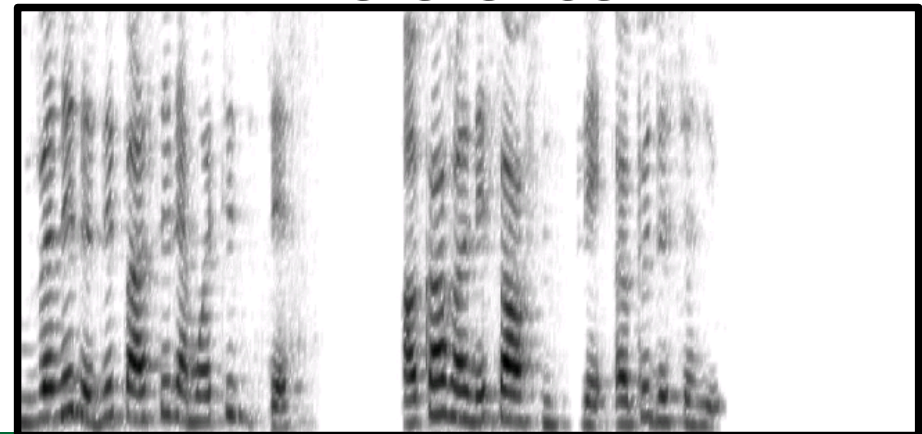
GSS

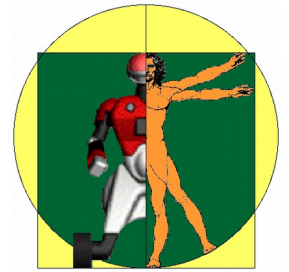


Post-filter output



Reference





Conclusion


Source separation post-filter

Based on optimal loudness-domain estimator

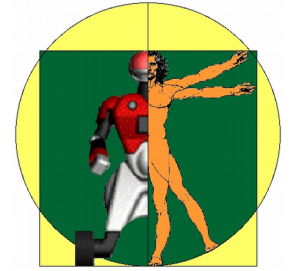
Interference estimated using other sources

Future work

Robustness to reverberation

 original  processed

Integration with speech recognition



Questions?

