Room Impulse Response Estimation by Iterative Reweighted L₁-Norm



Danilo Greco



s(k) : transmitted audio signal



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 $x_i(k)$: received signal at *i*-th microphone

$$x_i(k) = h_i(k) * s(k), \quad i = 1, \cdots, M$$



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Estimate $h_i(k)$ given $x_i(k)$ for $i=1,\cdots,M$

Applications

Dereverberation Speech enhancement



Applications

Dereverberation Speech enhancement





Room aware sound reproduction

Applications

Dereverberation Speech enhancement





Room aware sound reproduction



Room Geometry estimation [Dokmanic et al. PNAS 2013]

A short taxonomy on RIR estimation



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Statistical methods

$$x_i(k) = h_i(k) * s(k), \quad i = 1, \cdots, M$$

A short taxonomy on RIR estimation



The Cross – Relation Identity (1)

For every couple of microphones:

$$h_i(k) * h_j(k) * s(k) = h_j(k) * h_i(k) * s(k), \quad i \neq j$$



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For every couple of microphones:

$$h_{i}(k) * h_{j}(k) * s(k) = h_{j}(k) * h_{i}(k) * s(k), \quad i \neq j$$

$$h_{i}(k) * x_{j}(k) = h_{j}(k) * x_{i}(k)$$

$$i \neq j$$

$$h_{i}(k) * h_{i}(k) + h_{j}(k) * h_{j}(k)$$

The Cross – Relation Identity (2)

$$x_j(k) * h_i(k)$$
 = $\mathbf{X}_j \mathbf{h}_i$

Shift to matrix form

$$\begin{array}{c} & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ & & \\ \end{array} \end{array} \begin{array}{c} & & & \\ & & \\ & & \\ & & \\ & & \\ \end{array} \end{array} \begin{array}{c} & & & \\ &$$

The Cross – Relation Identity (2)

$$x_j(k) * h_i(k) = \mathbf{X}_j \mathbf{h}_i$$

Shift to matrix form

In absence of noise

$$\mathbf{X}_i \mathbf{h}_j = \mathbf{X}_j \mathbf{h}_i$$

Quadratic cost function

$$\begin{split} J(\mathbf{h}) &= \sum_{i \neq j} \|\mathbf{X}_i \mathbf{h}_j - \mathbf{X}_j \mathbf{h}_i\|_2^2 = \mathbf{h}^\top \mathbf{Q} \mathbf{h} \\ \min_{\mathbf{h}} J(\mathbf{h}) \quad s.t. \quad \|\mathbf{h}\|_2^2 = 1 \end{split}$$

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Singular value problem [Tong et al., 1994]:

solution given by the singular vector corresponding to the smallest singular value of $\, {f Q} \,$

[Tong et al., 1994]: L. Tong, G. Xu and T. Kailath. «Blind identification and equalization based on second order statistics: a time domain approach», IEEE Trans. On Information Theory, vol. 40, no. 2, pp.340-349, Mar. 1994..

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Singular value problem [Tong et al., 1994]:

solution given by the singular vector corresponding to the smallest singular value of $\, {f Q} \,$

<u>Drawbacks</u>:

- Channels must be co-prime
- RIR length must be known
- Sensitivity to «holes» in the signal spectrum

[Tong et al., 1994]: L. Tong, G. Xu and T. Kailath. «Blind identification and equalization based on second order statistics: a time domain approach», IEEE Trans. On Information Theory, vol. 40, no. 2, pp.340-349, Mar. 1994..

RIR sparsity prior





Sparsity holds for direct path and early reflections

RIR sparsity prior





Sparsity holds for direct path and early reflections

Reconstruction of <u>RIR direct path and early reflections</u> is sufficient for room geometry reconstruction and has proven to work also in speech enhancement [Yu et al 2012] and dereverberation [Lin et al, 2007].

[Yu et al 2012]: Yu, Meng, et al. "Multi-Channel Regularized Convex Speech Enhancement Model and Fast Computation by the Split Bregman Method." *Audio, Speech, and Language Processing, IEEE Transactions on* 20.2 (2012): 661-675. [Lin et al, 2007]: Lin, Yuanqing, et al. "Blind channel identification for speech dereverberation using l1-norm sparse learning." *Advances in Neural Information Processing Systems*. 2007.

Penalty inducing sparsity



Penalty inducing sparsity



[Kowalczyk et al, 2013]: Kowalczyk, Konrad, et al. "Blind System Identification Using Sparse Learning for TDOA Estimation of Room Reflections." *Sig. Proc. Letters, IEEE*20.7 (2013): 653-656.

Penalty inducing sparsity



Drawbacks:

- Non-convex problem due to the quadratic equality constraint - L_1 norm penalizes larger coefficients more: the solution is not in general the same of L_0 norm.

[Kowalczyk et al, 2013]: Kowalczyk, Konrad, et al. "Blind System Identification Using Sparse Learning for TDOA Estimation of Room Reflections." Sig. Proc. Letters, IEEE20.7 (2013): 653-656.

Anchor constraint and non-negativity constraint



[Lin et al 2007]: Lin, Yuanqing, et al. "Blind sparse-nonnegative (BSN) channel identification for acoustic timedifference-of-arrival estimation." *Applications of Signal Processing to Audio and Acoustics, 2007 IEEE Workshop on*. IEEE, 2007.

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Anchor constraint and non-negativity constraint



Drawbacks:

- amplitude distortion, peak of the anchor overly enhanced
- does not solve the L_1 penalty limitations

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New approach

Convex solution, no distortion due to anchor

New approach

$$\min_{\mathbf{h}} J(\mathbf{h}) + \lambda \|\mathbf{h}\|_{1} \quad s.t. (\|\mathbf{h}\|_{1} = 1), \quad \mathbf{h} \ge 0.$$

$$\mathbf{L}_{1} \text{ equality constraint}$$

Convex solution, no distortion due to anchor

<u>BUT</u> L_1 norm appears both as a constraint and a penalty:

New approach

$$\min_{\mathbf{h}} J(\mathbf{h}) + \lambda \|\mathbf{h}\|_{1} \quad s.t. (\|\mathbf{h}\|_{1} = 1), \quad \mathbf{h} \ge 0.$$

$$\mathbf{L}_{1} \text{ equality constraint}$$

Convex solution, no distortion due to anchor

<u>BUT</u> L_1 norm appears both as a constraint and a penalty:

no more sparsity - inducing effect !!

Toy problem in two dimensions (1)



Quadratic cost function without penalties

Toy problem in two dimensions (1)



Quadratic cost function without penalties



Quadratic cost function without L_1 penalty

Iterative reweighted L₁ penalty

Solve a sequence of sub - problems for z = 1, ..., Z:

$$\hat{\mathbf{h}}^{(z)} = \min_{\mathbf{h}} J(\mathbf{h}) + \lambda \|\mathbf{p}^{(z)} \odot \mathbf{h}\|_1 \quad s.t. \quad \|\mathbf{h}\|_1 = 1, \quad \mathbf{h} \ge 0$$

Iterative reweighted L₁ penalty

Solve a sequence of sub - problems for
$$z = 1, ..., Z$$
:
 $\hat{\mathbf{h}}^{(z)} = \min_{\mathbf{h}} J(\mathbf{h}) + \lambda \| \mathbf{p}^{(z)} \odot \mathbf{h} \|_{1} \quad s.t. \quad \| \mathbf{h} \|_{1} = 1, \quad \mathbf{h} \ge 0$
Weight update rule
 $p(l)^{(z)} = \frac{1}{\hat{h}(l)^{(z-1)} + \epsilon} \quad for \quad l = 1, \cdots, L$

Smaller elements of vector \mathbf{h} are more penalized than bigger ones.

Toy problem in two dimensions (2)



Weighted L_1 penalty p(1) > p(2)

Toy problem in two dimensions (2)



Initialization and convergence

How to initialize $\mathbf{p}^{(1)}$

<u>Use the solution of the anchor – constrained problem</u>

$$\hat{\mathbf{h}}^{(1)} = \underset{\mathbf{h}}{argminJ}(\mathbf{h}) + \lambda \|\mathbf{h}\|_1 \quad s.t. \quad h_1(a) = 1, \quad \mathbf{h} \ge 0$$

Initialization and convergence

How to initialize $\mathbf{p}^{(1)}$?

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At convergence
$$\hat{\mathbf{h}}^{(z-1)} \approx \hat{\mathbf{h}}^{(z)}$$
 therefore
 $p(l)^{(z)} \hat{h}(l)^{(z)} \approx 1 \quad for \quad \hat{h}(l)^{(z)} \neq 0$ Weighted L_1 norm is
 $p(l)^{(z)} \hat{h}(l)^{(z)} \approx 0 \quad for \quad \hat{h}(l)^{(z)} = 0$

Example of the iterative procedure



Example of the iterative procedure



<u>Spurious peaks</u> are gradually eliminated from the estimated RIR

Even if non-negativity is not perfect in the GT RIR, the estimated RIR well preserves <u>peaks position and</u> <u>energy</u>

Experiments

- Simulated room of 5 x 4 x 6 m
- 2 microphones and a source randomly placed
- RIRs simulated with the image method [Allen & Berkley, 1979]
- Synthetic and real signals: white noise, rustle, male voice
- Variable SNR: 0, 6, 14, 20, 40 dB



- 50 Monte Carlo simulations for each SNR

[Allen & Berkley, 1979]: Allen, Jont B., and David A. Berkley. "Image method for efficiently simulating small-room acoustics." *The Journal of the Acoustical Society of America* 65.4 (1979): 943-950.

Ground Truth









Metrics of evaluation

Average peak position mismatch

$$\left(A_{PPM} = \sum_{i=1}^{50} \sum_{p=1}^{P_i} \frac{|\tau_{p,i}^{gt} - \tau_{p,i}^{e}|}{50P_i}\right)$$

Peak position accuracy over the <u>inliers</u>

Metrics of evaluation



$$A_{PPM} = \sum_{i=1}^{50} \sum_{p=1}^{P_i} \frac{|\tau_{p,i}^{gt} - \tau_{p,i}^{e}|}{50P_i}$$

Peak position accuracy over the <u>inliers</u>

Average percentage of unmatched peaks

$$A_{PUP} = \sum_{i=1}^{50} \frac{K - P_i}{50K}$$

Percentage of <u>outliers</u> (> 20 samples)

Metrics evaluated on the first (sparse) part of the RIR

Quantitative results

Synthetic: white noise

SNR	EIG	L1NN	IL1P	
0 dB	4.51 [0.34]	1.96 [0.16]	1.17 [0.11]	
6 dB	4.24 [0.34]	0.77 [0.10]	0.40 [0.04]	
14 dB	3.86 [0.31]	0.34 [0.03]	$0.28 \ [0.01]$	
20 dB	3.42 [0.30]	0.28 [0.03]	$0.27 \ [0.01]$	
40 dB	3.20 [0.28]	$0.26 \ [0.00]$	0.27 [0.00]	
	ADDI			
	^{11}PPM	$[\square P U P]$		

Quantitative results

Synthetic: white noise

Real: <u>rustle</u>

SNR	EIG	L1NN	IL1P
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6 dB	4.24 [0.34]	0.77 [0.10]	0.40 [0.04]
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40 dB	3.20 [0.28]	$0.26 \ [0.00]$	0.27 [0.00]
0 dB	4.53 [0.39]	2.46 [0.18]	$\fbox{0.14}$
6 dB	4.50 [0.37]	1.29 [0.12]	$0.54 \ [0.02]$
14 dB	3.82 [0.38]	0.39 [0.02]	0.28 [0.00]
20 dB	3.65 [0.35]	0.29 [0.01]	0.28 [0.00]
40 dB	3.31 [0.29]	$0.28 \ [0.00]$	0.28 [0.00]
	A_{PPM}	$[A_{PUP}]$	

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0 dB	4.66 [0.38]	2.87 [0.23]	3.31 [0.16]
6 dB	4.91 [0.37]	1.96 [0.19]	1.36 [0.08]
14 dB	4.44 [0.40]	0.98 [0.10]	$0.58\ [0.01]$
20 dB	4.30 [0.35]	0.50 [0.04]	$0.39 \ [0.01]$
40 dB	3.64 [0.35]	0.32 [0.01]	0.29 [0.00]
	0 dB 6 dB 14 dB 20 dB 40 dB 0 dB 6 dB 14 dB 20 dB 40 dB 0 dB 6 dB 14 dB 20 dB 40 dB	$\begin{array}{c ccccccccccccccccccccccccccccccccccc$	$0 \ dB$ $4.51 \ [0.34]$ $1.96 \ [0.16]$ $6 \ dB$ $4.24 \ [0.34]$ $0.77 \ [0.10]$ $14 \ dB$ $3.86 \ [0.31]$ $0.34 \ [0.03]$ $20 \ dB$ $3.42 \ [0.30]$ $0.28 \ [0.03]$ $20 \ dB$ $3.42 \ [0.30]$ $0.28 \ [0.03]$ $40 \ dB$ $3.20 \ [0.28]$ $0.26 \ [0.00]$ $0 \ dB$ $4.53 \ [0.39]$ $2.46 \ [0.18]$ $6 \ dB$ $4.50 \ [0.37]$ $1.29 \ [0.12]$ $14 \ dB$ $3.82 \ [0.38]$ $0.39 \ [0.02]$ $20 \ dB$ $3.65 \ [0.35]$ $0.29 \ [0.01]$ $40 \ dB$ $3.31 \ [0.29]$ $0.28 \ [0.00]$ $0 \ dB$ $4.66 \ [0.38]$ $2.87 \ [0.23]$ $6 \ dB$ $4.91 \ [0.37]$ $1.96 \ [0.19]$ $14 \ dB$ $4.44 \ [0.40]$ $0.98 \ [0.10]$ $20 \ dB$ $4.30 \ [0.35]$ $0.50 \ [0.04]$

 A_{PPM} $[A_{PUP}]$

Conclusions

- The proposed method yields a <u>convex</u>, <u>computationally efficient</u> formulation for the blind sparse non-negative RIR estimation problem.
- The iterative weighted L_1 penalty fairly approximates the <u>optimal L_0 penalty</u>.
- The proposed method <u>outperforms</u> current approachs in almost all SNR conditions and for different sound sources, <u>both synthetic and real.</u>
- <u>Future works will assess the method effectiveness on a set of applications like</u> <u>speech enhancement and room geometry reconstruction.</u>