Localization Cues Preservation in Hearing Aids by Combining Noise Reduction and Dynamic Range Compression

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Introduction

Dynamic range compression (DRC)

Aim: amplify the soft sound without reaching the pain threshold

Dynamic Range Compression (DRC):

- Attenuate the output if the level exceeds a given threshold
- Number of frequency bands : 8 $\rightarrow$ 32

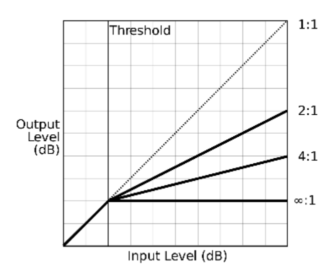


Figure 1: DRC broken-stick function

DRC - issue

Issue: left and right DRC gain are different:

- Hearing loss compensation distorts the localization cues
- Localization performance decreasing[1]
- Speech in noise understanding performance decreasing[2]

[1] [Hassager et al., 2017b, Wiggins and Seeber, 2011, Van den Bogaert et al., 2006]
[2] [Schwartz and Shinn-Cunningham, 2013]

SNR-aware DRC[3]

- **Aims** :
  - reducing the speech dynamic range
  - preserving the original noise dynamic range
  - improving the output SNR
- **Idea**: fast compression for the speech period, slowly otherwise

Advantage

improve the localization in presence of reverberation [Hassager et al., 2017a]

Drawbacks

The attenuation of the noise only period depends on the previous speech content!

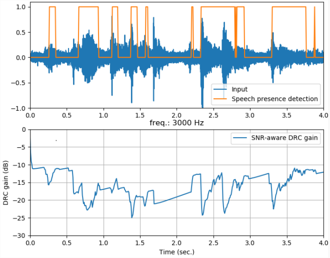


Figure 2: Noisy speech and speech presence detection (top), DRC gain at 3 kHz of the SNR-aware DRC

3[May et al., 2018]

Our approach

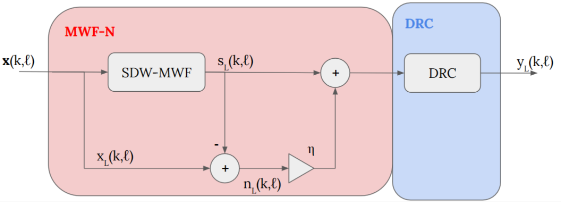


Figure 3: Standard association of noise reduction and DRC.

Our approach

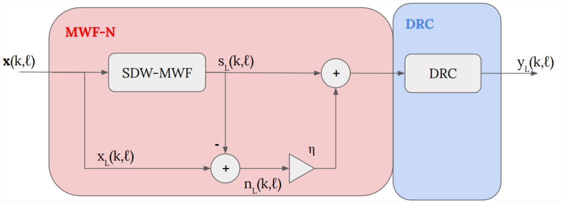


Figure 3: Standard association of noise reduction and DRC.

Objectives :

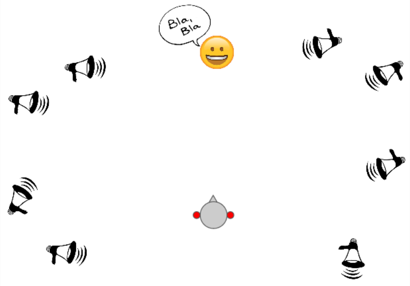
- Reducing the speech dynamic range
- Preserving the noise dynamic range
- Improving the SNR
- Preserving the localization cues of both components

Idea

Merge noise reduction and DRC

Data Model

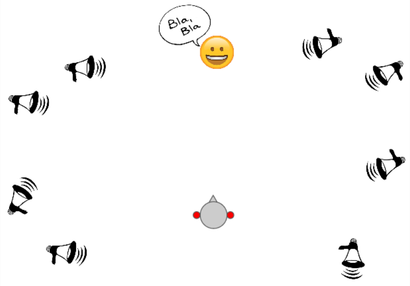
Acoustical scenario



At each microphone $x_m(t)$:

- one speech source (plane wave), $s(t)$, filtered by $h_m(t)$.
- a noise component (spatially diffuse), $n_m(t)$.

Acoustical scenario



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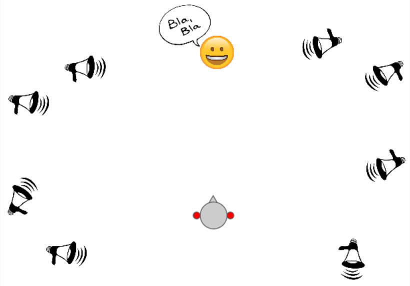
$$\text{Time domain: } x_m(t) = (h_m \star s)(t) + n_m(t) \tag{1}$$

Acoustical scenario



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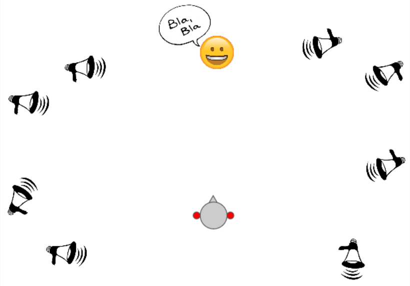
$$\text{STFT domain: } x_m(k,\ell) = h_m(k,\ell)s(k,\ell) + n_m(k,\ell) \tag{2}$$

Acoustical scenario



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$$\text{STFT domain: } x_m(k,\ell) = h_m(k,\ell)s(k,\ell) + n_m(k,\ell) \tag{2}$$

$$\text{matrix notation: } \boldsymbol{x}(k,\ell) = \boldsymbol{h}(k)\ s(k,\ell) + \boldsymbol{n}(k,\ell). \tag{3}$$

$\boldsymbol{x}(k,\ell) \in \mathbb{C}^M$, $\boldsymbol{h}(k,\ell) \in \mathbb{C}^M$ and $\boldsymbol{n}(k,\ell) \in \mathbb{C}^M$.

Acoustical scenario

Speech sparsity assumption in the STFT domain:

- $\mathcal{H}_0$: $\boldsymbol{x}(k,\ell) = \boldsymbol{n}(k,\ell)$,
- $\mathcal{H}_1$: $\boldsymbol{x}(k,\ell) = \boldsymbol{h}(k)s(k,\ell) + \boldsymbol{n}(k,\ell)$

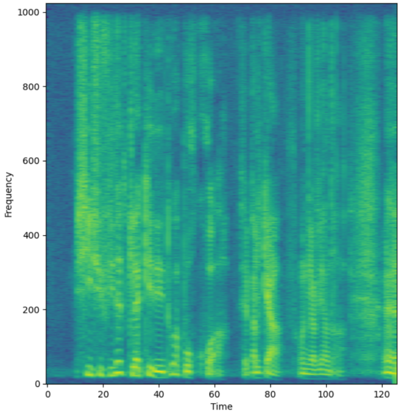

Figure 4: Speech spectrogram

Probabilistic model

Discrete Fourier coefficients are modeled as random variables:

$$s(k,\ell) \sim \mathcal{N}_{\mathbb{C}}\left(0, \phi_s(k,\ell)\right) \tag{4}$$

$$\boldsymbol{n}(k,\ell) \sim \mathcal{N}_{\mathbb{C}}\left(0, \Phi_{\boldsymbol{nn}}(k,\ell)\right) \tag{5}$$

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Noise covariance matrix model (spatially diffuse):

$$\Phi_{\boldsymbol{nn}}(k,\ell) = \phi_n(k,\ell)\Gamma_{\mathrm{diff}}(k) \tag{6}$$

Proposed Algorithm

Source separation

$\hat{s}_L(k,\ell)$: speech source estimate at the left ear (same for right ear)

$$\hat{s}_L(k,\ell) = \boldsymbol{w}^H(k,\ell)\ \boldsymbol{x}(k,\ell) \tag{7}$$

with $\boldsymbol{w}(k,\ell) \in \mathbb{C}^M$.

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Optimization problem:

$$\hat{\boldsymbol{w}}_L(k,\ell) = \underset{\boldsymbol{w}}{\operatorname{argmin}}\{\mathbb{E}\left[\left|s_L(k,\ell) - \boldsymbol{w}^H\boldsymbol{x}(k,\ell)\right|^2\right]\} \tag{8}$$

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Using the speech sparsity assumption

$$\begin{aligned}\mathbb{E}\left[\left|s_L(k,\ell) - \boldsymbol{w}^H\boldsymbol{x}(k,\ell)\right|^2\right] = P(\ell)\Big[&p(k,\ell)\mathbb{E}\left[|s_L(k,\ell) - \boldsymbol{w}^H\boldsymbol{x}(k,\ell)|^2 \mid \mathcal{H}_1\right] \\ &+(1-p(k,\ell))\mathbb{E}\left[|\boldsymbol{w}^H\boldsymbol{x}(k,\ell)|^2 \mid \mathcal{H}_0\right]\Big] \\ &+ (1-P(\ell))\left[\frac{1}{\mu_{\mathcal{H}_0}}\mathbb{E}\left[|s_L(k,\ell) - \boldsymbol{w}^H\boldsymbol{h}(k)s(k,\ell)|^2\right]\right. \\ &\left.+\mathbb{E}\left[|\boldsymbol{w}^H\boldsymbol{x}(k,\ell)|^2\right]\right]\end{aligned} \tag{9}$$

Informed Multichannel Wiener Filter (MWF) for the left ear:

$$\hat{\boldsymbol{w}}_L(k,\ell) = \left(\phi_s(k,\ell)\boldsymbol{h}(k)\boldsymbol{h}(k)^H + \mu(k,\ell)\Phi_{\boldsymbol{nn}}(k,\ell)\right)^{-1}\boldsymbol{h}(k)\phi_s(k,\ell)h_L(k)^*, \tag{10}$$

with

$$\mu(k,\ell) = P(\ell)\frac{1}{p(k,\ell)} + (1-P(\ell))\mu_{\mathcal{H}_0}. \tag{11}$$

- $p(k,\ell)$: narrowband speech presence probability estimation
- $P(\ell)$: broadband speech presence probability estimation

Dynamic range compression - i

Idea introduced by [Ngo et al., 2012] and similarly by [May et al., 2018]: use different DRC to process speech source and noise source

- DRC_s: to process the speech source when it is active at this T-F bin,
- DRC_n H_1: to process the speech source when it is NOT active at this T-F bin,
- DRC_n H_0: to process the speech source when it is NOT active at all bins,
- DRC_n: to process the noise component.

DRC	Attack (ms)	Release (ms)	Gain G_0 (dB)
DRC_s	10	60	0
DRC_n H_1	10	2000	-6
DRC_n H_0	10	2000	-10
DRC_n	2000	2000	-10

Multichannel estimator [Souden et al., 2010]

$P(\mathcal{H}_1|\boldsymbol{x}(k,\ell))$: *a posteriori* source presence presence probability, denoted $p(k,\ell)$

Bayes rule:

$$P(\mathcal{H}_1|\boldsymbol{x}(k,\ell)) = \frac{P(\boldsymbol{x}(k,\ell)|\mathcal{H}_1)P(\mathcal{H}_1)}{P(\boldsymbol{x}(k,\ell)|\mathcal{H}_1)P(\mathcal{H}_1) + P(\boldsymbol{x}(k,\ell)|\mathcal{H}_0)P(\mathcal{H}_0)} \tag{12}$$

$P(\boldsymbol{x}(k,\ell)|\mathcal{H}_1)$: data likelihood according to the Gaussian assumption

$P(\mathcal{H}_1)$: prior (adaptive [Cohen, 2002])

Source presence probability - ii

Broadband binary detector

$$\hat{P}(\ell) = \begin{cases} 1 & \text{if } \sum_k p(k,\ell) > t_{high} \text{ and } P(\ell-1) = 0 \\ 0 & \text{if } \sum_k p(k,\ell) > t_{low} \text{ and } P(\ell-1) = 1 \\ P(\ell-1) & \text{otherwise,} \end{cases} \tag{13}$$

Recursive filtering

$$P(\ell) = \alpha_P \hat{P}(\ell) + (1-\alpha_P) P(\ell-1) \tag{14}$$

Sum-up

- Similar to the proposition of [Ngo et al., 2012]
- Improvements :
  - DRC association more consistent with the literature
  - attack and release time constant decorrelation between DRC and broadband speech detection
  - binaural rather than monaural

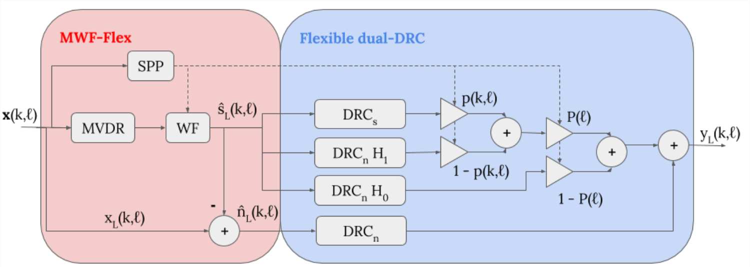
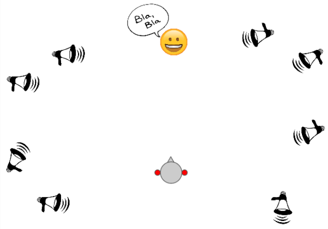


Figure 5: Our proposition

Experiments

Set-up

- Scenario:
  - speaker located in front of the listener
  - cafeteria noise
  - SNR: 5 dB
- Ideal scenario: SNR of 15 dB
- With 14 different HRTFs



Results - i

Interaural coherence

$$\mathrm{IC} = \max_{\tau} \left| \frac{\sum_t \tilde{y}_L(t+\tau)\tilde{y}_R(t)}{\sqrt{\sum_t |\tilde{y}_L(t)|^2 \sum_t |\tilde{y}_R(t)|^2}} \right| . \qquad (15)$$

Advantages

- Denoising across all the pipeline
- Interaural coherence closer to the ideal scenario

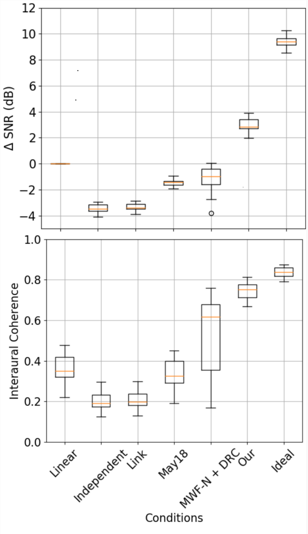


Figure 6: SNR improvement (top) and interaural coherence (bottom).

Effective Compression Ratio (ECR):

- < 1: more dynamic range
- > 1: less dynamic range

Drawbacks

- Speech cut $\rightarrow$ ECR deteriorated
- Residual noise component into the speech branch correlated with the speech $\rightarrow$ ECR < 1

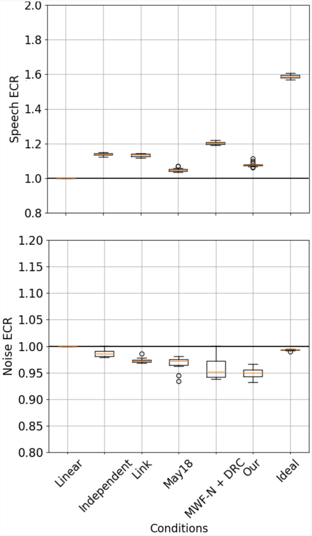


Figure 7: Speech ECR (top) and noise ECR (bottom).

Conclusion

- SNR improvement
- Better interaural coherence
- ECR deterioration due to estimation errors
- Future works: perceptive test

Thanks for your attention

Please share your comments and questions

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