## UGOSA

User-guided one-shot deep model adaptation for music source separation

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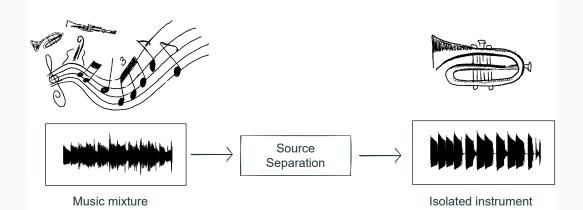


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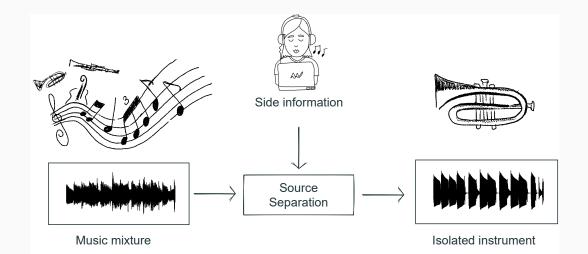
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## Audio Source Separation



## **User-driven Audio Source Separation**



#### NMF/NTF-based:

- Time activations [Laurberg et al., 2008, Ozerov et al., 2011, Duong et al., 2014a]
- TF activations [Lefevre et al., 2012, Jeong and Lee, 2015, Rafii et al., 2015]
- Interactive frameworks [Bryan and Mysore, 2013, Duong et al., 2014b]
- Humming [Smaragdis and Mysore, 2009], sing/play [FitzGerald, 2012], F0 [Durrieu and Thiran, 2012]

#### DL-based:

- Multi-task learning [Stoller et al., 2018, Hung and Lerch, 2020, Nakano et al., 2020]
- Class conditioning [Swaminathan and Lerch, 2019, Slizovskaia et al., 2019, Seetharaman et al., 2019, Karamatlı et al., 2019]

#### However:

- X require large datasets of mixtures and corresponding isolated sources;
- X if informed, require the side information also during training;

Fully-supervised DNNs:

- X require large datasets of mixtures and corresponding isolated sources;
- X if informed, require the side information also during training;
- X do not generalize well to test data with significant timbre variation.

**Proposal**: use the side information + specific training strategies

- Adaptation: fine-tuning of a pre-trained DNN using the side info
- One-shot: adaptation to one mixture and not a dataset
- Side information: time activations indicating where each instrument is active

Minimize difference between the estimated and the ground truth sources

$$L = \frac{1}{N} \sum_{n=1}^{N} \sum_{i=1}^{l} |\hat{s}_{i,n} - s_{i,n}|;$$

which represents the average absolute error between waveforms

- $\hat{s}_{i,n}$  estimated source *i* at time frame *n*;
- $s_{i,n}$  ground truth source *i* at time frame *n*.

Minimize the energy of the silent sources while forcing the mix reconstruction:

$$L = \frac{1}{N} \sum_{n=1}^{N} \left[ \underbrace{\sum_{i=1}^{l} |(h_{i,n} \cdot \hat{s}_{i,n}) - y_n|}_{\text{reconstruction loss}} + \underbrace{\lambda \sum_{i=1}^{l} |(1 - h_{i,n}) \cdot \hat{s}_{i,n}|}_{\text{activations loss}} \right]$$

where  $h_{i,n}$  are the binary activations of each instrument *i* at time frame *n*:

$$\begin{cases} h_{i,n} = 1 \text{ if instrument } i \text{ is active} \\ h_{i,n} = 0 \text{ if instrument } i \text{ is not active} \end{cases}$$

## **Experiments**

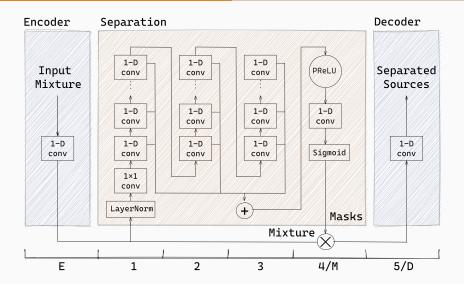
#### Data

- MUSDB18 [Rafii et al., 2017]
- 4 classes: bass, drums, vocals and other;
- first 10 songs of the test set only;
- binary activations of the ground truth sources.

#### Model:

- ConvTasnet adapted for music source separation [Luo and Mesgarani, 2019, Défossez et al., 2019]
- 10 epochs of fine-tuning using Ranger and Ir  $10^{-5}$

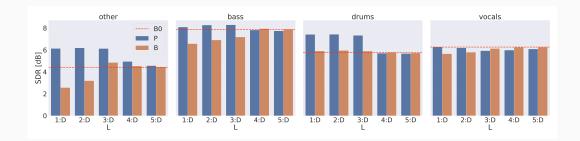
## **Fine-tuning strategies**



Example P-L2:D: from the 2nd block to the last one with the proposed loss;

## Results

SDR (dB)	#TP	other	bass	drums	vocals
P-L1:D	8.2M	6.1	8.1	7.4	6.3
P-L2:D	5.6M	6.2	8.3	7.4	6.2
P-L3:D	2.9M	6.1	8.3	7.3	5.9
P-L4:D	0.4M	4.9	7.8	5.7	6.0
P-L5:D	0.01M	4.6	7.7	5.7	6.1
B0	-	4.4	7.9	5.8	6.3



## Conclusions

Take-home

- ✓ no need for the side info during training, adaptation directly at test time;
- ✓ improvement of the separation, especially for underrepresented instruments;
- ✓ general approach that can be applied to other tasks and DL architectures.

However:

- X need at least a weak guiding signal;
- X the sources cannot be always activated.

Resources:

- https://github.com/giorgiacantisani/ugosa
- https://giorgiacantisani.github.io/projects/ugosa

Thank you for the attention!

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