



Deep learning for robust audio perception in human-robot interactions

Jean-Marc Odobez

IDIAP/EPFL Senior researcher – Head of PAU group

2019 ICRA workshop on

Sound Source Localization and its Applications for Robots



Switzerland



EPFL, Lausanne

Geneva

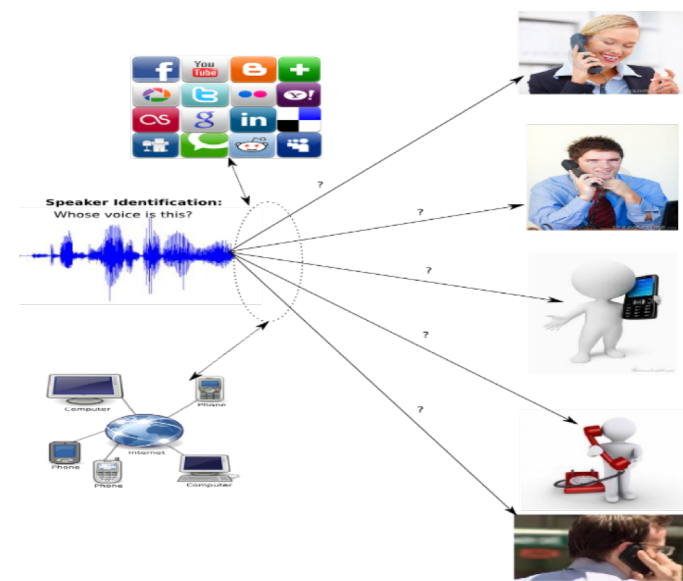
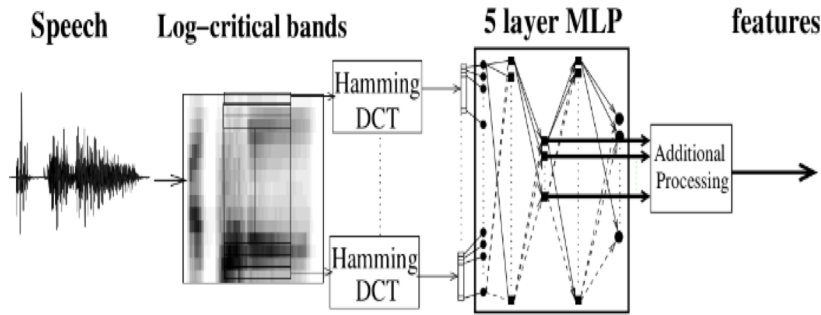
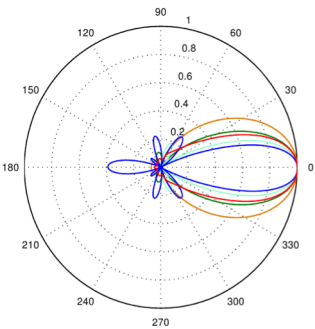


IDIAP Research Institute, in (very) brief

- Non for profit Foundation, created in 1991
 - academic affiliation with EPFL
- Human resources: around 100 people
 - 14 permanent researchers + 50 research associates (postdocs, PhD students) from more than 25 countries
- Main research areas - Artificial Intelligence for society
 - Machine Learning
 - Perceptual and Cognitive systems (speech, computer vision, natural language processing)
 - Human and Social Behavior (face-to-face communication, mobile, social media analysis,...)
 - Biometry
 - Robotics
 -



Speech group at Idiap



- Head: Prof. Hervé Bourlard
 - Researchers: Petr Motlicek, M. Magimai-Doss, Phil Garner
 - 25+ persons (researchers, phds, postdocs, interns, ...)
- Most speech related tasks
 - Forefront of Automatic Speech Recognition -- multilinguality
 - Speaker analysis (verification, identification, diarisation, role detection)
 - Microphone arrays and localization (beamforming, ad-hoc architectures)
 - Text-to-speech synthesis
 - Pathological speech processing
 - Speech assessment





PAU group research themes and objectives



Thematic : sensing, interpreting, understanding

- Perceptual component
extraction physical representations - detection, tracking, pose
- Activity understanding
gestures, behaviors - individual, group level - context
- Methods & Models
computer vision, (multimodal) signal processing, sociology
machine learning: statistical models; deep learning
- Applications
surveillance, human-robot interfaces, sociology, multimedia content analysis

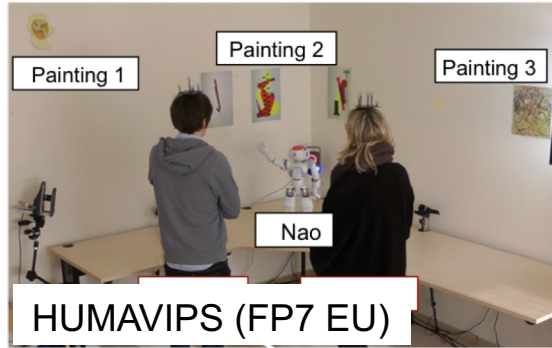




Interaction analysis



Idiap - KTH dataset



HUMAVIPS (FP7 EU)



MUMMER (H2020 EU)

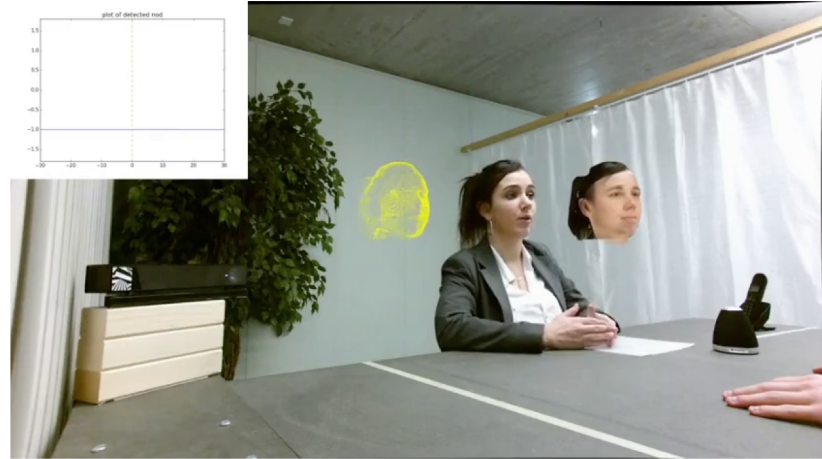
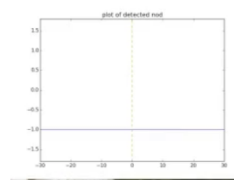
- Set-ups & Tasks (tracking, re-
id, non-verbal cues)

HeadFusion
360 degree head
tracking

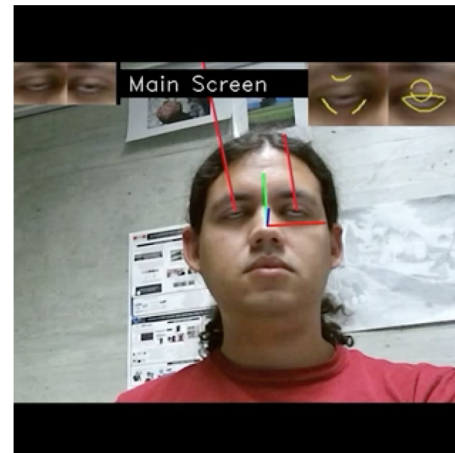
HeadFusion: 360° Head Pose Tracking combining
3D Morphable Model and 3D Reconstruction

Yu Yu, Kenneth Alberto Funes Mora, Jean-Marc Odobez
Idiap Research Institute and EPFL

Head gestures
(nods)



Gaze & attention



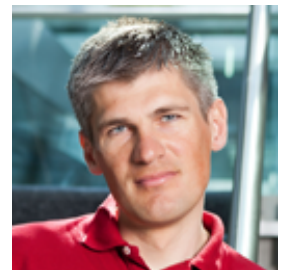
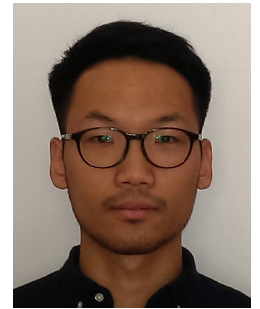


Deep learning for robust audio perception in human-robot interactions

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Joint work with

- Weipeng He (Phd student)
- Petr Motlicek (researcher)



Outline

- Joint sound source localization and discrimination with deep learning
- Multiple Sound source localization NN adaptation using weak labels



Interacting with robots : MuMMER EU project



- GOAL: Develop a humanoid robot
 - public shopping mall
 - entertaining, give information, directions
 - autonomous, natural interactions
- Participants
 - University of Glasgow (UK)
 - Heriott-Watt University (UK)
 - Idiap Research Institute (CH)
 - LAAS-CNRS (France)
 - Softbank Robotics Europe (France)
 - VTT Technical Research Center (Finland)
 - Ideapark (Finland)



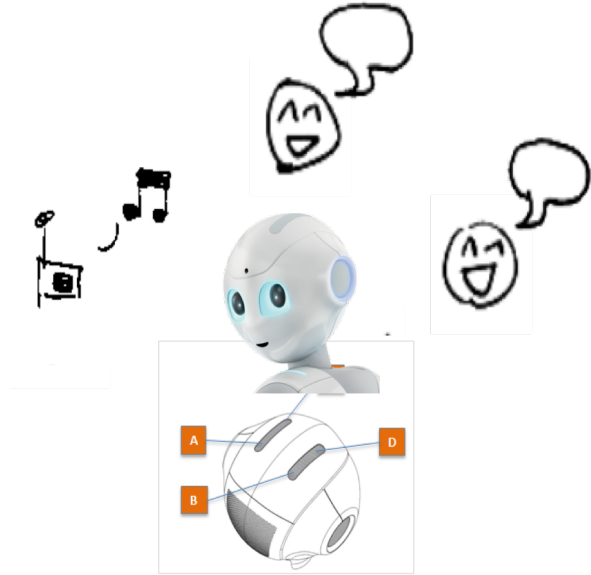
Sound source localization & discrimination



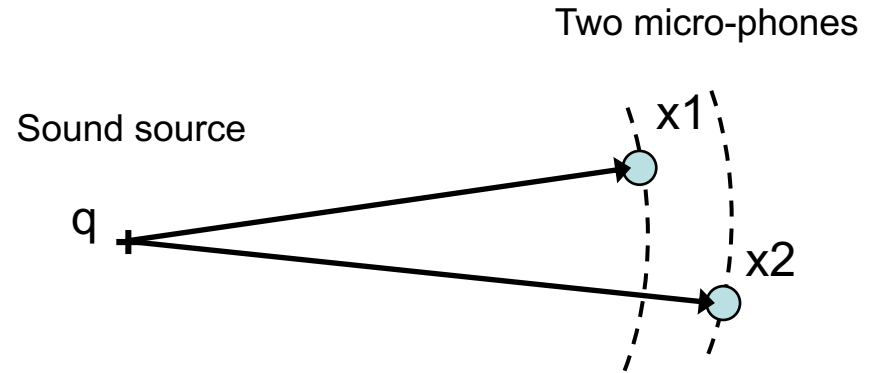
- Challenges
 - Unknown number of sound sources
 - Strong noise (robot ego-noise, background)
 - Speech and non-speech sources
 - Speech overlap (simultaneous speakers)
 - Short utterances during interactions



Sound source localization



Micro-phone array



$$R_{12}(\tau) = \frac{1}{2\pi} \int \Psi_{12}(\omega) X_1(\omega) X_2(\omega)^* e^{j\omega\tau} d\omega$$

GCC-PHAT

- Traditional approaches (localization) : signal processing
 - Interaural time and intensity differences
 - Time difference of arrival (TDOA)
 - E.G. : GCC-PHAT: Generalized Cross-Correlation with Phase Transform

=> relies on modeling assumptions

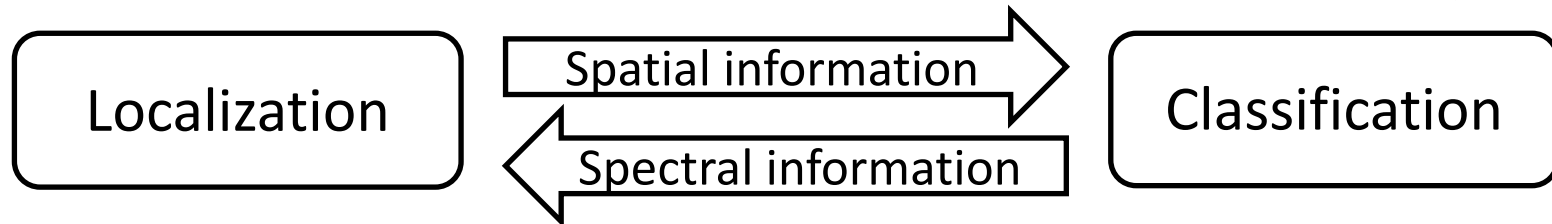
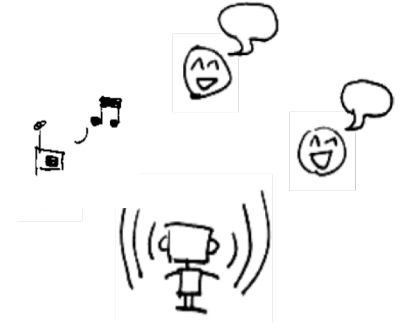
(head model, geometry knowledge, obstacles, propagation, ...)

Sound source discrimination



- Previous methods : solve the problem sequentially
- Issues:
 - beamforming : enhances the signal coming from a given direction
 - direction is approximately known
 - different audio representation/processing for localization and classification
 - related: which signal frequencies comes from which direction ?

Sound source localization & discrimination

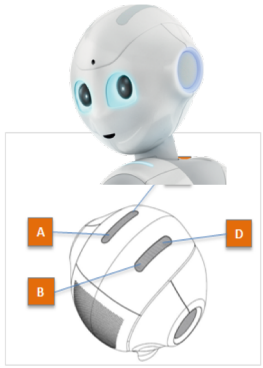


- Proposition: learning-based joint localization & discrimination
 - both tasks help each other
 - fewer assumption required
 - direct optimization for the tasks

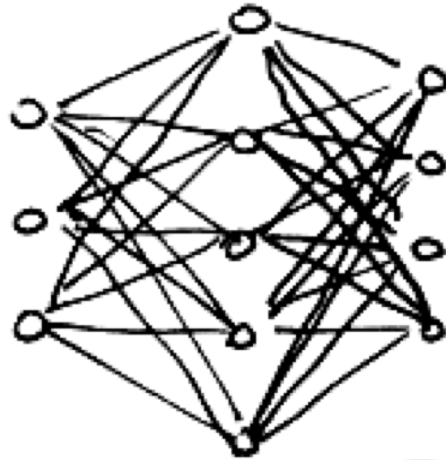
[Deep Neural Networks for Multiple Speaker Detection and Localization](#), He, Motlicek, Odobez, Int. Conference on Robotics and Automation (ICRA) 2018

[Joint Localization and Classification of Multiple Sound Sources Using a Multi-task Neural Network](#), He, Motlicek, Odobez, Interspeech 2018

Sound source localization & discrimination



Input



Architecture



Output



SSL : Sound source Localization

SNS : Speech/Non speech discrimination

Training



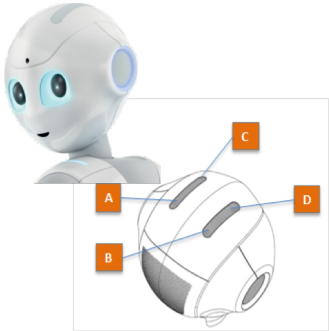
Training data



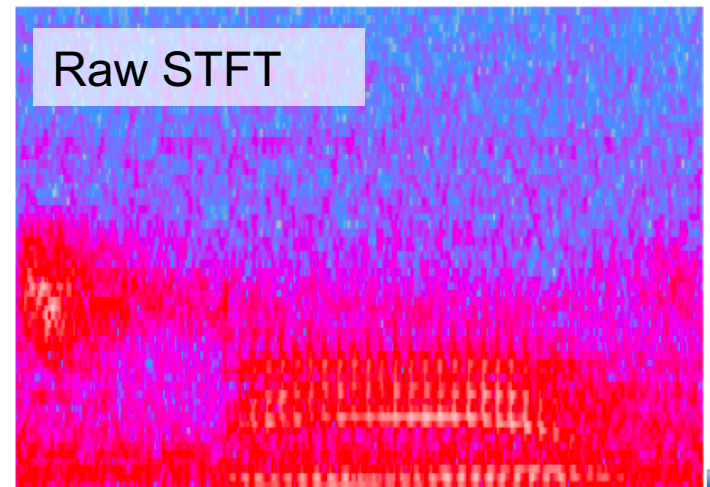
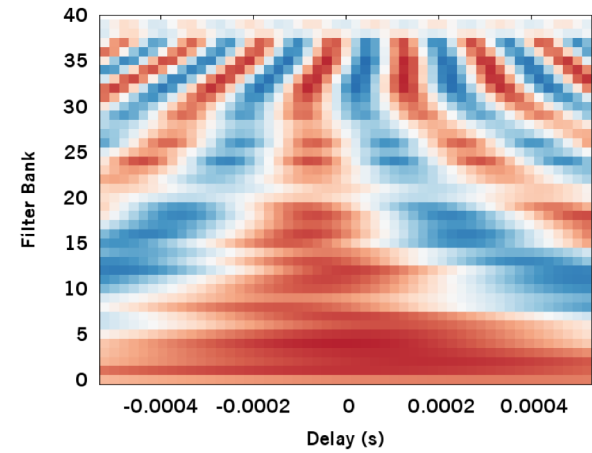
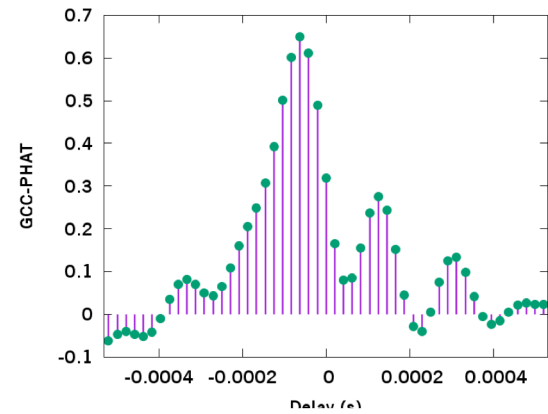
- How to proceed?



SSL & SNS : Input

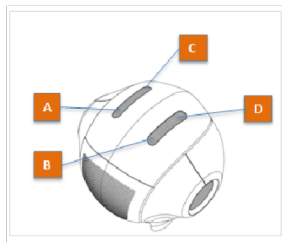


- Input signals
- Per **pair of** microphone (6)
 - GCC-PHAT delay coefficients
 - GCC-PHAT on filter banks
 - Ok for localization
 - Lacks spectral information for SNS
- Short-Time Fourier transform (SFTF) per microphone

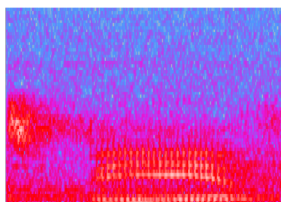


SSL & SNS : Output & Loss Function

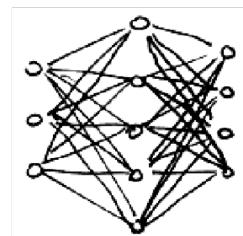
4-channel audio



Raw STFT

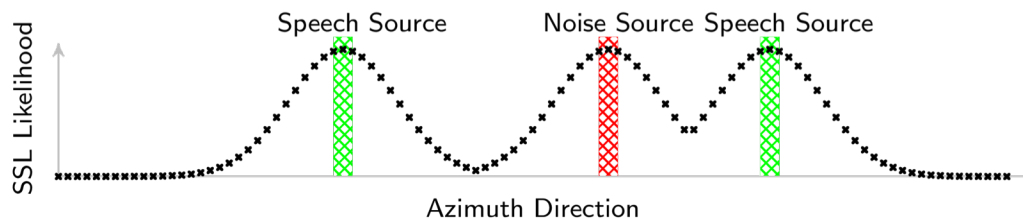


Neural Network

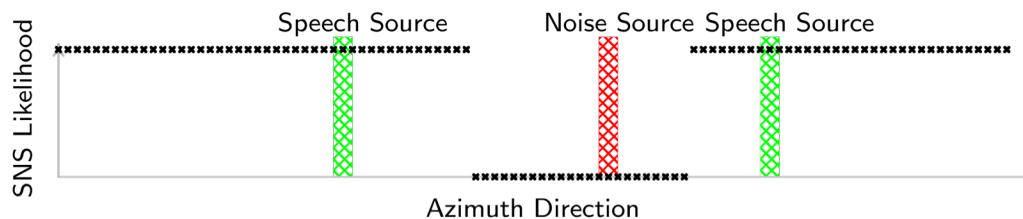


- Likelihood for each sound direction

p :



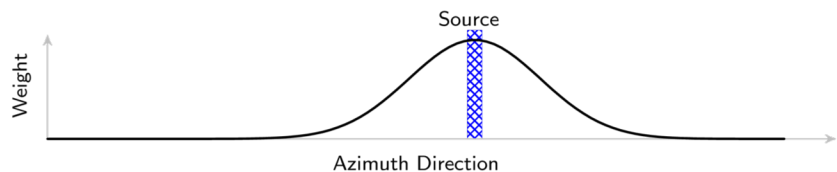
q :



$$Loss = \|\hat{p} - p\|_2^2 + \sum_t w_i |\hat{q}_i - q_i|^2$$

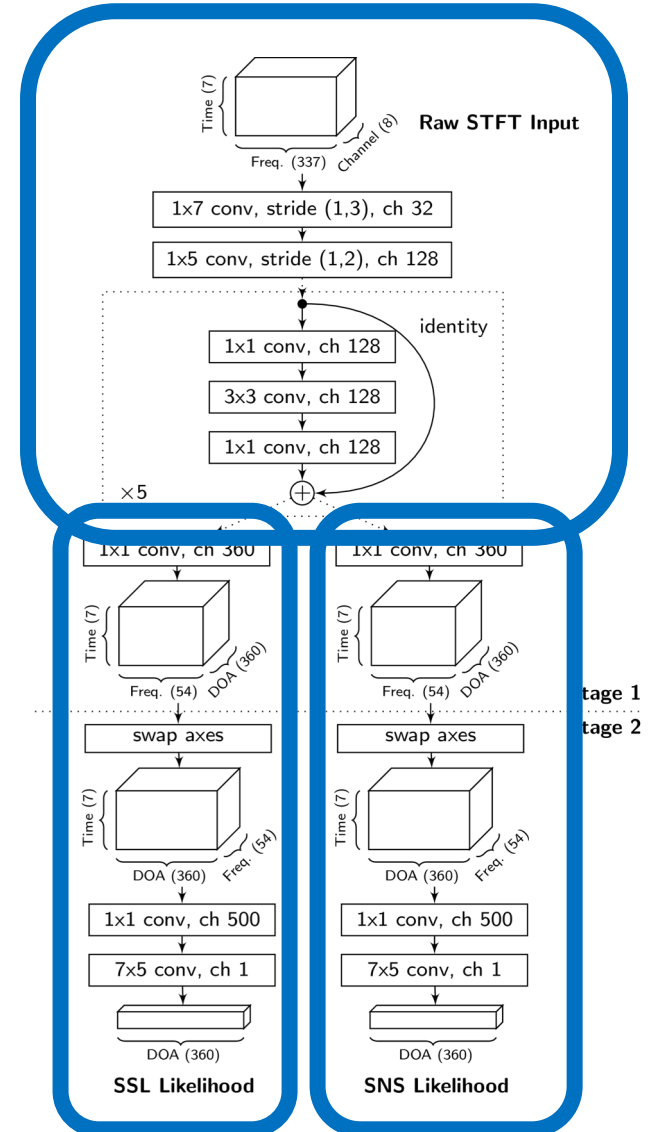
emphasis on directions next to active sources

w :



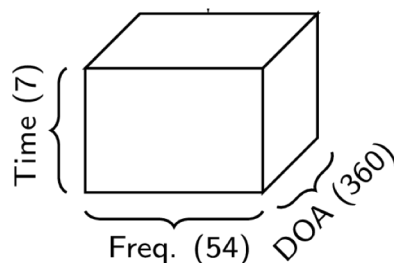
SSL & SNS : Network structure

- Fully convolutional
 - Residual network trunk
 - Two task-specific branches



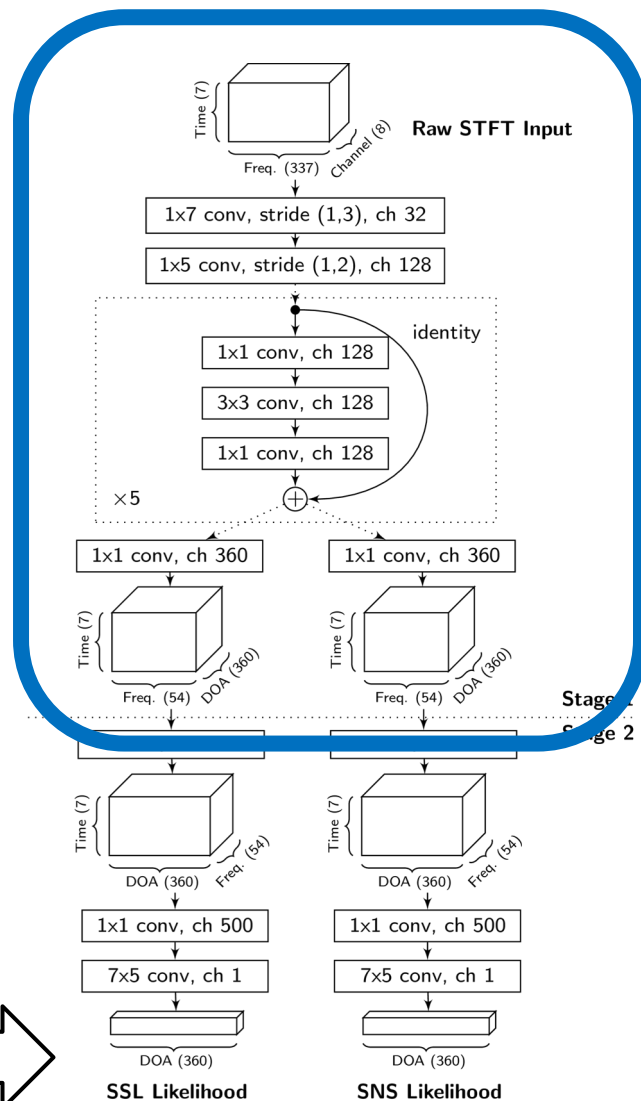
SSL & SNS : Network training

- End-to-end
 - Not working well
- Two-stage approach
 - Stage 1
 - Convolutions in Time-Frequency domain



- Output: 360 (DOA) channels
- early SSL & SNS prediction for each TF point

Train whole network end-to-end

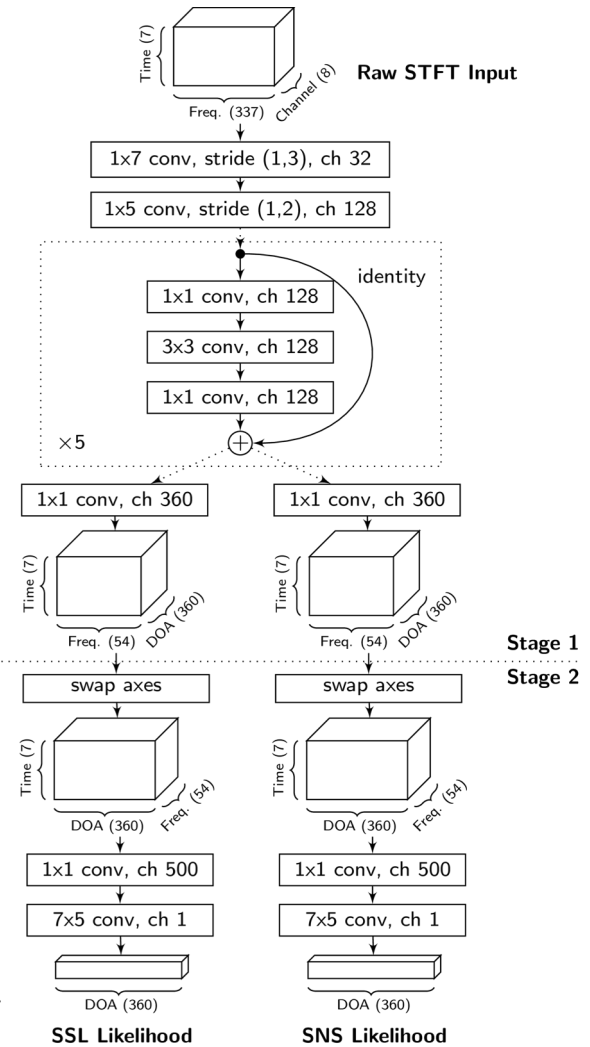


SSL & SNS : Network training

- End-to-end
 - Not working well
- Two-stage approach

(1) Supervision on output of stage 1

(2) Train whole network end-to-end

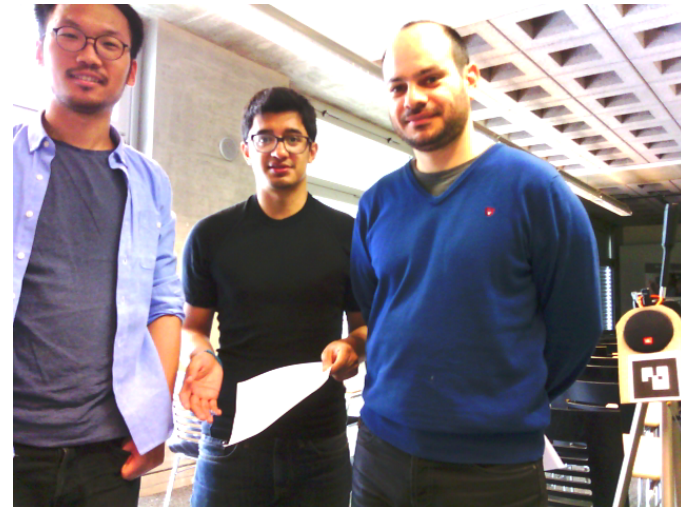


SSL & SNS : Experiments

- Training
 - Loudspeakers: 32 hours, 148 speakers
 - Speech: AMI Corpus
 - Non-speech: Google AudioSet
 - **Pepper moves to collect data with variabilities**
=> faster data-collection



- Test
 - Loudspeaker: 17 hours, 16 (different) speakers
 - Human talkers: 8 minutes, 7 speakers
(with loudspeakers Non Speech sources)
- Sound Source Localization for Robots (SSLR) Dataset

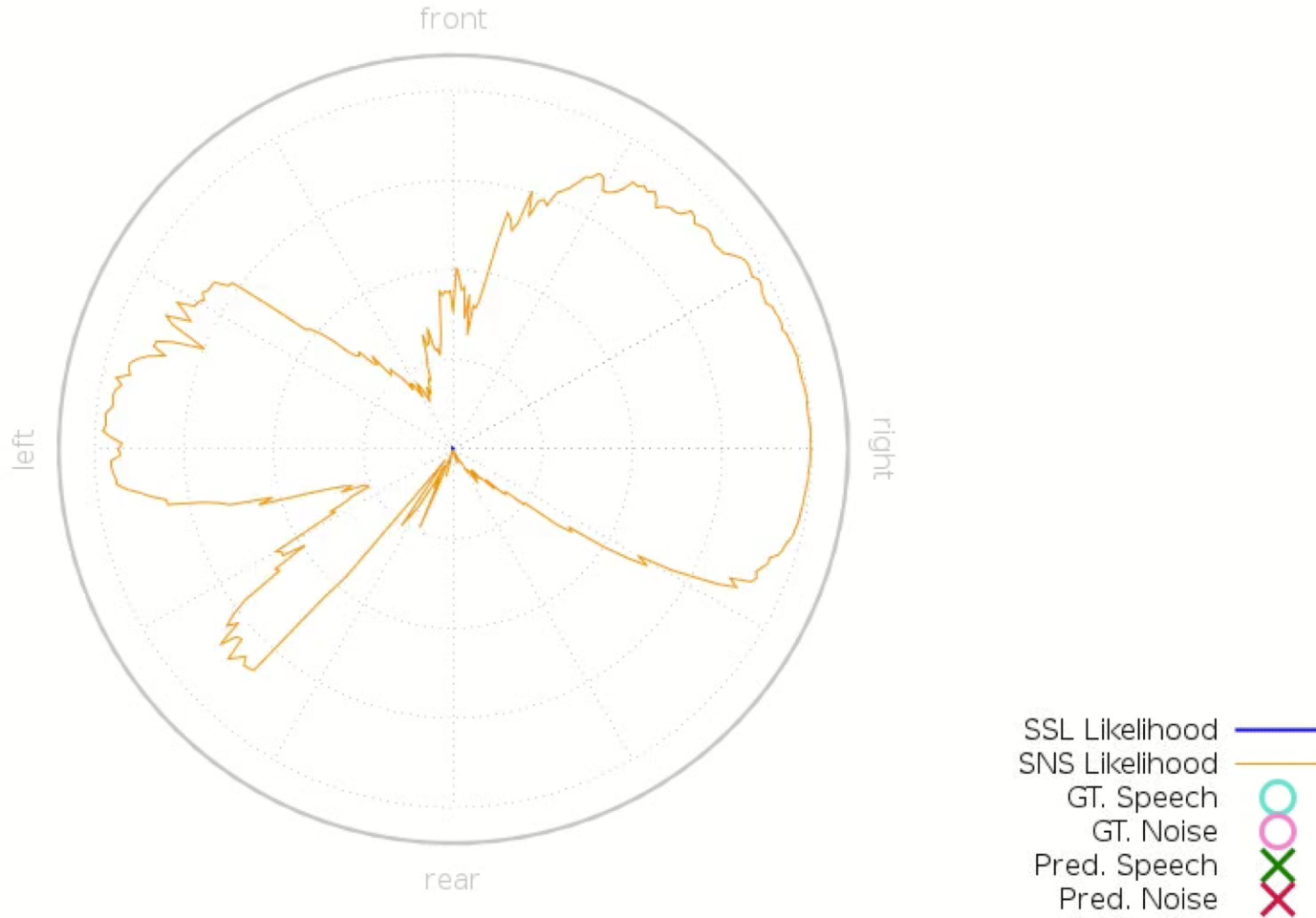


<https://www.idiap.ch/dataset/sslr>

AMI Corpus: <http://groups.inf.ed.ac.uk/ami/corpus/>
AudioSet: <https://research.google.com/audioset/>

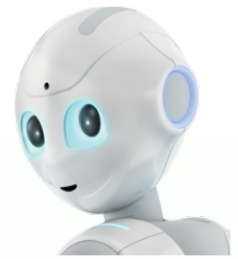
SSL & SNS : Experiments – qualitative results

Method MTNN-CTX; Time 0.00s; Frame #000000







- Loudspeaker recording

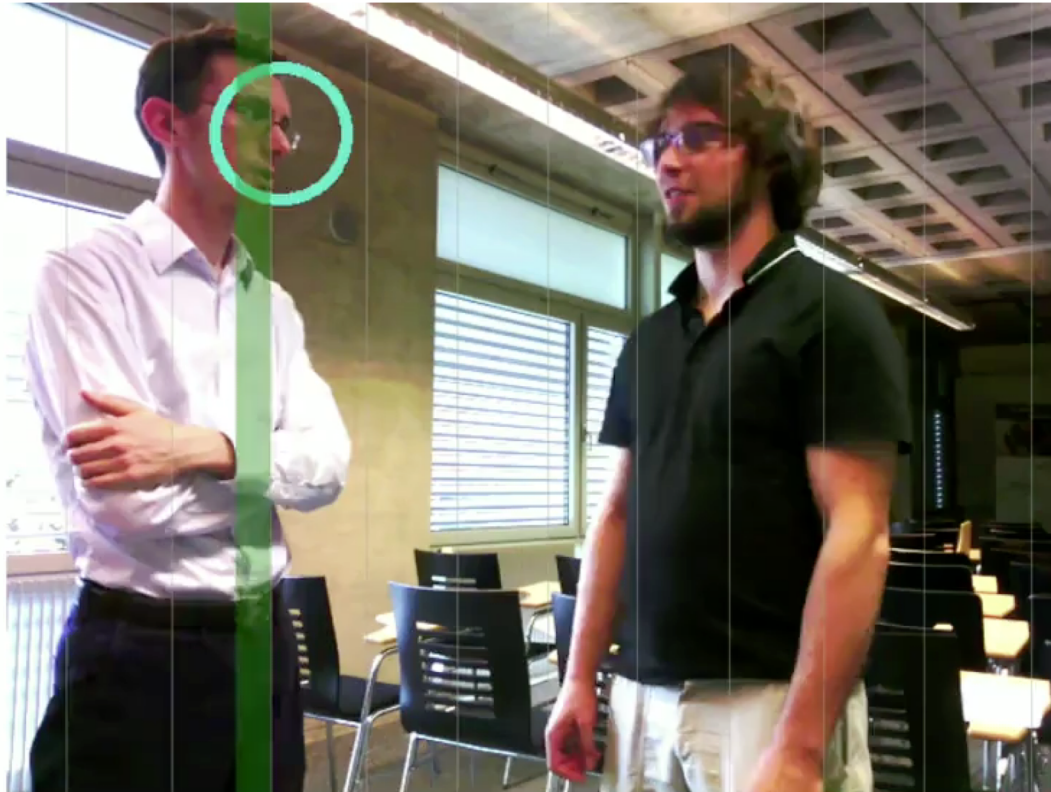
SSL & SNS : Experiments - qualitative



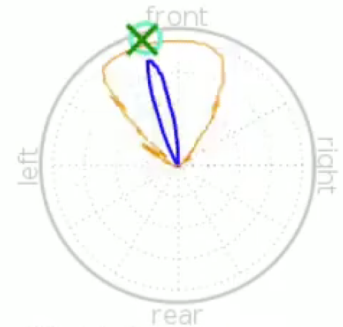
Method: MTNN-CTX

File: s5_01

- G.T. Speech: 
- G.T. Noise: 
- Pred. Speech: 
- Pred. Noise: 



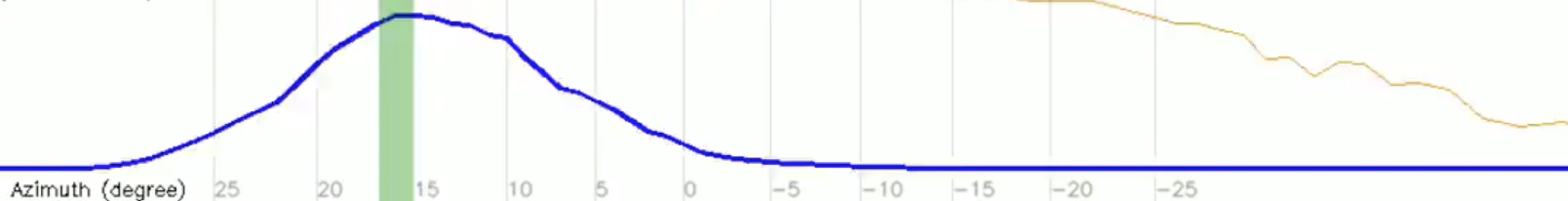
Top view:



- SSL Likelihood 
- SNS Likelihood 
- GT. Speech 
- GT. Noise 
- Pred. Speech 
- Pred. Noise 

Time: 1.991s

Output value (likelihood):



- Blue curve: likelihood of a sound source
- Yellow curve: is the source speech (1) or not (0)

SSL & SNS : Experiments – evaluation

- Tasks

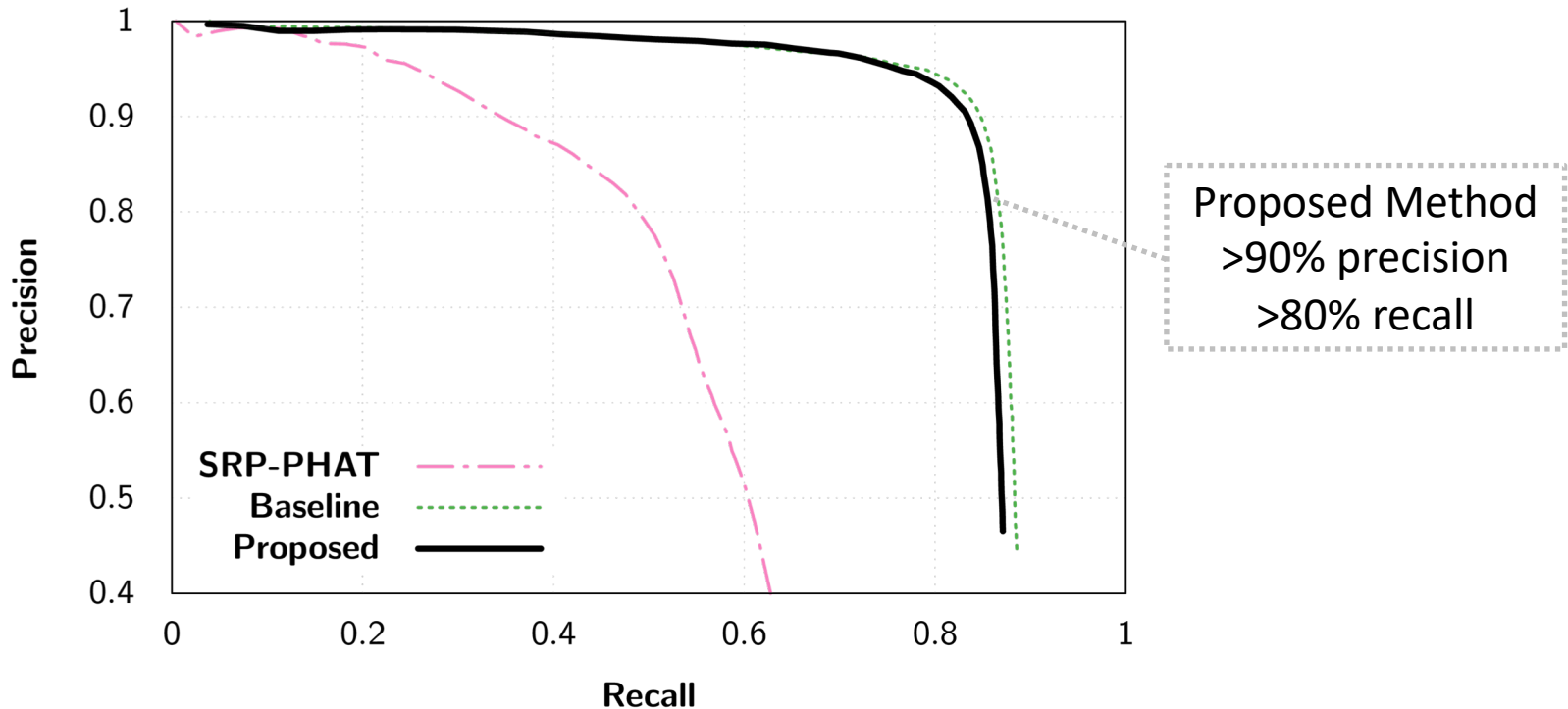
- Sound Localization
- Speech/Non-speech Classification
- Speech Localization

- Methods

- Baseline: two-step approach
 - localization NN
 - MVDR + classification (NN on beamformed signal)
- Proposed method

SSL & SNS : Experiments – sound localization

- Human recordings



(Similar conclusion can be drawn for loudspeaker recordings)

SSL & SNS : Experiments – speech/non-speech

(Assuming sound direction is known)

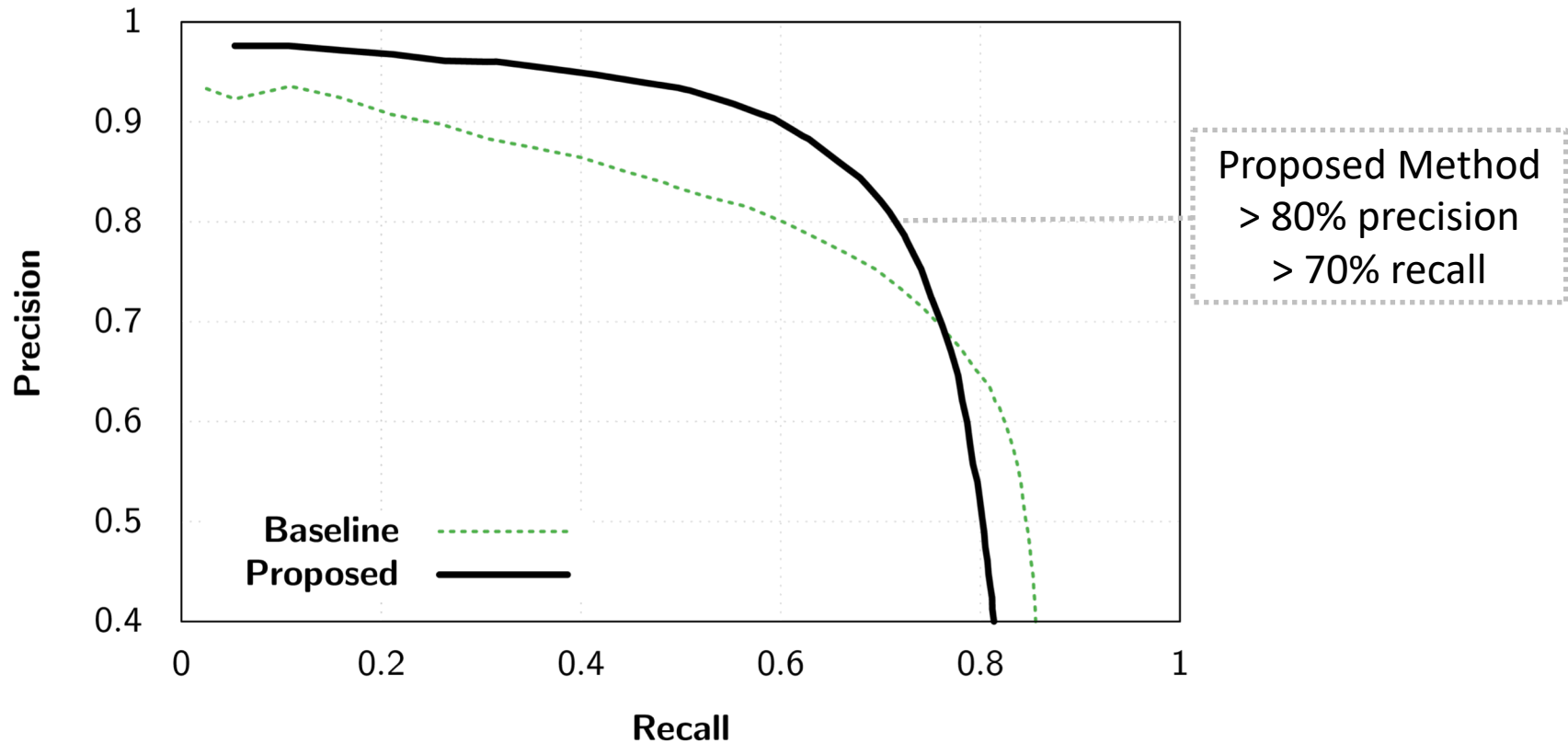
| | Accuracy | Loudspeaker | Human |
|------------------------|----------|-------------|-------------|
| Baseline | | 0.80 | 0.68 |
| Proposed Method | | 0.95 | 0.85 |

- Proposed method
 - much better
 - good generalization



SSL & SNS : Experiments – speech localization

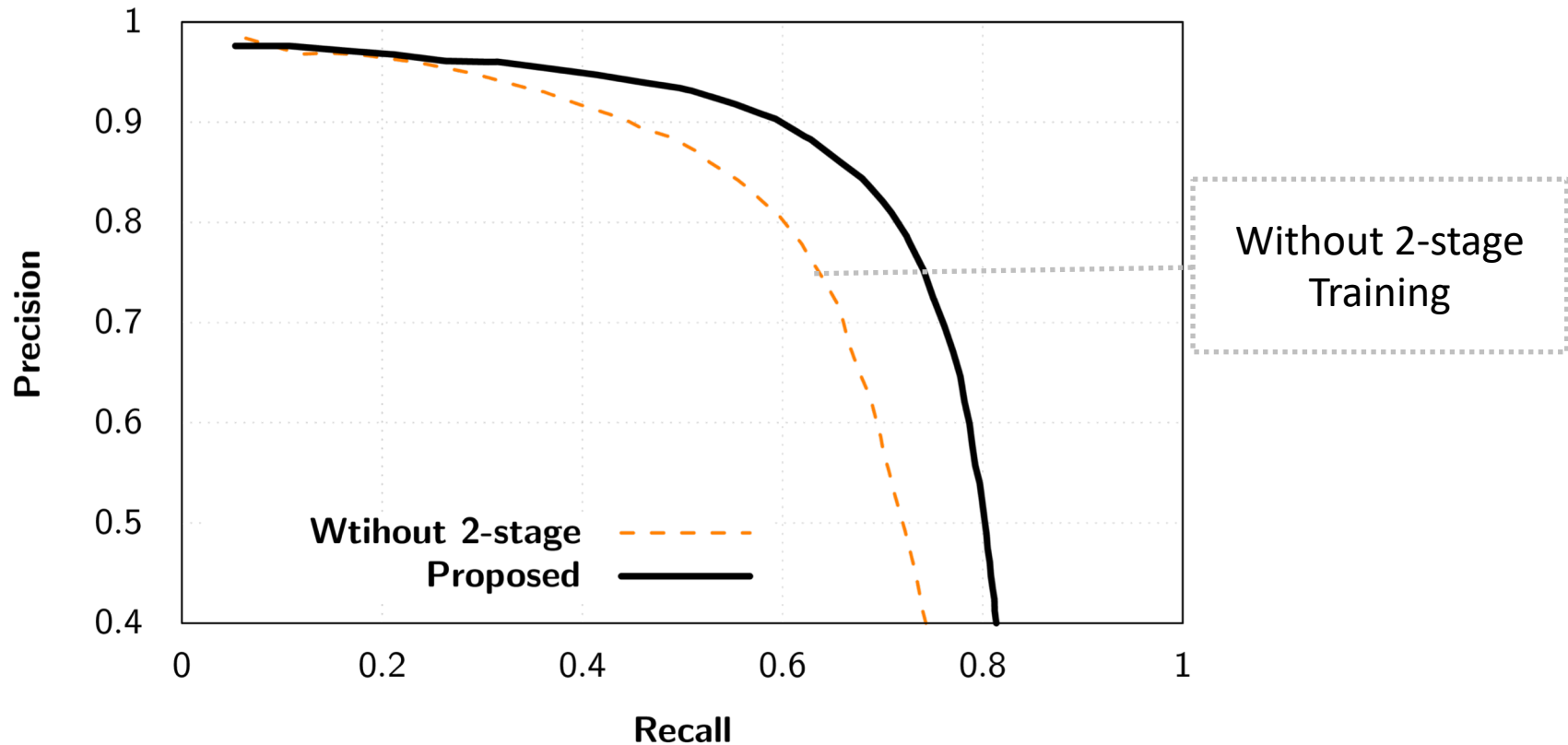
- Human recordings



(Similar conclusion can be drawn for loudspeaker recordings)

SSL & SNS : Experiments – speech localization

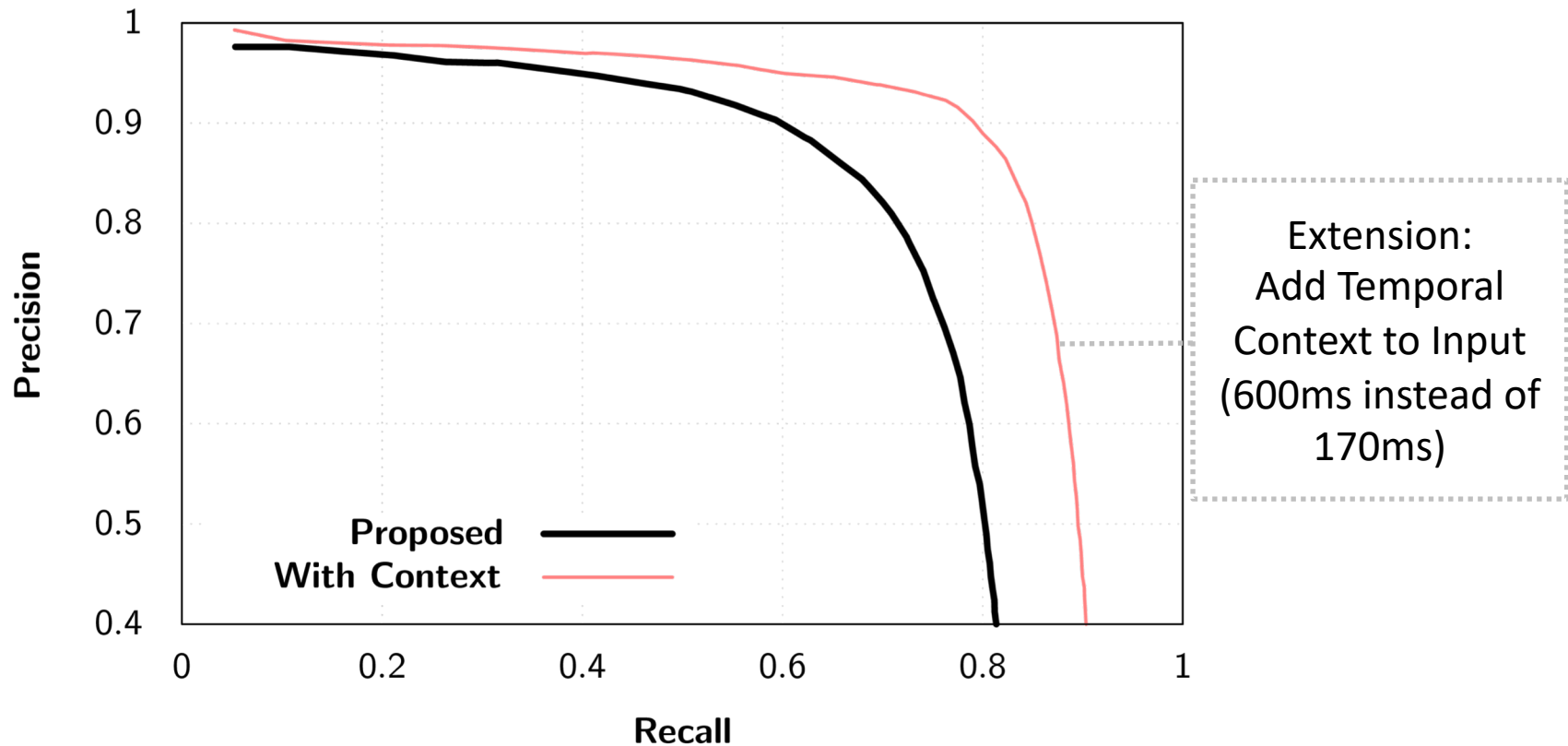
- Human recordings



(Similar conclusion can be drawn for loudspeaker recordings)

SSL & SNS : Experiments – speech localization

- Human recordings



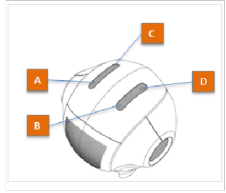
(Similar conclusion can be drawn for loudspeaker recordings)

Outline

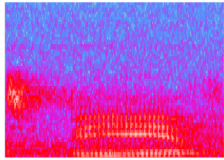
- Joint sound source localization and discrimination with deep learning
- Multiple Sound source localization NN adaptation using weak labels

Learning to localize sound

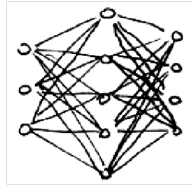
4-channel audio



Raw STFT



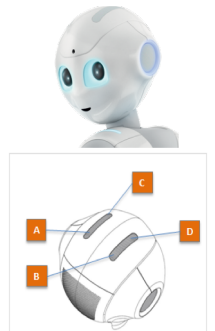
Neural Network



Output: spatial likelihood



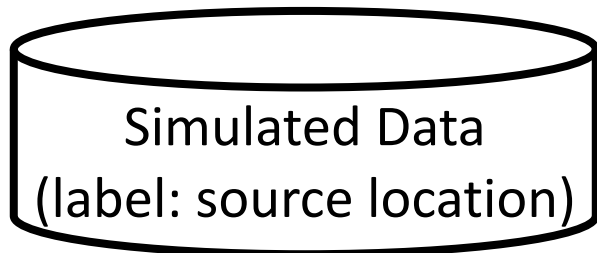
- Current issue - training data
 - diversity in source signals (voices, noise), power, positions, noise, etc.
 - **device specific**
 - collection and annotation can be costly
- Approach & motivations
 - train network **with simulated data** –apply to real data?
 - control diversity, exploit large datasets
 - **reality gap** : mismatch between simulation & real conditions
 - device physical body & microphone response pattern, room features,...



We need domain adaptation !

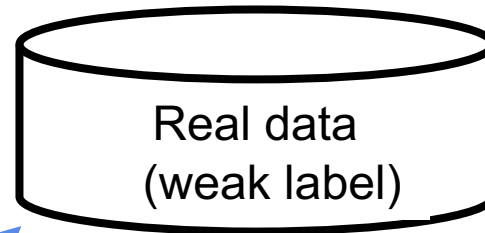
Domain adaptation – problem formulation

- Source domain

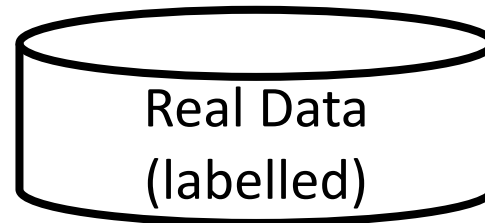


Training Set

- Target domain



Adaptation Set



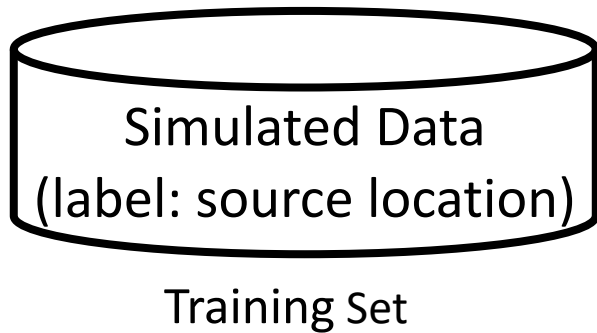
Application/Test Set

No label

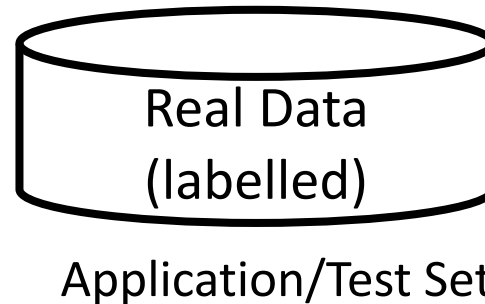
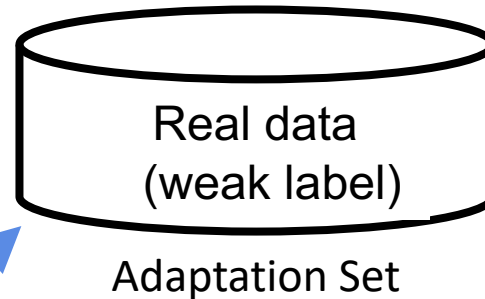
- Domain adversarial training
(feature level)

Domain adaptation – problem formulation

- Source domain



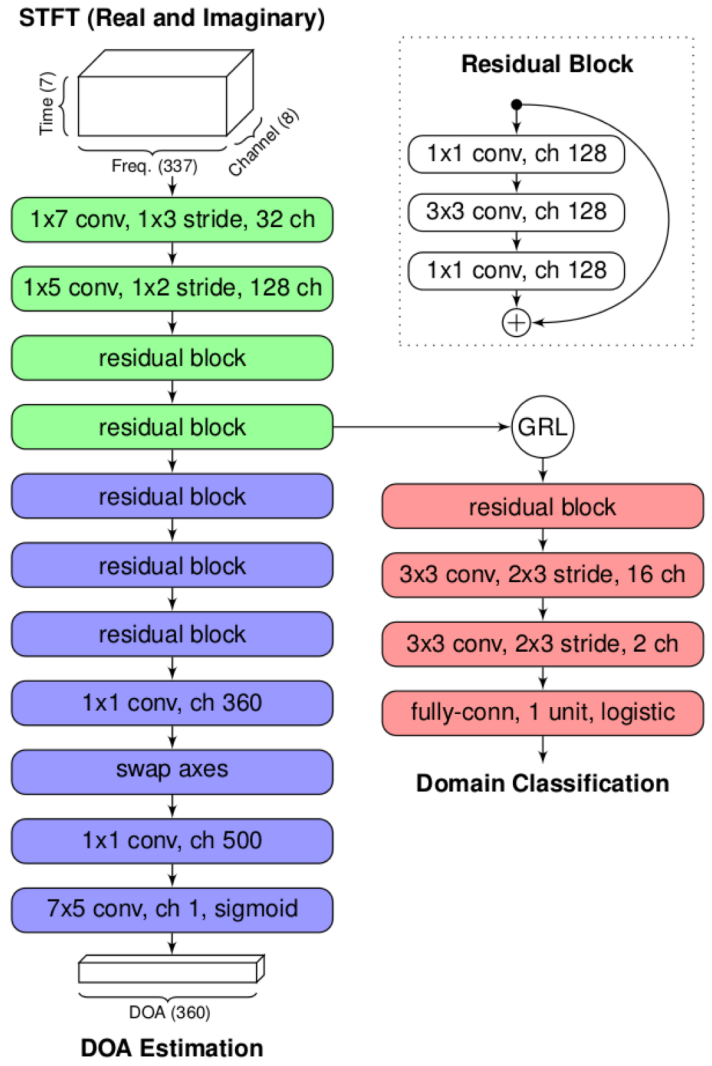
- Target domain



- Weak labels: number of sources
- Relevant information
 - Easy annotation

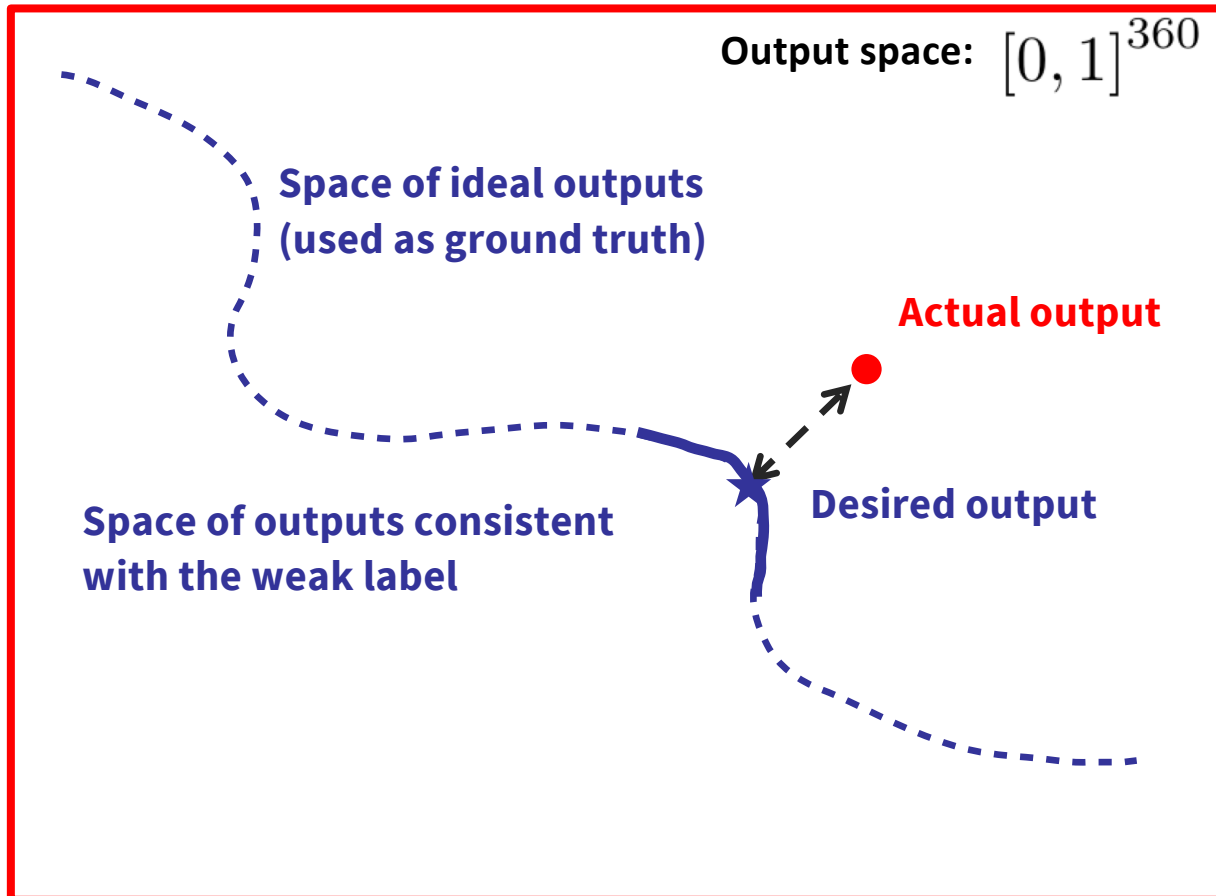
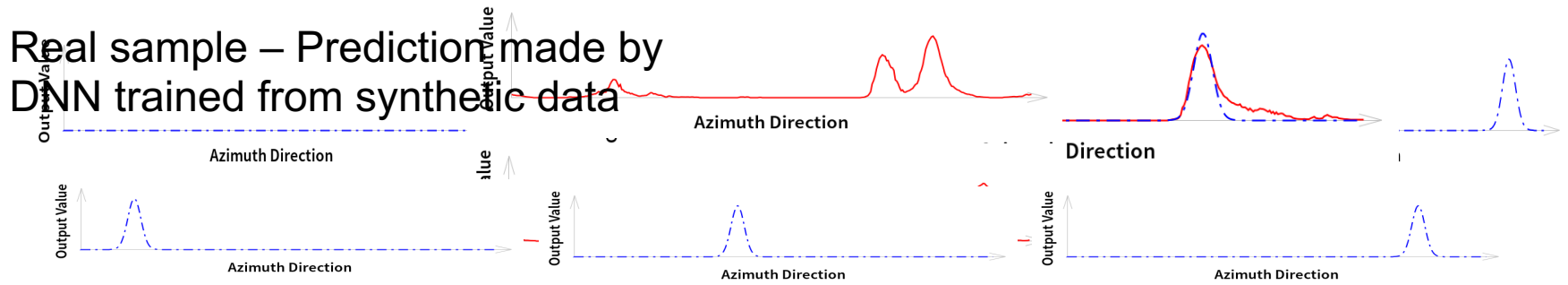
Domain adaptation – domain adversarial training

- Goal: learn features (green part)
 - Perform well for the task (purple part)
(on simulated data)
 - Are domain independent (red part)
- => domain classifier can not distinguish those produced from the real or simulated data



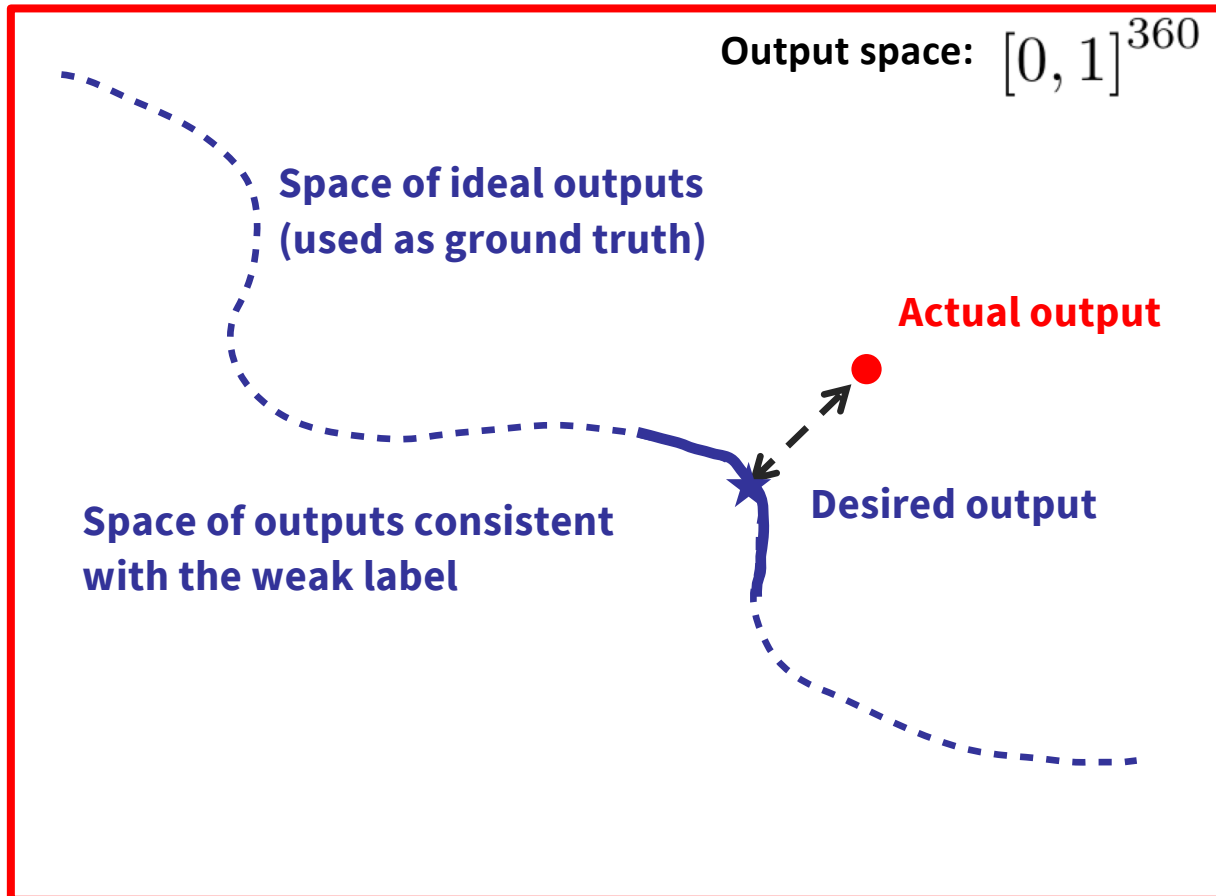
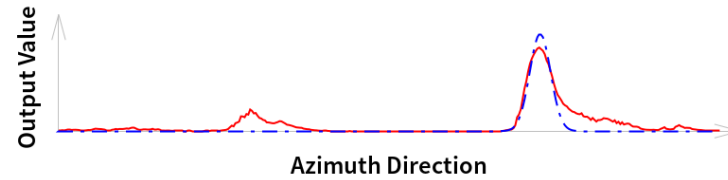
Ganin et al. "Domain-adversarial training of neural networks," Journal of Machine Learning Research, 2016.

Domain adaptation – weak supervision

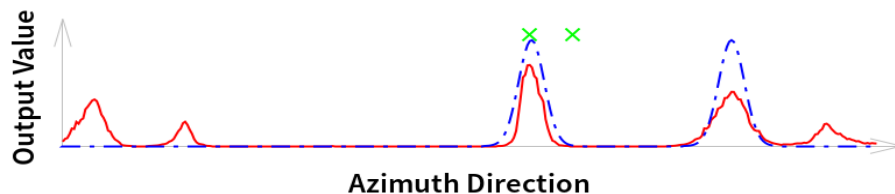
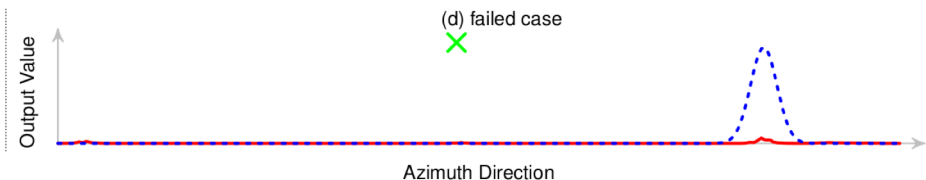
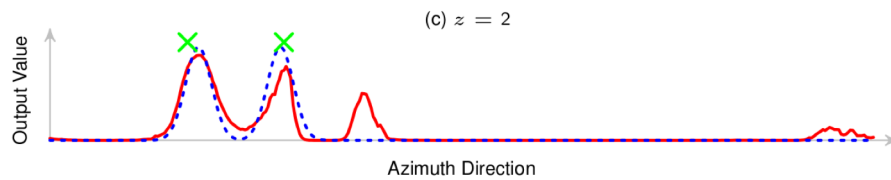
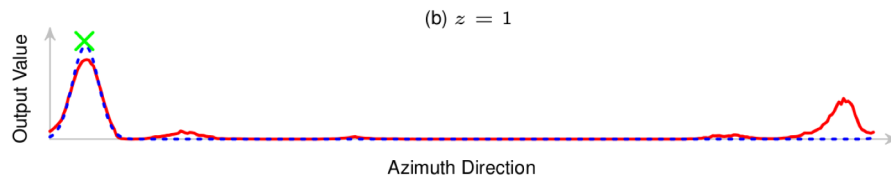
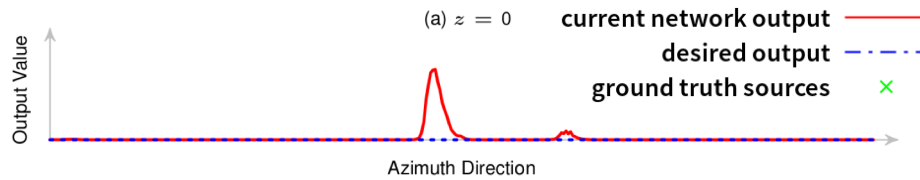


Domain adaptation – weak supervision

Real sample – Prediction made by DNN trained from synthetic data



Domain adaptation – weak supervision



• Adaptation

- real samples
- generate outputs from the network trained from synthetic data
- collect desired outputs using the weak label information

⇒ fine-tune the network with the (real sample, desired output) dataset

Experiments

- Data : clean segment from the AMI corpus
 - Source domain: simulation with RIR generator (several rooms & reverberation coefficients)
 - Target domain: real data (loudspeaker + robot)



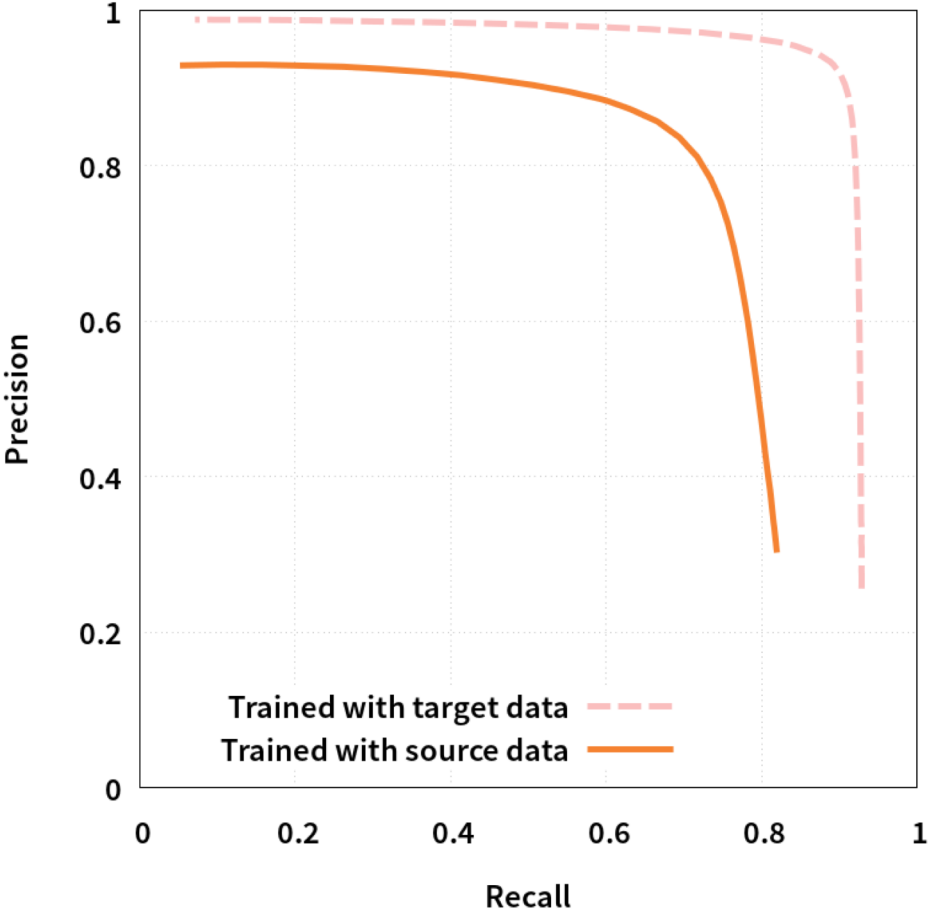
- Evaluation
 - Frames of 170ms
 - Maximum 2 sources
 - Correct detection: error < 5 degrees

RIR: Room-Impulse-Response simulator .

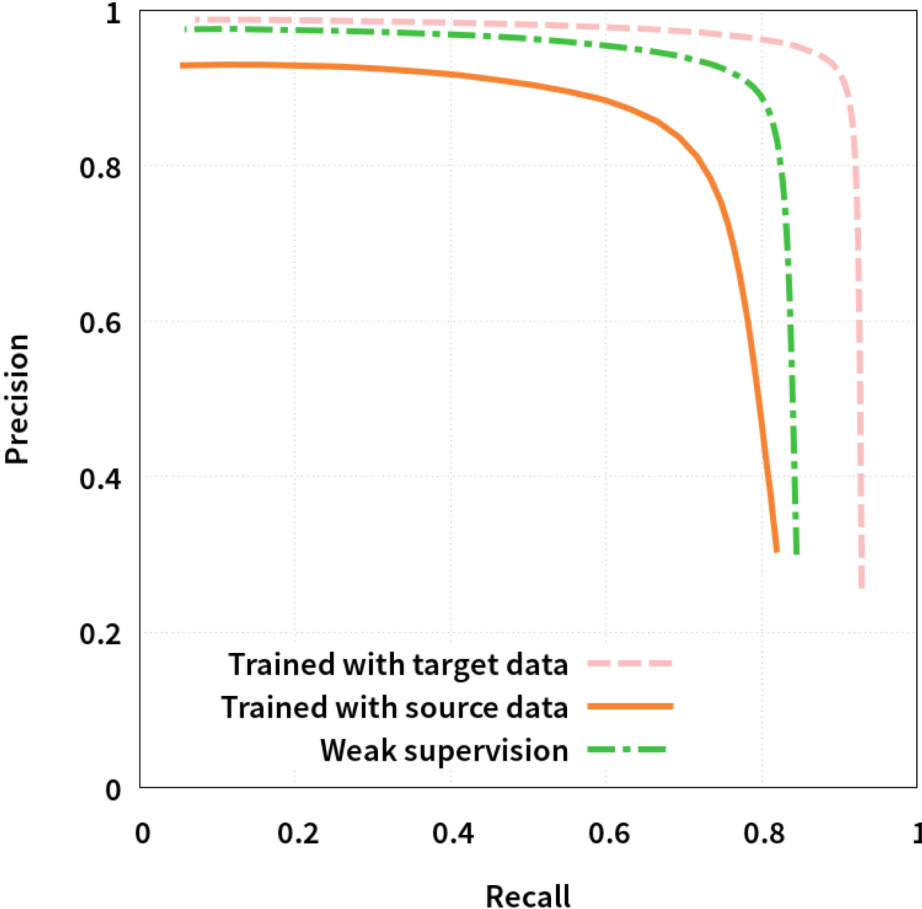
J.B. Allen and D.A. Berkley, "Image method for efficiently simulating small-room acoustics,"
Journal Acoustic Society of America, 65(4), April 1979, p 943.



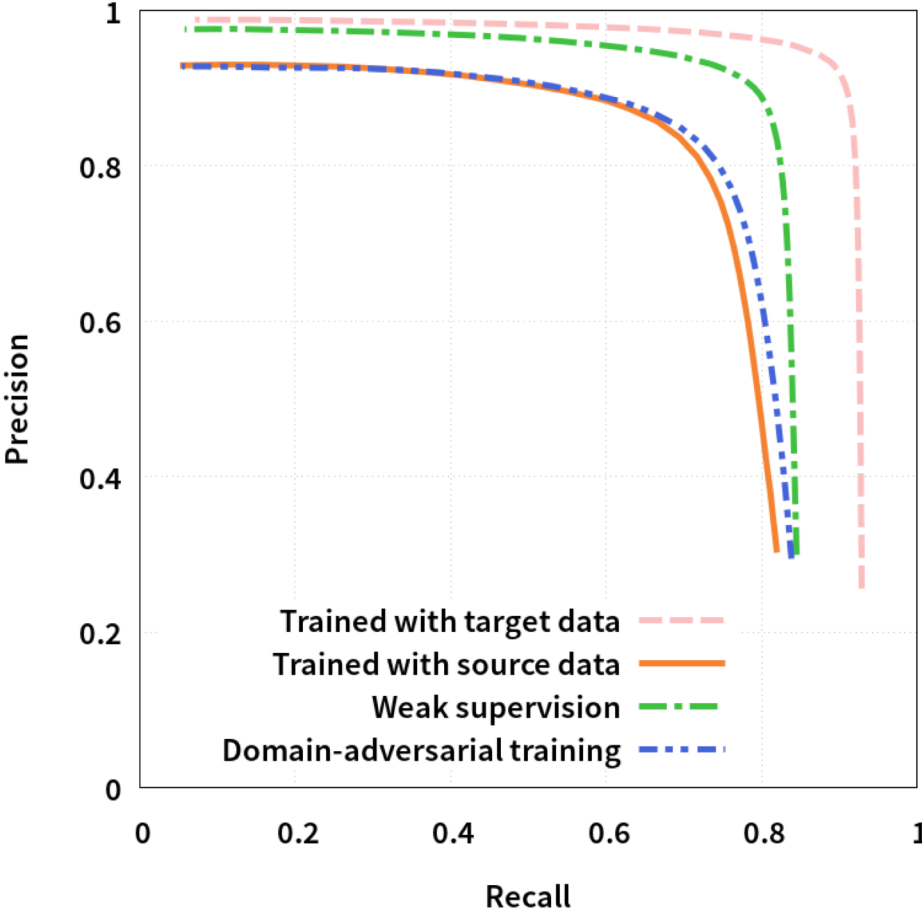
Results – supervised training



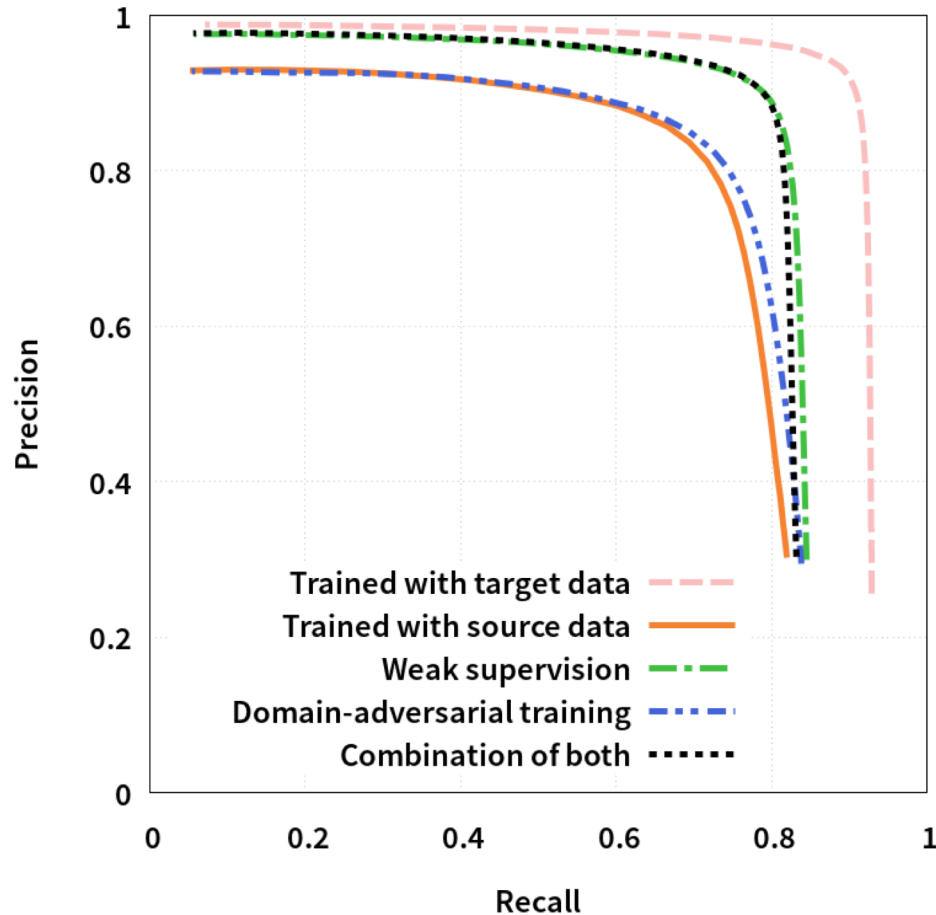
Results – weak supervision



Results - domain adversarial



Results – weak + adversarial combined



- Adversarial training
 - difficult to find the trade-off between domain-invariance and discriminative power

SLS & SNS : conclusions

- Deep learning-based methods
 - Tasks: Sound localization ---- **joint sound localization and classification**
 - **Likelihood** output **encoding** : easy handling of multiple sources
 - **Two-stage training**: adding intermediate supervision
 - **Robot embodiment**: simplifies training data collection
 - Easy addition of temporal context
 - Significant better performance compared to baselines
- Domain adaptation (**quick generalization to other devices**): synthetic to real domain
 - **Weak supervision** (known number of sources) => significant improvement
 - Domain-adversarial training fails to yield significant results
- Next steps
 - Curriculum learning (Done => closes the reality gap)
 - Working on **joint localization & voice embedding**
 - Better simulators

Thanks for your Attention ! Questions ?