THE ANDREW & ERNA VITERBI FACULTY OF ELECTRICAL ENGINEERING

SIPL
Signal and Image Processing Lab

TECHNION
Israel Institute of Technology

Work in Progress

Deep Learning Based Target Cancellation for Speech Dereverberation

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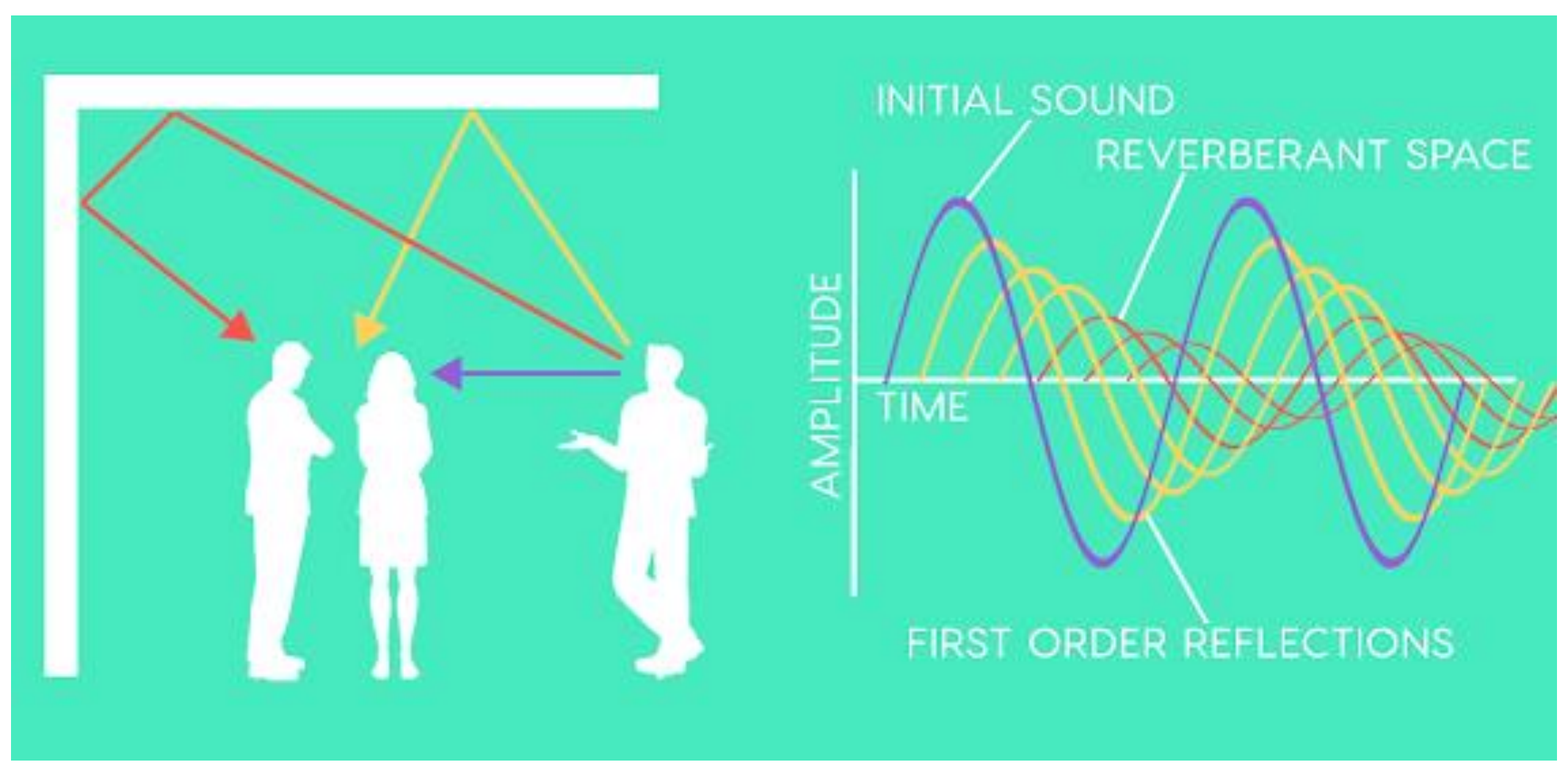
Introduction

- Reverberation is the process of multi-path propagation of a sound from its source to a receiver
- Reverberation reduces speech intelligibility, can cause hearing aids to malfunction, impairs performance of speech recognition and more…



The process of reverberation

Project Goals

- Dereverberation of speech signals using deep learning methods
  - Single channel – start with a solution that works for reverberations recorded by a single microphone
  - Dual channel – expand the solution to two microphones, placed at a certain configuration
  - Real-time – the process of dereverberation should not cause a distinguishable delay

Challenges

- Existing DNN-based solutions are usually complex and lack code, making the results hard to reproduce
- Implementation of an end-to-end DNN-based solution from scratch

Dataset

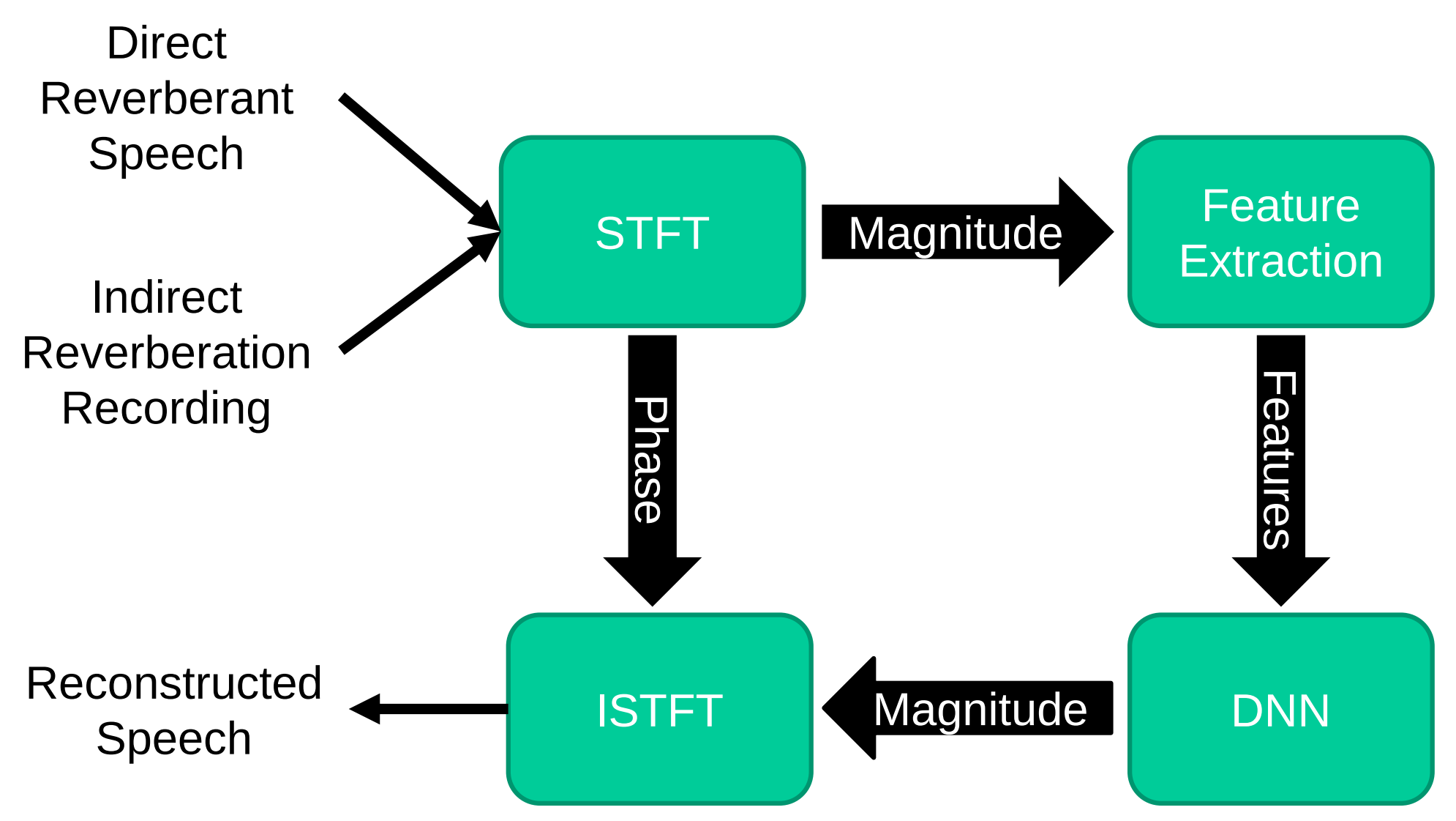
- TIMIT Corpus - 6300 speech recordings carried by 630 speakers from 8 different American dialect regions

Sentence Type	#Sentences	#Speakers	Total	#Sentences per Speaker
Dialect (SA)	2	630	1260	2
Compact (SX)	450	7	3150	5
Diverse (SI)	1890	1	1890	3
Total	2342		6300	10

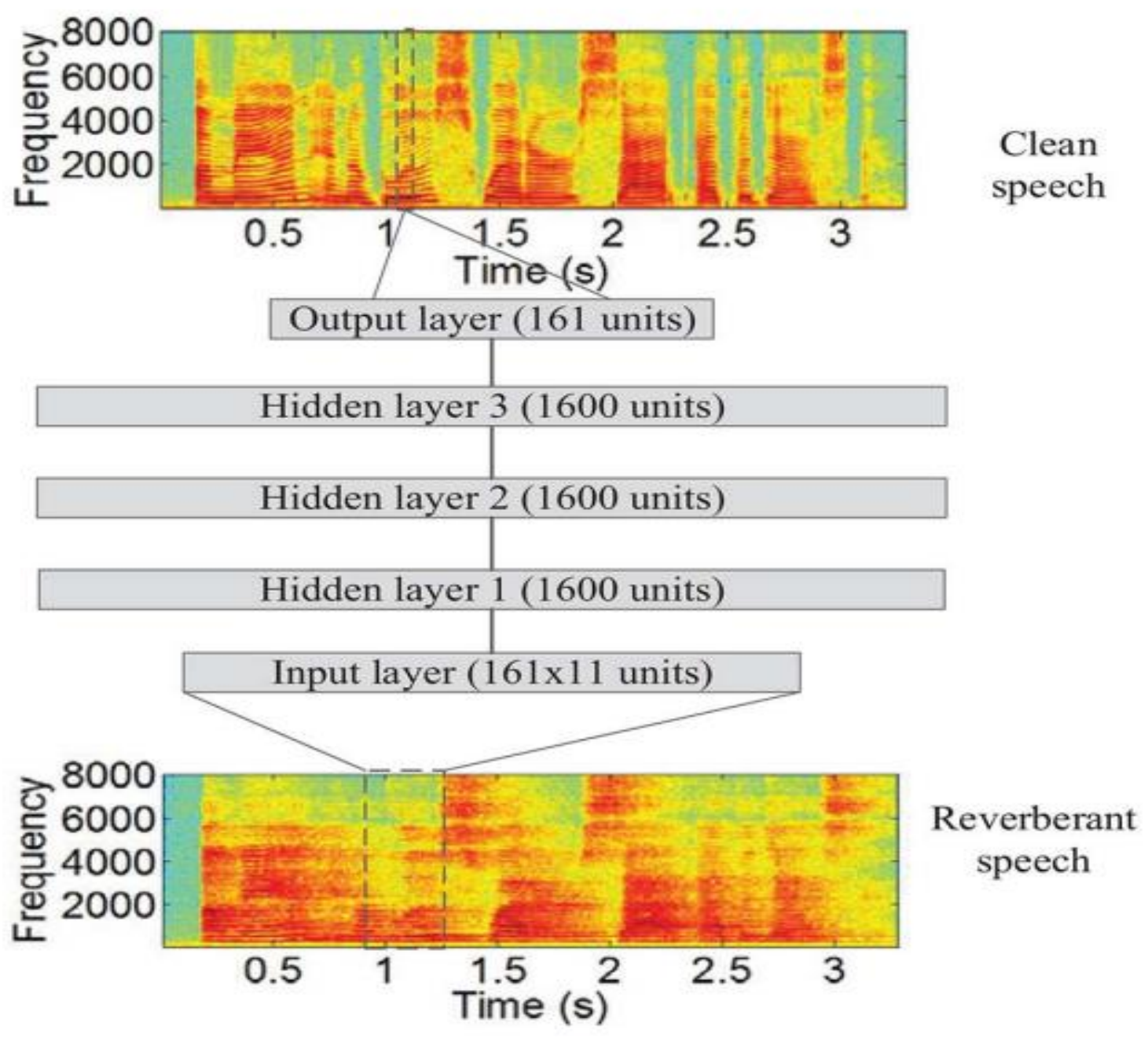
Reverberant Speech

- Simulated using Room Impulse Response Generator by [Jarrett et al., 2012]
- Simulation of 2 different rooms and 3 different reverberation times (T60) – 0.3, 0.6, 0.9 sec for training and 0.2, 0.5, 0.8 sec for testing

Chosen Solution

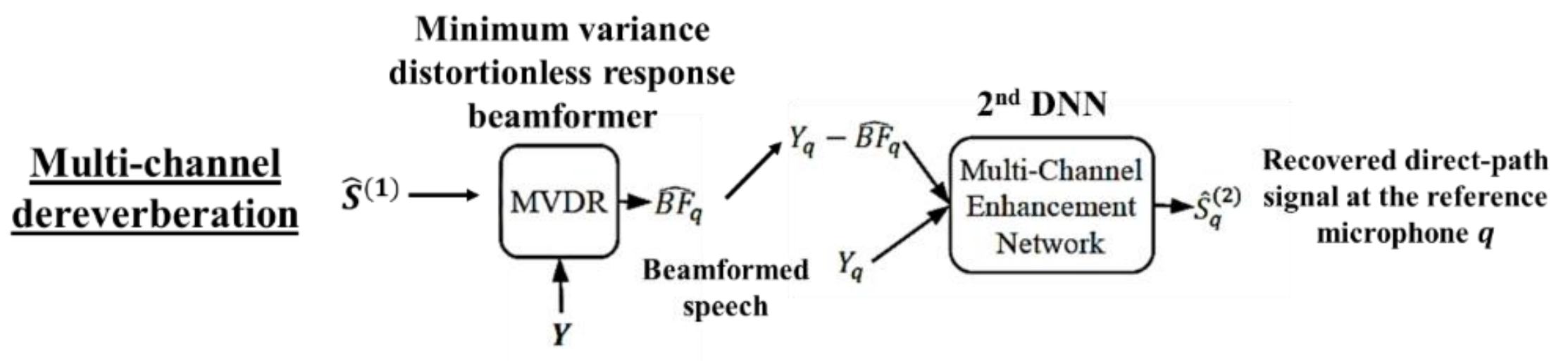


- STFT domain input-output
- Feed Forward DNN based on [Han et al.,2015]


Frequency
8000
6000
4000
2000
0.5
1
1.5
2
2.5
3
Time (s)
Clean speech
Output layer (161 units)
Hidden layer 3 (1600 units)
Hidden layer 2 (1600 units)
Hidden layer 1 (1600 units)
Input layer (161x11 units)
Reverberant speech

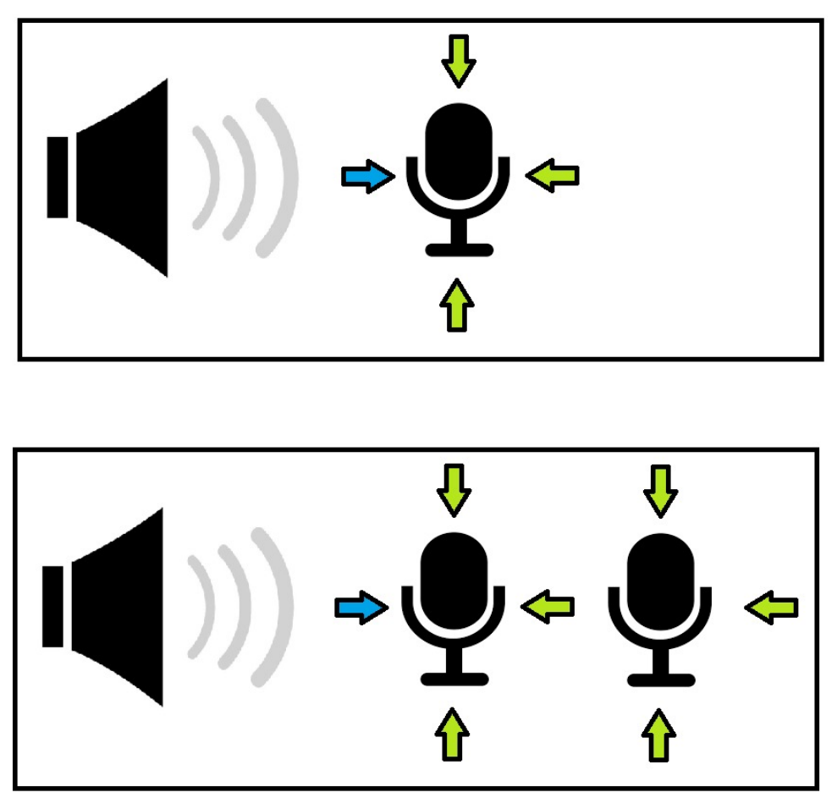
Dual Channel Approach

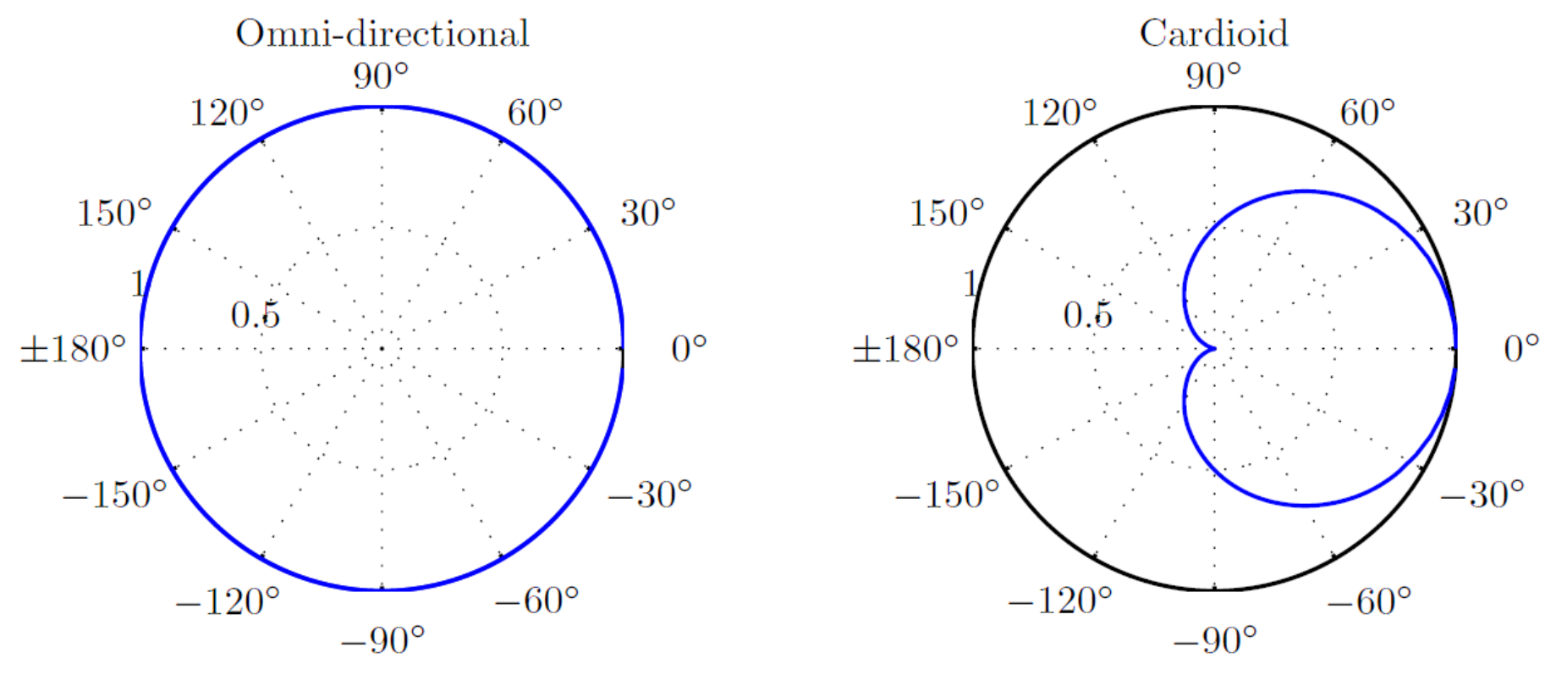
- It is possible to enhance performance by feeding the DNN a 2nd input – consisting of reverberations only (without the direct path signal)
- This is usually done by a beamformer which utilizes a microphone array to derive the reverberations


Multi-channel dereverberation
$\hat{S}^{(1)}$
Minimum variance distortionless response beamformer
MVDR
Y
$\widehat{BF}_q$
Beamformed speech
$Y_q - \widehat{BF}_q$
Y_q
2nd DNN
Multi-Channel Enhancement Network
$\hat{S}_q^{(2)}$
Recovered direct-path signal at the reference microphone q

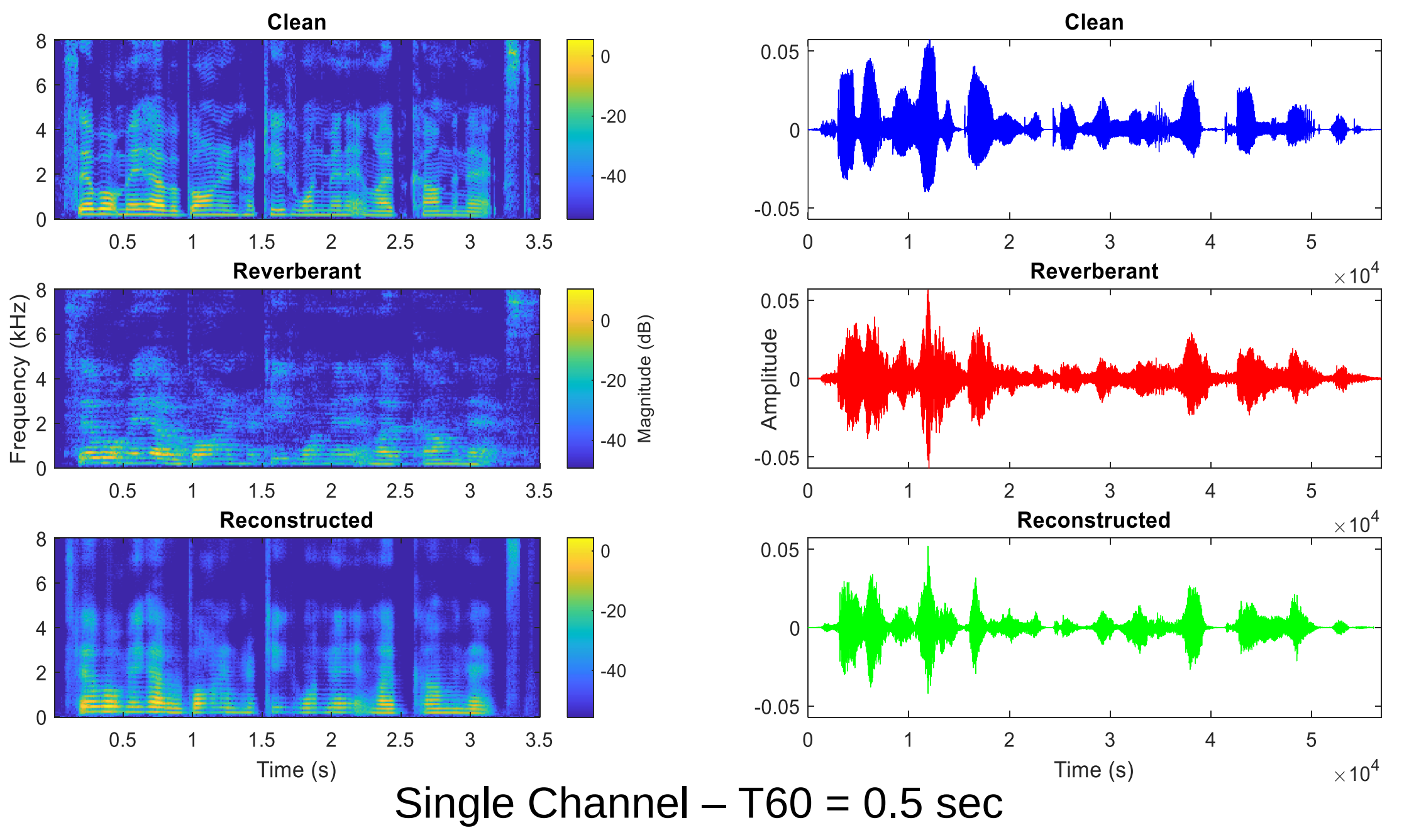
Solution by [Zhong-Qiu Wang et al., 2020]

- We propose to replace the microphone array by adding a single microphone which is directed at the opposite direction of the speaker, hence “hears” only reverberations

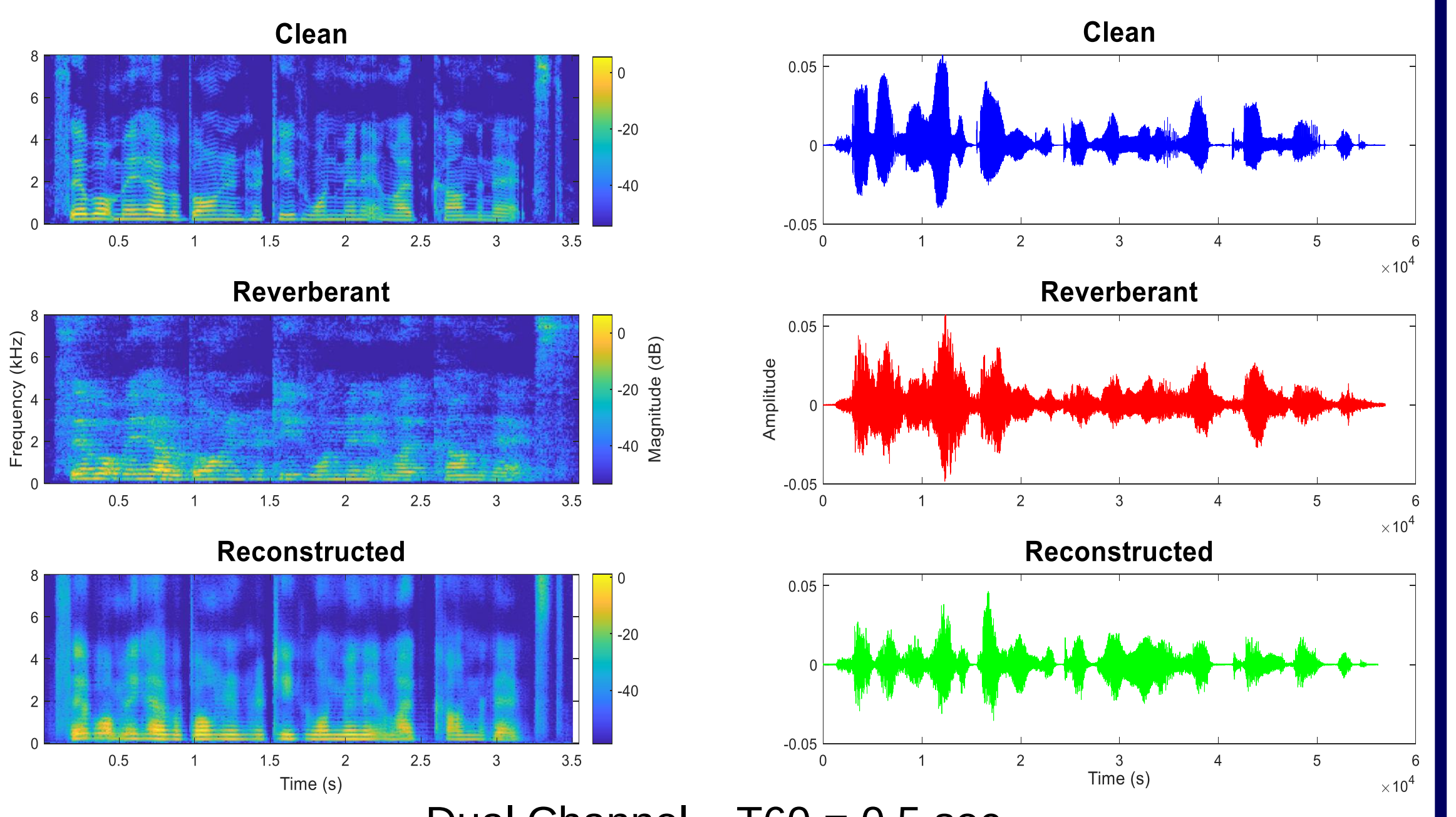




Results



Single Channel – T60 = 0.5 sec



Dual Channel – T60 = 0.5 sec

Speech Quality

Clean		T60 = 0.2 sec	T60 = 0.5 sec	T60 = 0.8 sec
3.77	Rev	3.71	3.10	3.03
	Rec - Single	3.06	2.62	2.65
	Rec – Dual	2.99	2.58	2.53

DNSMOS Score – Speech Quality Metric – Higher is better

Spectral Distance

	T60 = 0.2 sec	T60 = 0.5 sec	T60 = 0.8 sec
Rev	3.29	4.56	5.10
Rec - Single	2.97	3.47	3.93
Rec – Dual	3.09	3.24	3.34

Mean distance between spectrograms – Lower is better

Conclusions

- Successful removal of reverberations, at a cost of decreasing speech quality caused by phase – magnitude inconsistency
- The dual channel approach:
  - Insufficient results
  - Should be tested in more complex architectures
- Work in progress:
  - Pseudo-phase sensitive loss functions for training
  - Advanced STFT reconstruction – HiFi-GAN [Kong et al., 2020]

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