

Multi-channel Source Separation 2

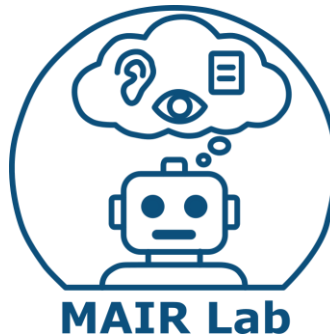
안인규 (Inkyu An)

Speech And Audio Recognition
(오디오 음성인식)

<https://mairlab-km.github.io/>

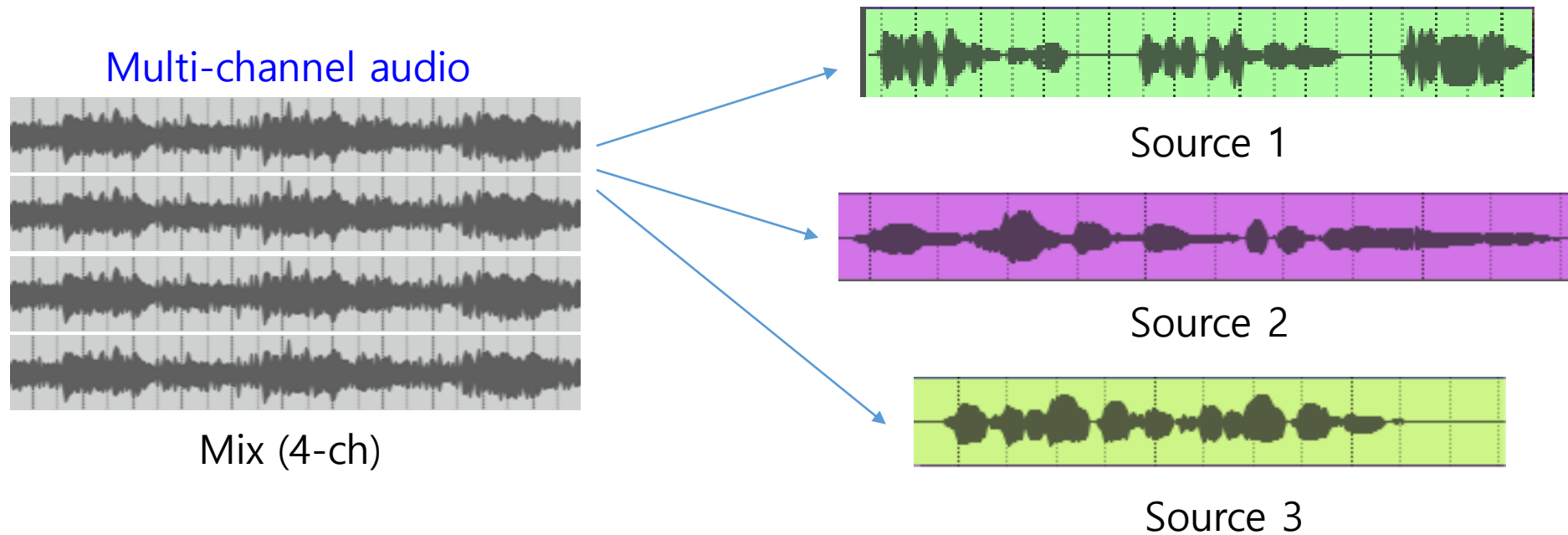


This lecture material refers to
https://github.com/yandexdataschool/speech_course?tab=readme-ov-file and
<https://github.com/markovka17/dla>



Multi-channel Source Separation

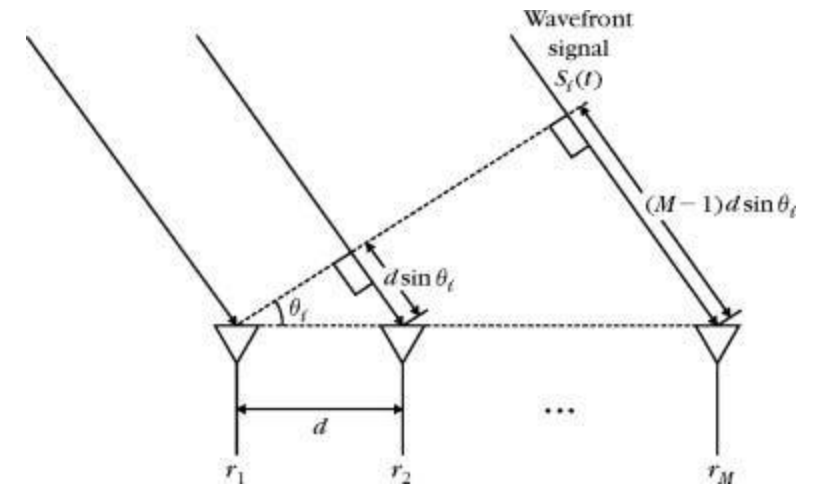
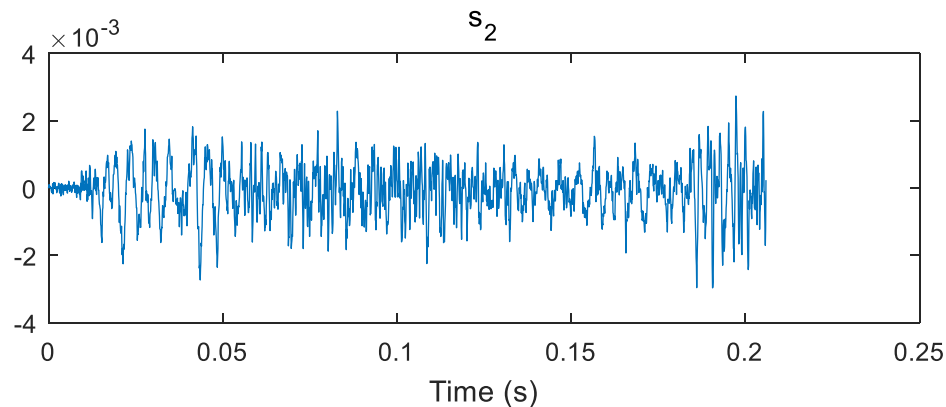
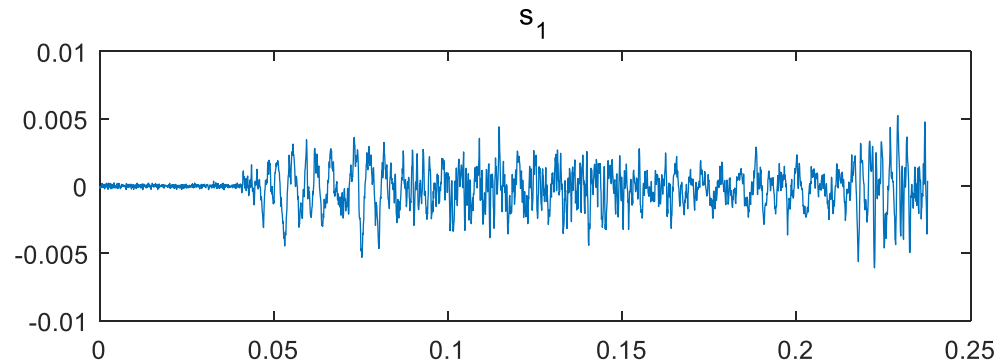
- **Goal:** extract K sources from the noisy mixture of multi-channel audio



- How can we measure the multi-channel audio signal?

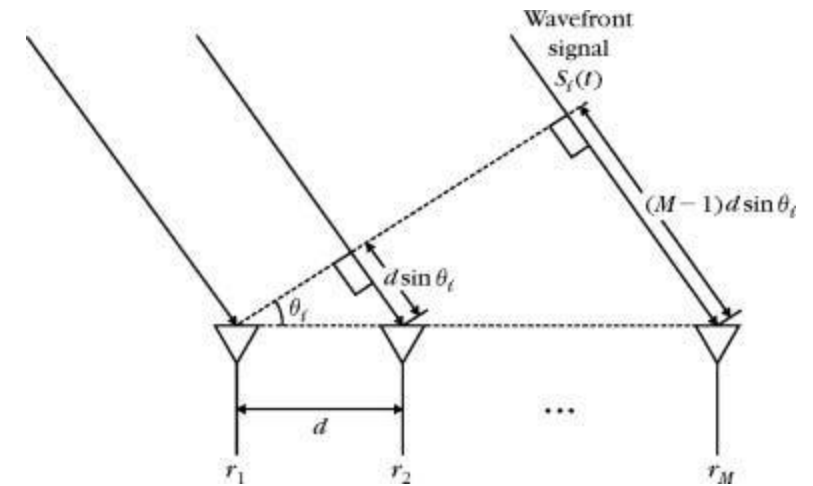
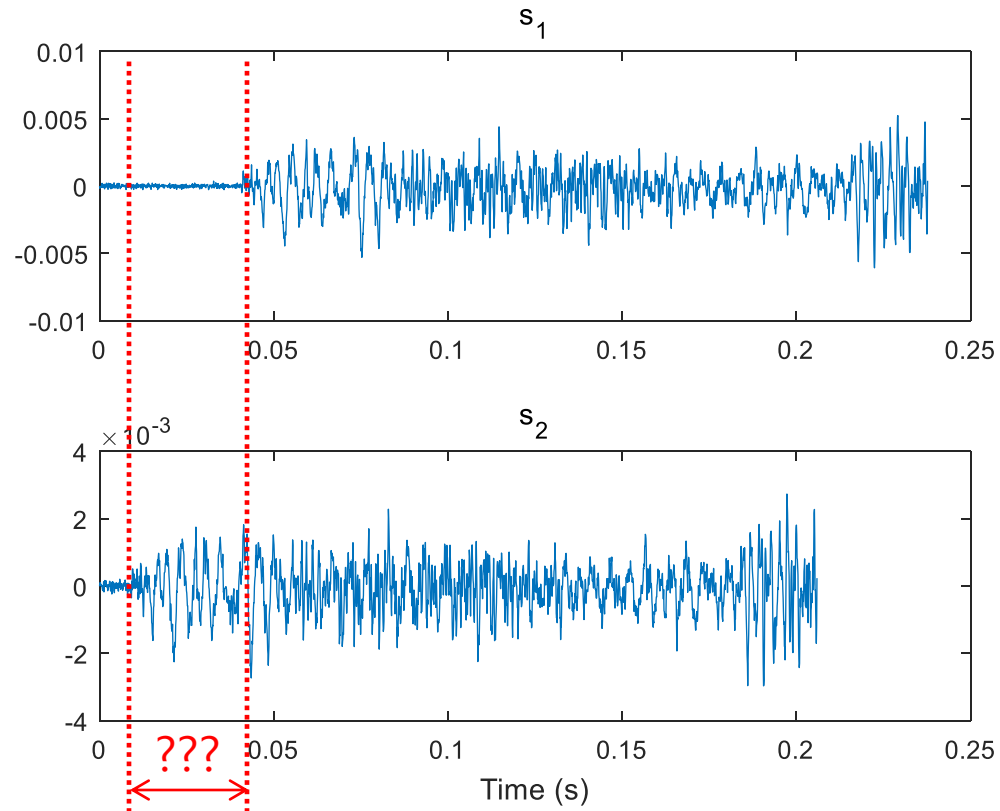
Multi-channel Source Separation

- Microphone Array
 - Consider the Uniform Linear Array
 - What's different about the audio collected at each microphone?



Multi-channel Source Separation

- Microphone Array
 - Consider the Uniform Linear Array
 - What's different about the audio collected at each microphone?



Multi-channel Source Separation

- We have to know the direction-of-arrival (DoA) of the sound

$$y(\boldsymbol{\theta}, t) = \underset{\substack{\uparrow \\ \text{Convolution}}}{w(\boldsymbol{\theta}, -t)^H} * x(t) \quad \xleftrightarrow{\text{FFT \& iFFT}} \quad y(\boldsymbol{\theta}, f) = \underset{\substack{\text{Beamforming} \\ \text{output}}}{w(\boldsymbol{\theta}, f)^H} \underset{\substack{\text{Weight} \\ \text{(Spatial filter)}}}{x(f)} \underset{\substack{\text{Measured} \\ \text{pressure signals}}}{}$$

- However, It is have to find out the accurate DoA...

Problem Definition

- Problem Definition

- 마이크에서 녹음되는 신호는 어떻게 정의할 수 있을까?

- Problem definition:
$$\mathbf{y}(t) = \begin{bmatrix} y_1(t) \\ \vdots \\ y_M(t) \end{bmatrix} = \mathbf{x}(t) + \mathbf{n}(t)$$

- $\mathbf{y}(t)$: observation signals of M microphones
 - $\mathbf{x}(t)$: original signal of M microphones
 - $\mathbf{n}(t)$: noise of M microphones

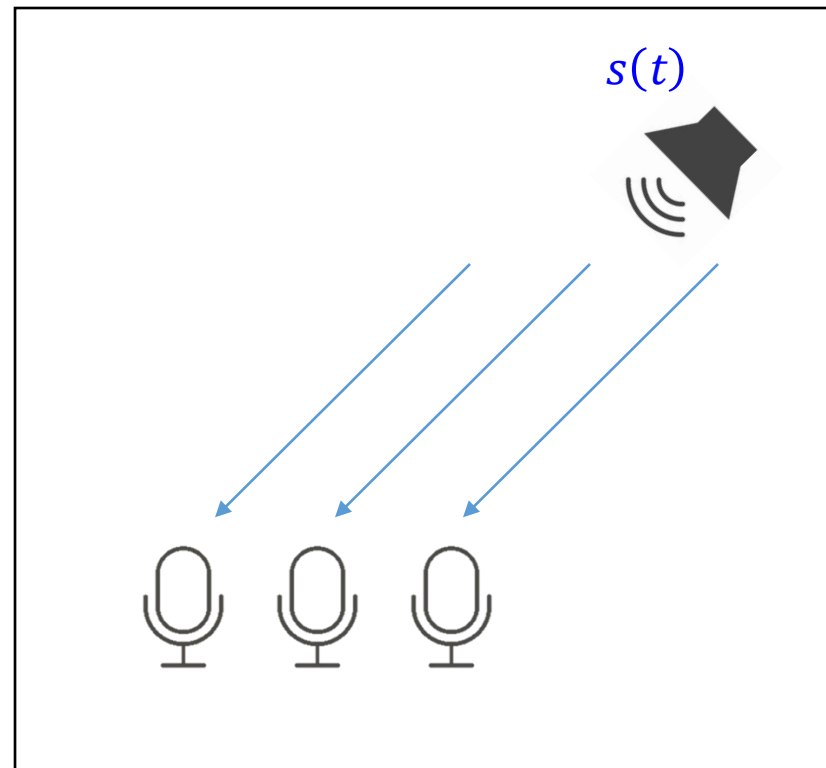
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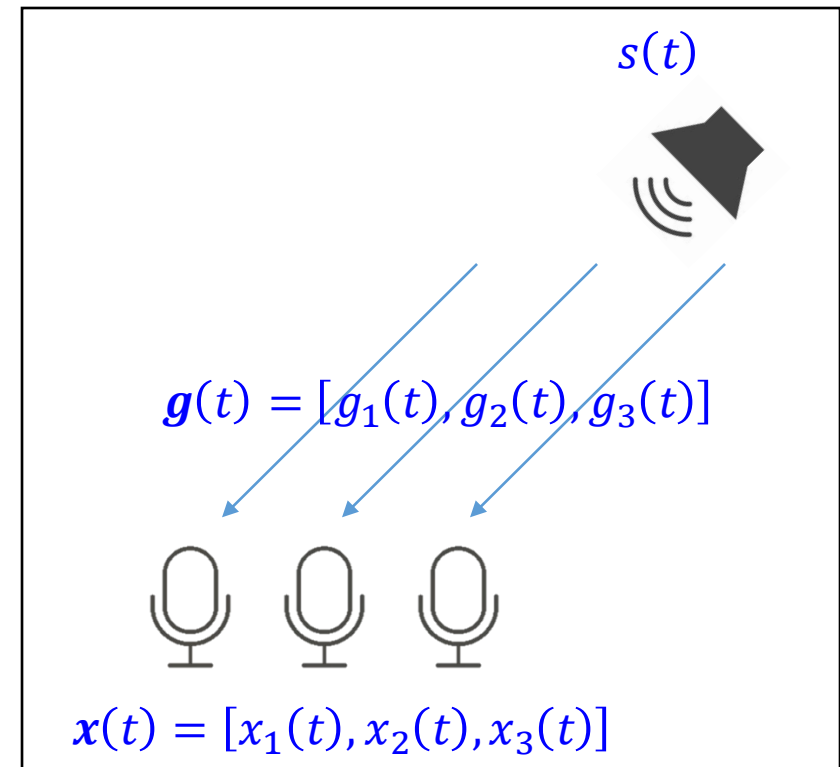
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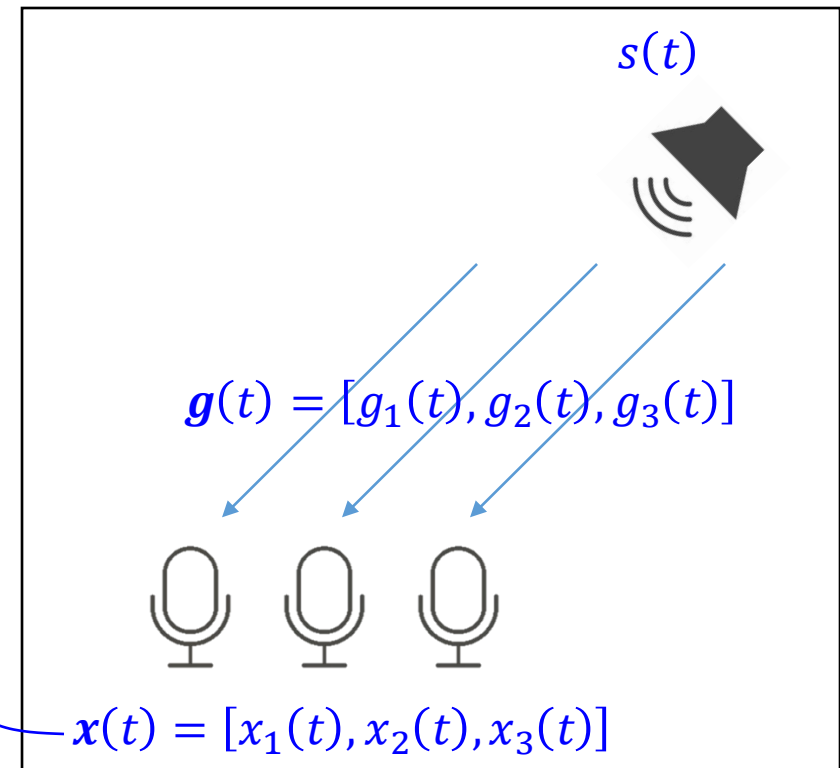
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Problem Definition

- Wiener Filter

- Linear filter: $\hat{x}(t) = \mathbf{w}^H * y(t)$

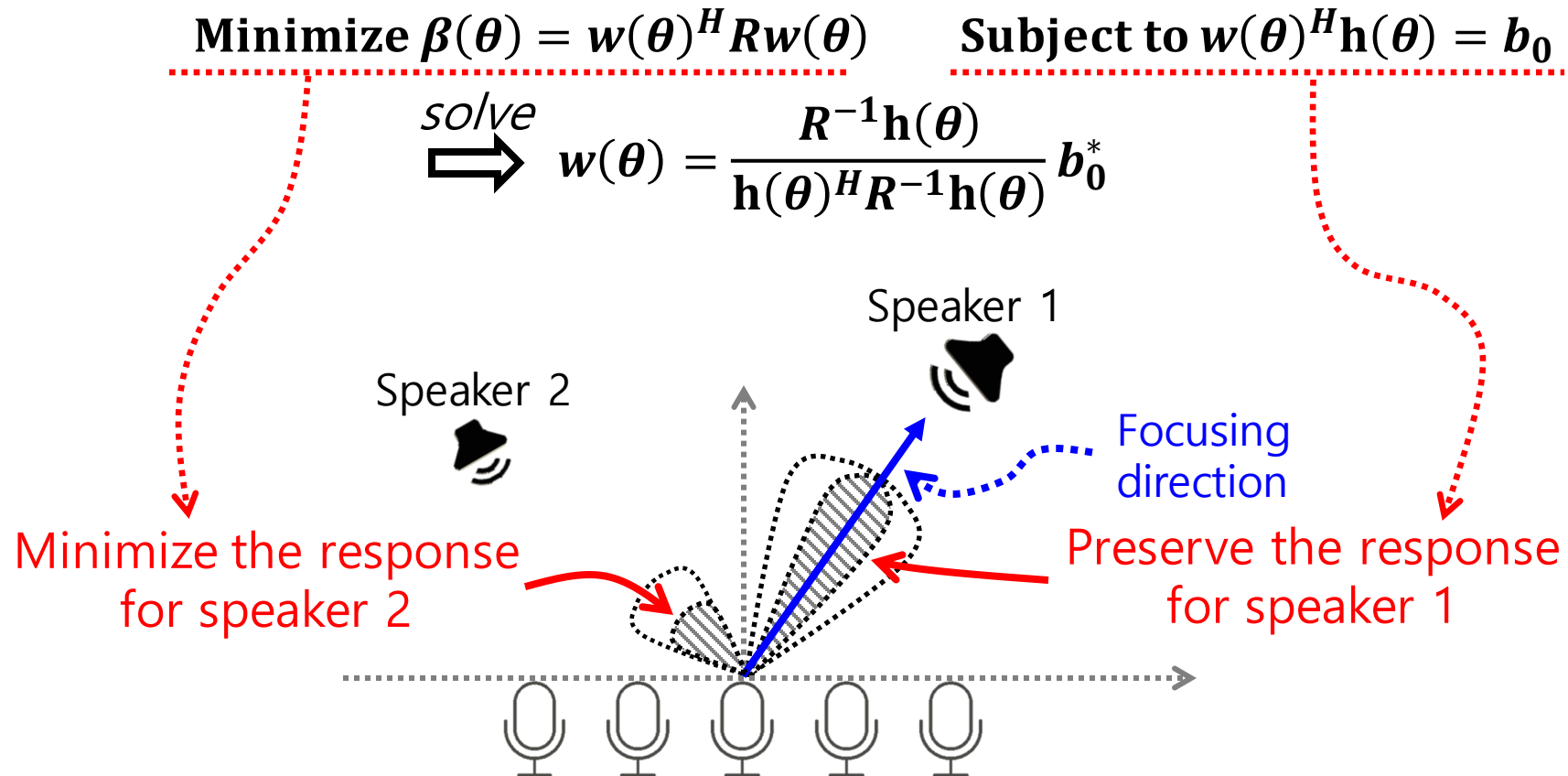
- Given, $y(t) = \begin{bmatrix} y_1(t) \\ \vdots \\ y_M(t) \end{bmatrix} = \mathbf{x}(t) + \mathbf{n}(t) \xrightarrow{\text{FFT}} \mathbf{Y}(f) = \mathbf{X}(f) + \mathbf{N}(f)$

- Goal1: design a spatial filter $\mathbf{w}(t)$ to minimize the response for noise
 - noise를 줄여서 $\mathbf{x}(t)$ 를 복원하도록 $\mathbf{w}(t)$ 를 설계
 - 평균 제곱 오차 (Mean Square Error, MSE)를 최소화하는 $\mathbf{w}(t)$ 를 계산

$$J(\mathbf{w}) = E[|y(t) - \mathbf{w}^H * y(t)|^2] \xrightarrow{\text{FFT}} J(\mathbf{w}) = E[|\mathbf{Y}(f) - \mathbf{W}^H * \mathbf{Y}(f)|^2]$$

Multi-channel Source Separation

- MVDR beamformer
 - Design a spatial filter w that minimizes the total beamforming power but maintain the response in the aiming angle



Problem Definition

- Wiener Filter-based MVDR beamformer

- Linear filter: $\hat{x}(t) = \mathbf{w}^H * y(t)$

- Given, $y(t) = \begin{bmatrix} y_1(t) \\ \vdots \\ y_M(t) \end{bmatrix} = \mathbf{x}(t) + \mathbf{n}(t) \xrightarrow{\text{FFT}} \mathbf{Y}(f) = \mathbf{X}(f) + \mathbf{N}(f)$

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- Goal2: Preserve the response for $\mathbf{x}(t)$

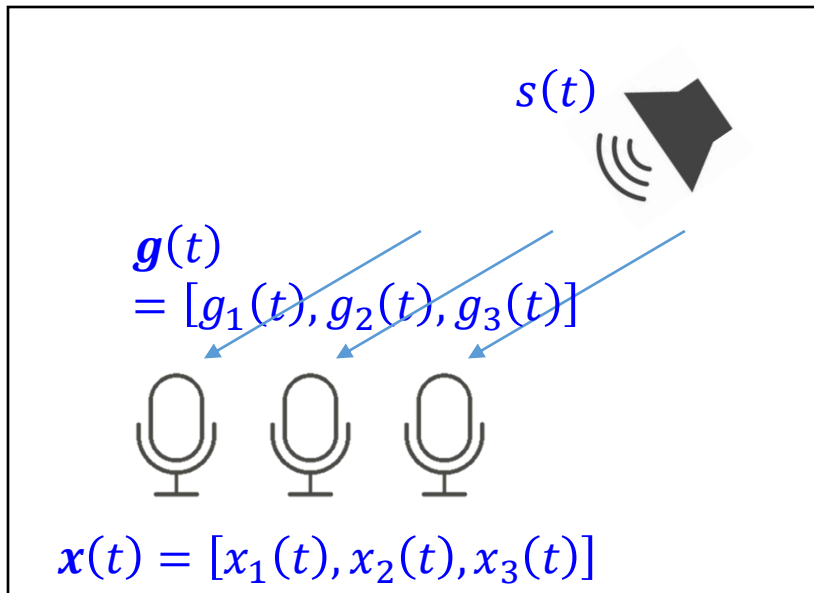
$$E[|\mathbf{X}(f) - \mathbf{W}^H * \mathbf{Y}(f)|^2] = G(f)$$

Problem Definition

- Requirement of Wiener Filter-based MVDR beamformer
 - Goal1 + Goal2: design a spatial filter $w(t)$ to minimize the response for noise while preserving the response for $x(t)$

$$J(\mathbf{w}) = \min_{\mathbf{w}} E[|Y(f) - \mathbf{w}^H * \mathbf{Y}(f)|^2]$$

$$\text{subset to } E[|X(f) - \mathbf{w}^H * \mathbf{Y}(f)|^2] = G(f)$$



$$\begin{aligned} \mathbf{w} &= \frac{E[|\mathbf{N}(f)|^2]^{-1} \cdot E[|\mathbf{X}(f)|^2]}{\text{Tr}(E[|\mathbf{N}(f)|^2]^{-1} \cdot E[|\mathbf{X}(f)|^2])} u_{n_0} \\ &= \frac{\Phi^N(f)^{-1} \cdot \Phi^S(f)}{\text{Tr}(\Phi^N(f)^{-1} \cdot \Phi^S(f))} u_{n_0} \end{aligned}$$

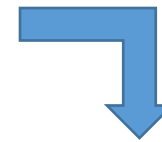
$$\mathbf{Y}(f) = \begin{bmatrix} Y_1(f) \\ \vdots \\ Y_M(f) \end{bmatrix} = \mathbf{X}(f) + \mathbf{N}(f)$$

Problem Definition

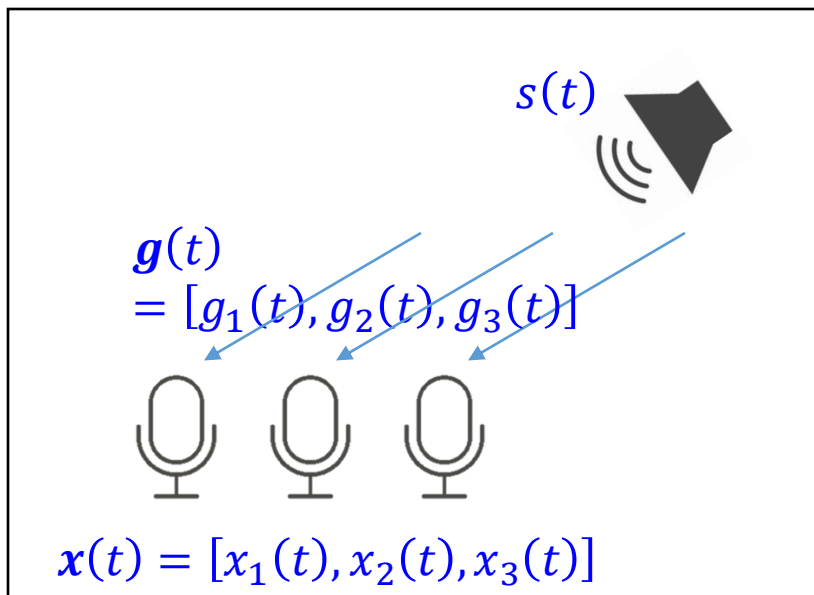
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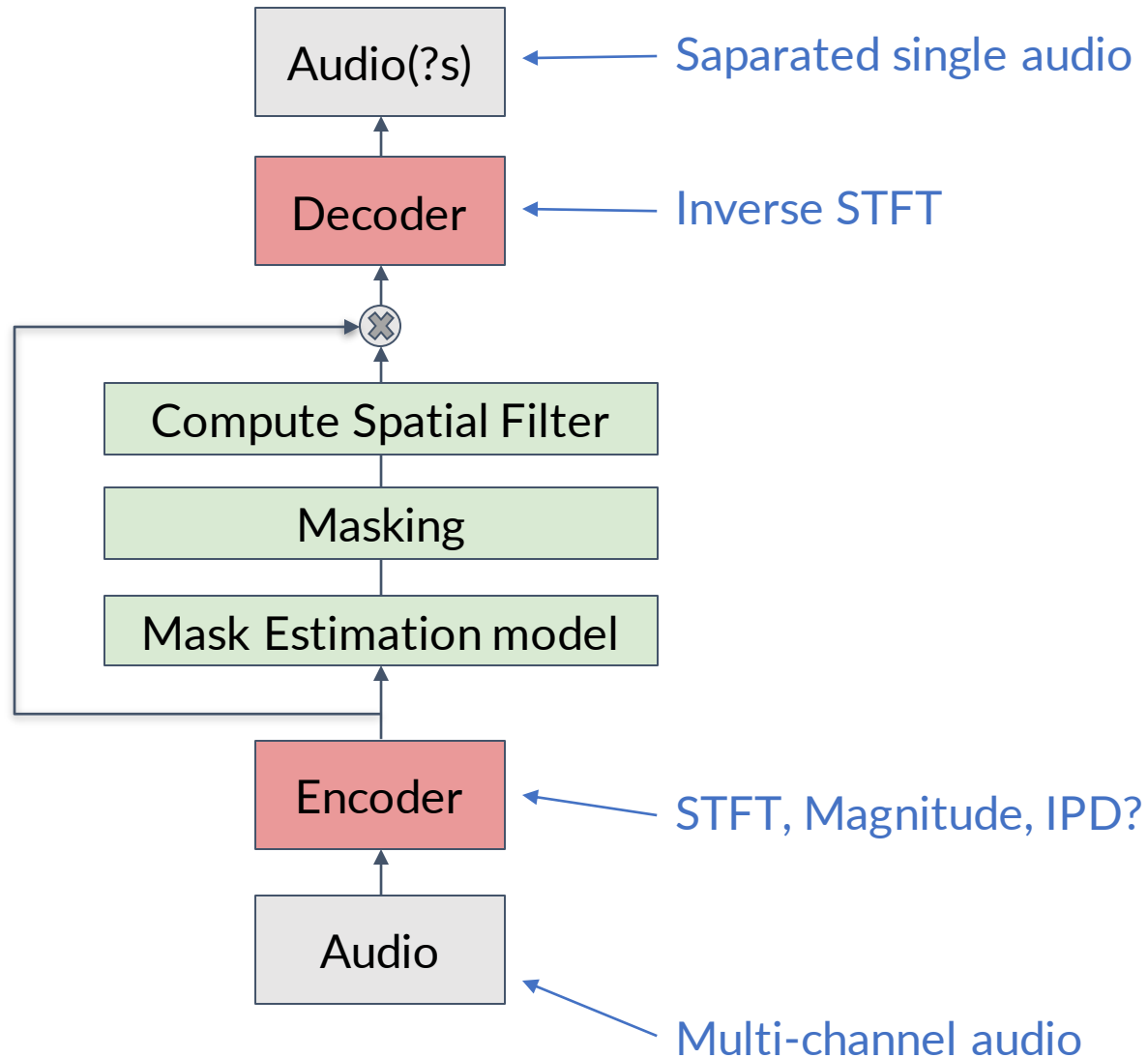
Original signal과 Noise signal을 알아야 한다.



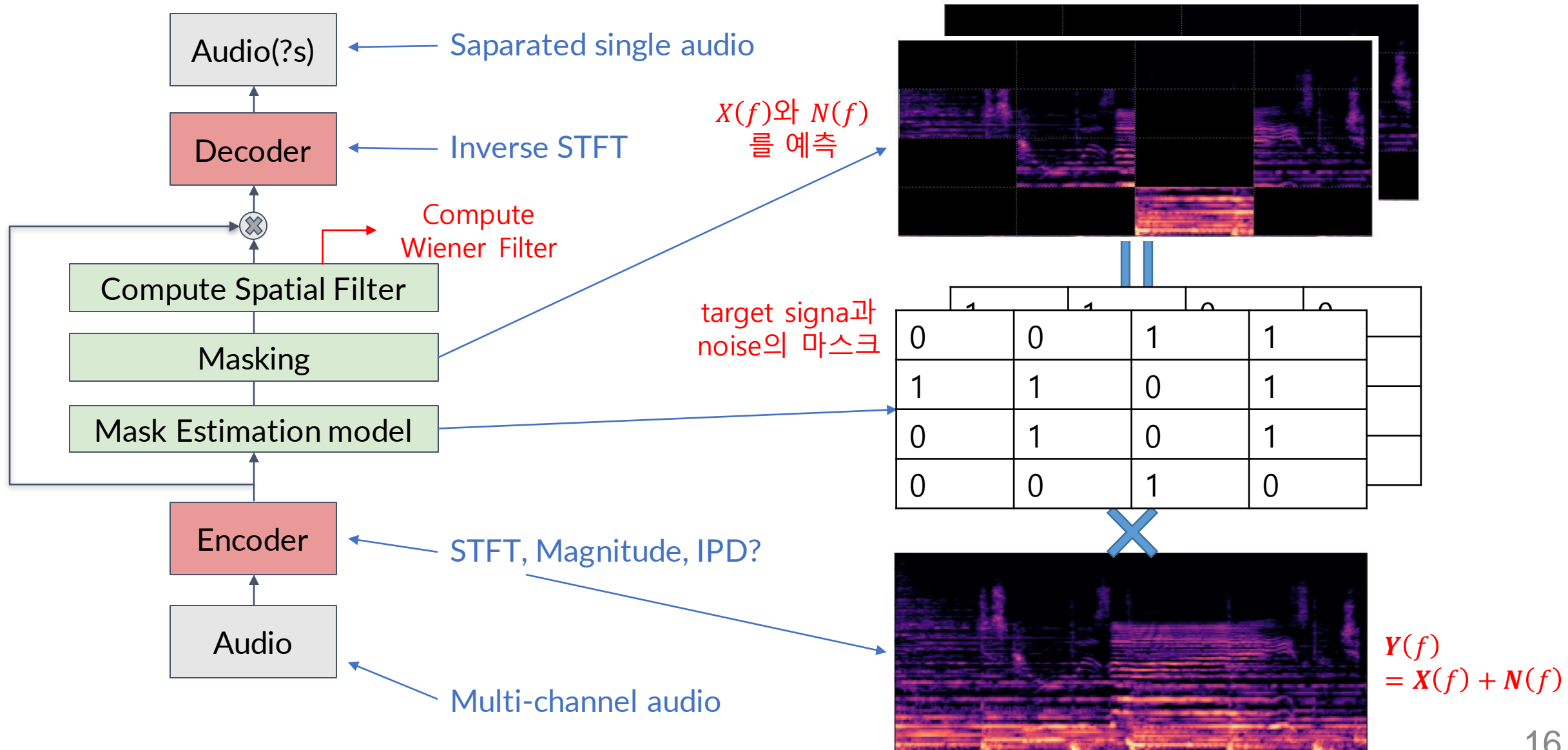
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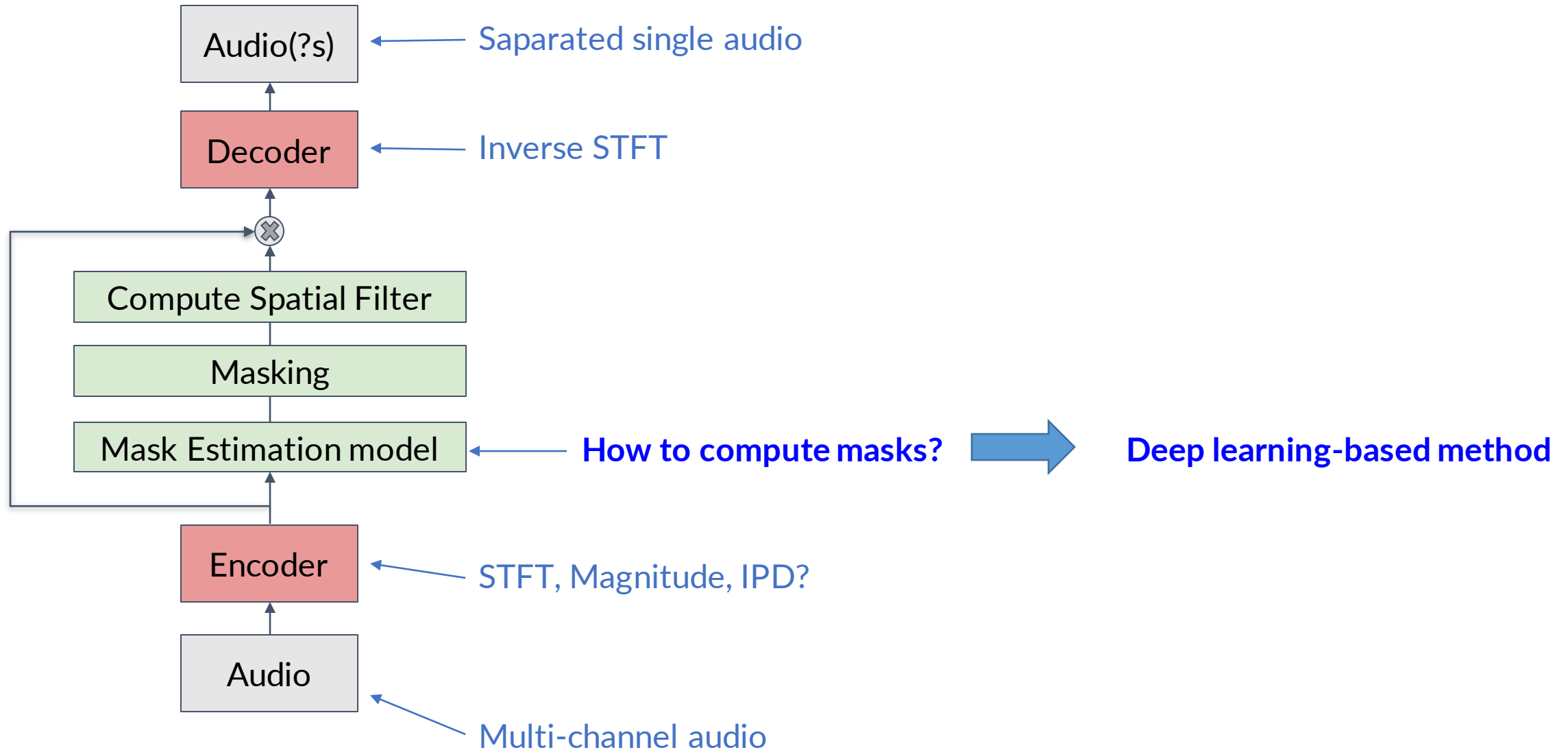
Multi-Channel Encoder-Separation-Decoder (ESD)



Multi-Channel Encoder-Separation-Decoder (ESD)

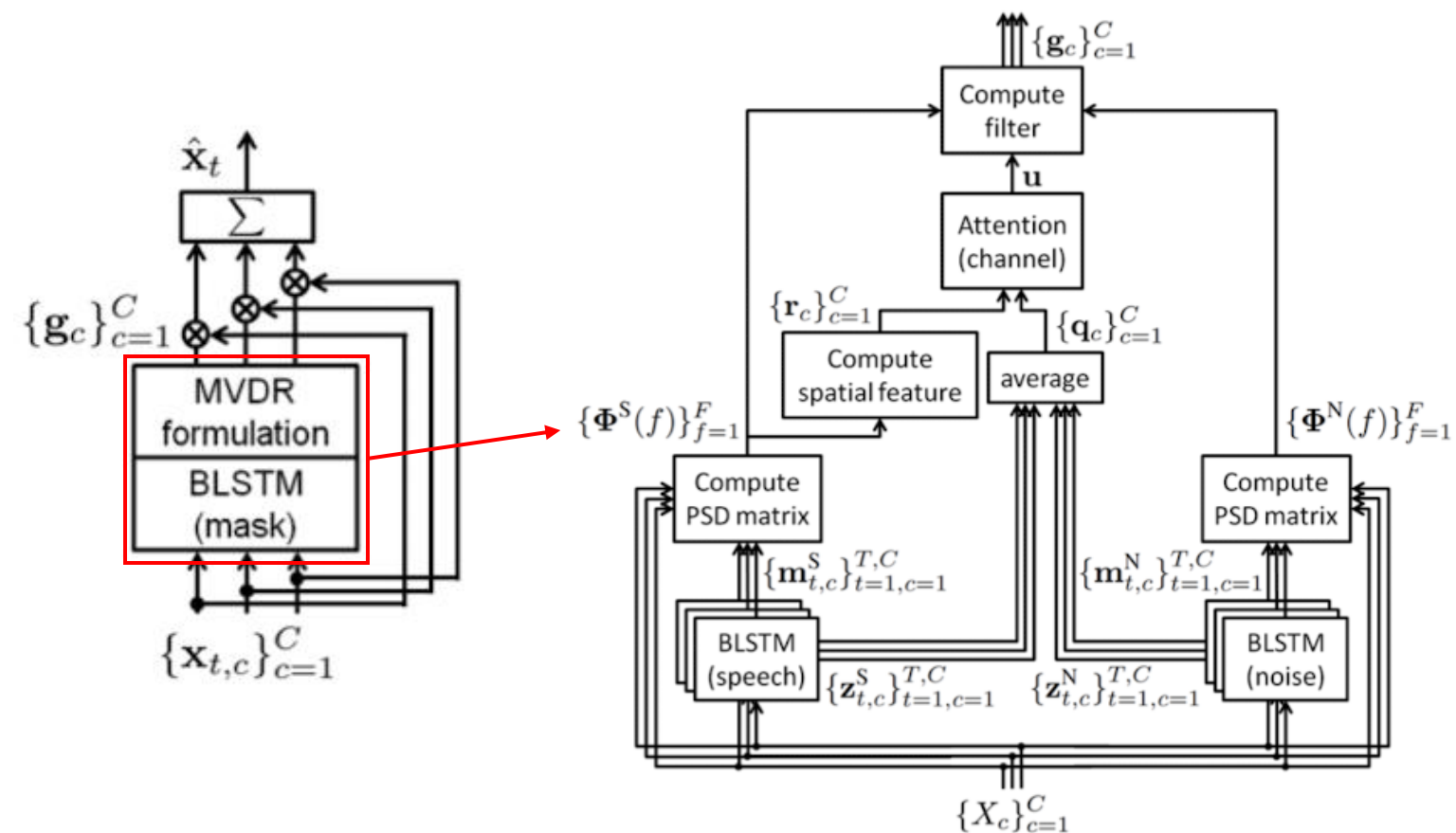


Multi-Channel Encoder-Separation-Decoder (ESD)



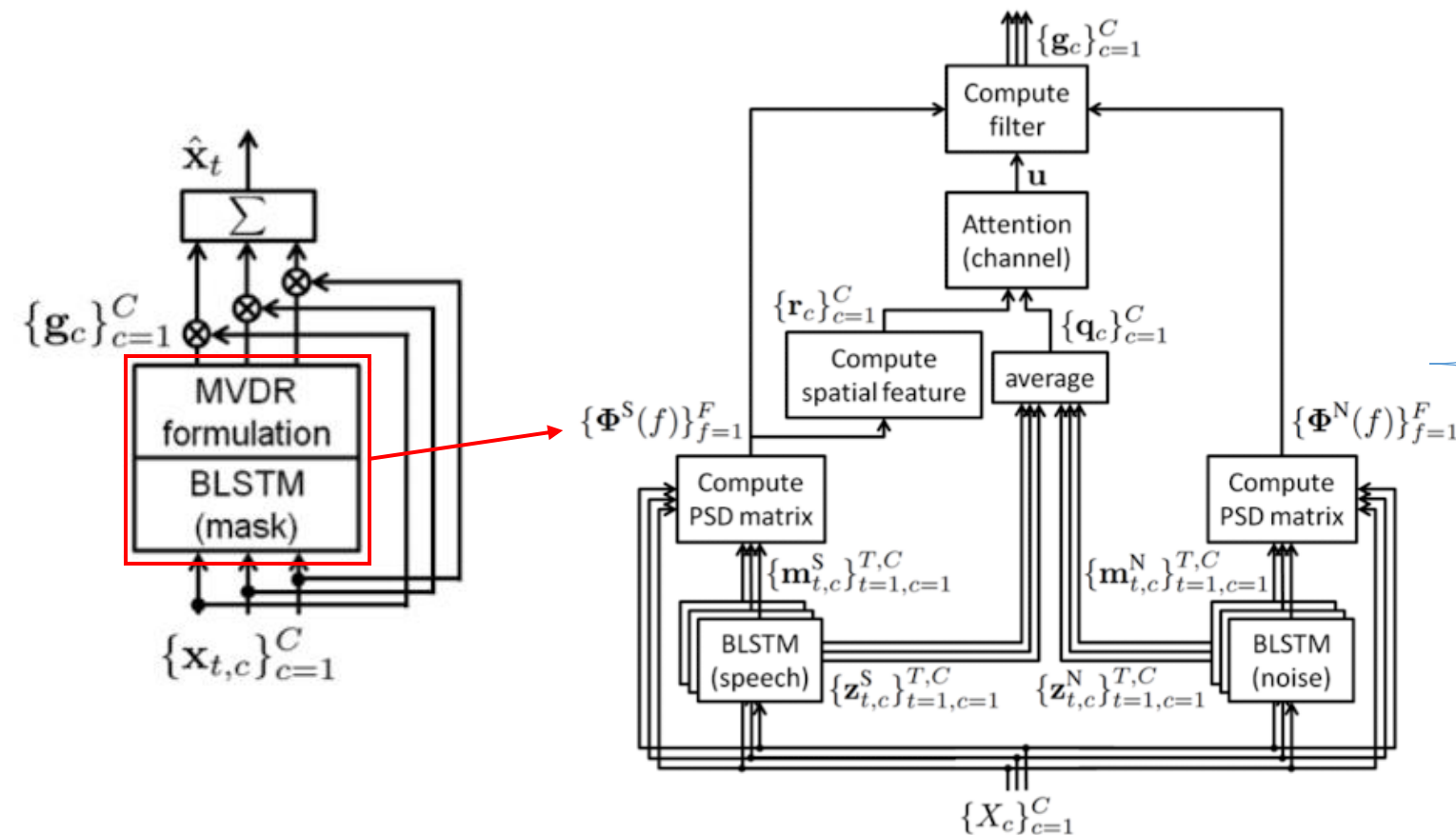
Multi-Channel ASR

- Multichannel End-to-end Speech Recognition, ICML17
 - The neural Beamformer is based on the MVDR formalizations



Multi-Channel ASR

- Multichannel End-to-end Speech Recognition, ICML17
 - The neural Beamformer is based on the MVDR formalizations



- The filter:

$$\mathbf{g}(f) = \frac{\Phi^N(f)^{-1} \Phi^S(f)}{\text{Tr}(\Phi^N(f)^{-1} \Phi^S(f))} \mathbf{u},$$

- The cross-channel power spectral density (PSD) matrices :

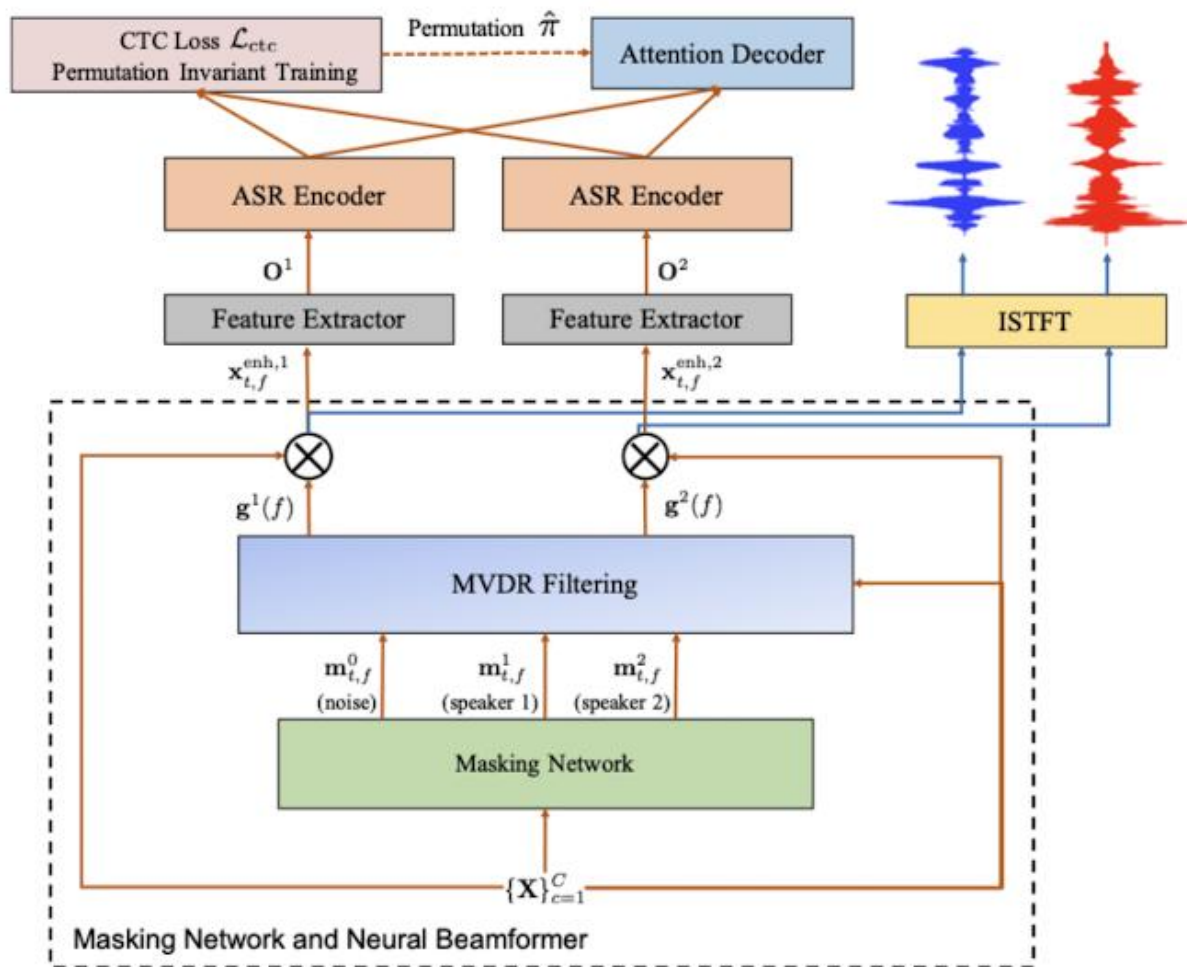
$$\Phi^S(f) = \frac{1}{\sum_{t=1}^T m_{t,f}^S} \sum_{t=1}^T m_{t,f}^S \mathbf{x}_{t,f} \mathbf{x}_{t,f}^\dagger,$$

$$\Phi^N(f) = \frac{1}{\sum_{t=1}^T m_{t,f}^N} \sum_{t=1}^T m_{t,f}^N \mathbf{x}_{t,f} \mathbf{x}_{t,f}^\dagger,$$

Target and noise masks can be obtained using a DL model

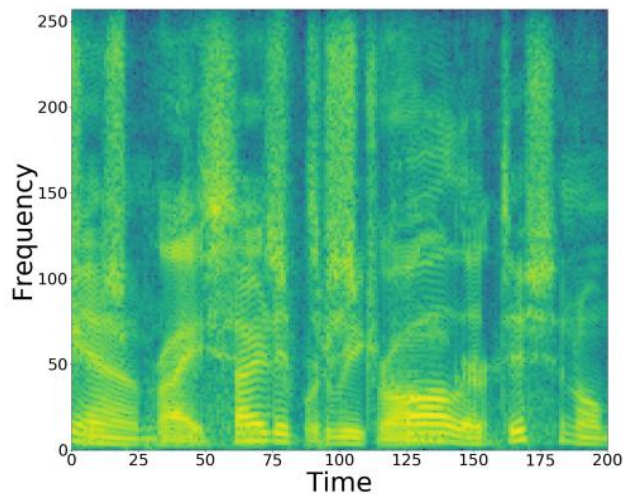
Multi-Channel ASR

- MINO-Speech: End-To-End Multi-Channel Multi-Speaker Speech Recognition, ASRU19

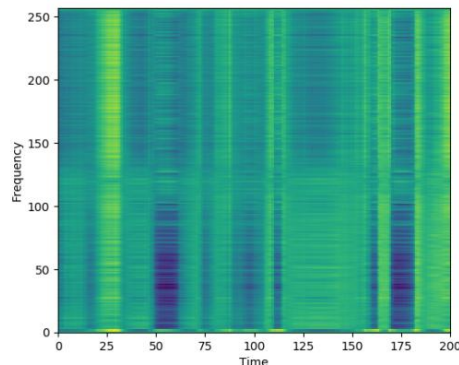


Multi-Channel ASR

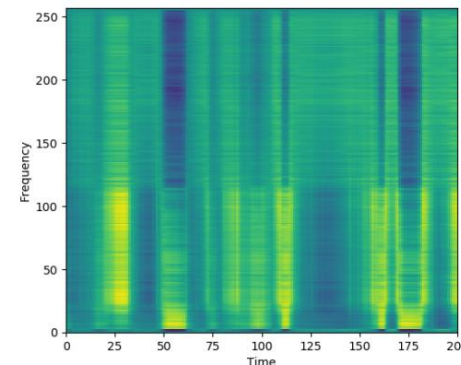
- MINO-Speech: End-To-End Multi-Channel **Multi-Speaker** Speech Recognition, ASRU19



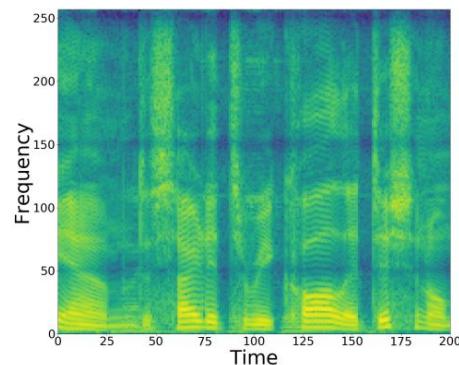
(e) Overlapped Speech



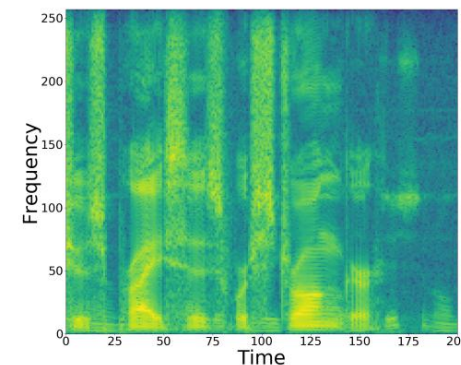
(a) Mask for Speaker 1



(b) Mask for Speaker 2



(c) Separated Speech for Speaker 1



(d) Separated Speech for Speaker 2

Multi-Channel ASR

- MINO-Speech: End-To-End Multi-Channel Multi-Speaker Speech Recognition, ASRU19

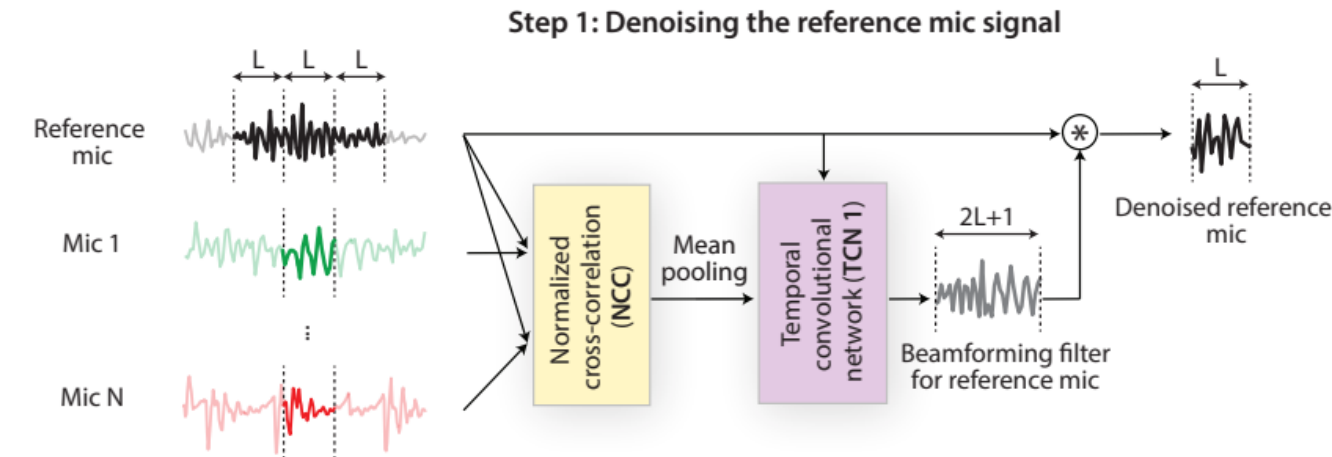
Table 1. Performance in terms of average CER and WER [%] on the spatialized anechoic wsj1-2mix corpus.

Model	dev CER	eval CER
2-spkr ASR (1st channel)	22.65	19.07
BeamformIt Enhancement (2-spkr ASR)	15.23	12.45
BeamformIt Separation (1-spkr ASR)	77.30	77.10
MIMO-Speech	7.29	4.51
+ Curriculum Learning (SNRs)	6.34	3.75

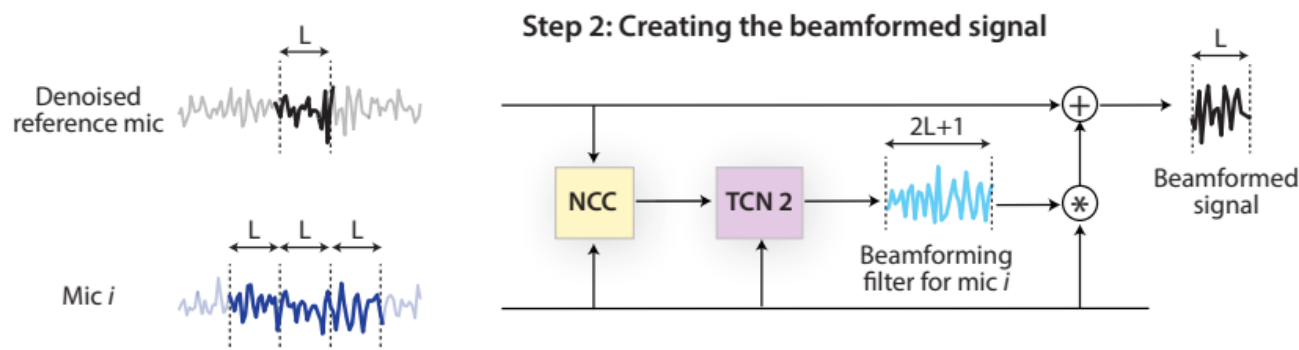
Model	dev WER	eval WER
2-spkr ASR (1st channel)	34.98	29.43
BeamformIt Enhancement (2-spkr ASR)	26.61	21.75
BeamformIt Separation (1-spkr ASR)	98.60	98.00
MIMO-Speech	13.54	8.62
+ Curriculum Learning (SNRs)	12.59	7.55

Multi-Channel ASR

- FaSNet: Low-Latency Adaptive Beamforming for Multi-Microphone Audio Processing, ASRU 2019
 - Time domain neural beamforming $\hat{x}(t) = \mathbf{w}^H * y(t)$

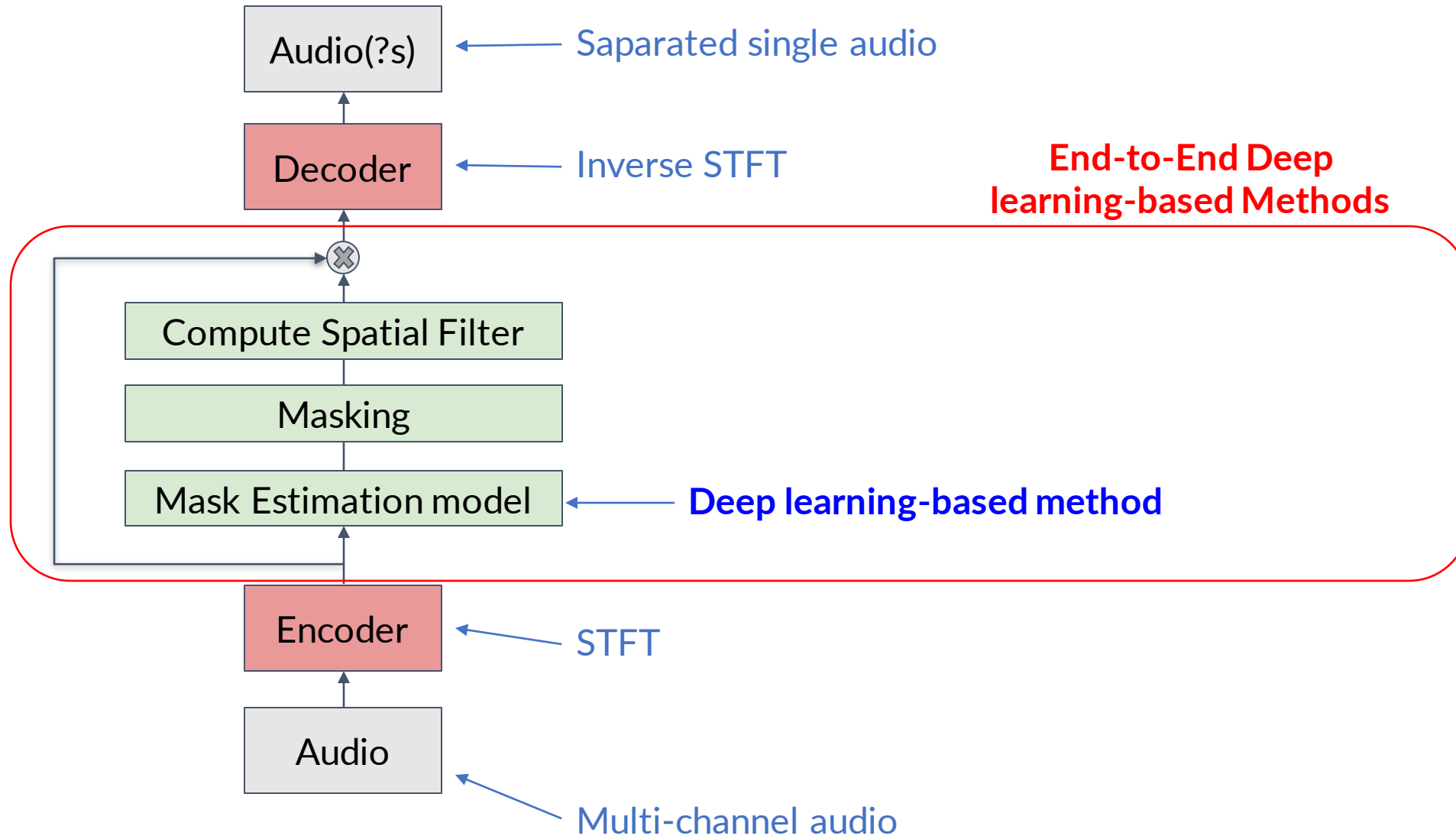


Target signal의 clue를 찾은 후,



이를 활용해 spatial clue를 고려하도록 설계

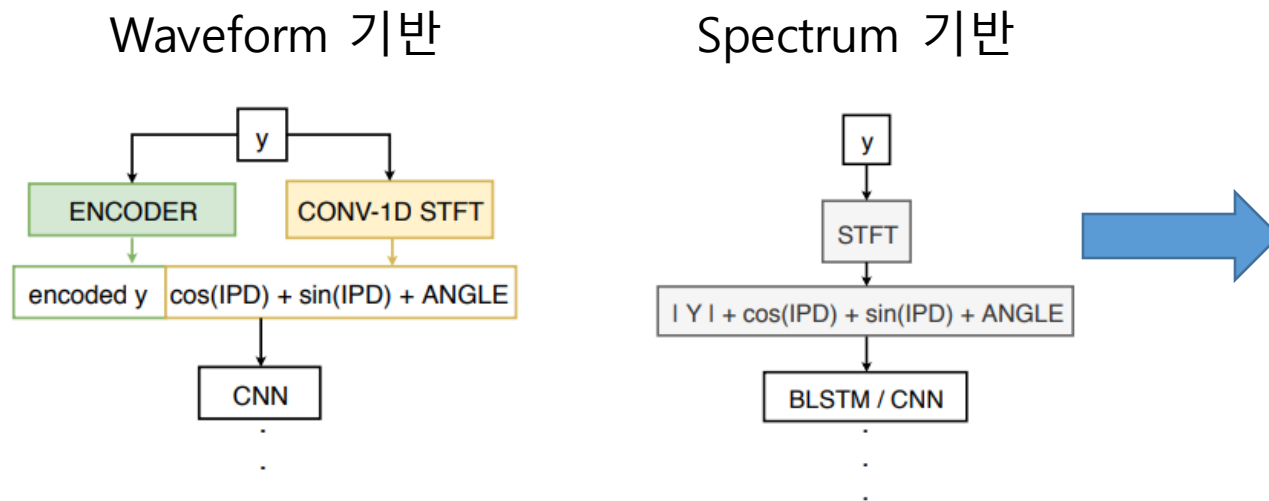
Multi-Channel Encoder-Separation-Decoder (ESD)



Multi-Channel ASR

- A comprehensive study of speech separation: spectrogram vs waveform separation, Interspeech 2019

IPD (Inter-Phase Difference):
$$\text{IPD}_{i,t,f} = \angle\left(\frac{Y_{i_1,t,f}}{Y_{i_2,t,f}}\right), i = 1 : 6$$



- Waveform / Spectrum 중 어떤 것이 더 효과적이지?
- STFT / IPD / Angle 중 어떤 input feature가 효과적이지?

Figure 1: Multi-channel speech separation; (left) waveform separation, (right) spectrogram separation.

Multi-Channel ASR

- A comprehensive study of speech separation: spectrogram vs waveform separation, Interspeech 2019

Table 1: *Experimental setup for time/frequency-domain models.*

Model	Input	# of Parameters	Setting	Normalization
F-CNN-1	$ Y_0 $	8.78M	N=257	gLN
F-CNN-2	$ Y_0 + \cos(\text{IPD}) + \sin(\text{IPD})$	9.58M	N=257 × 13	gLN
F-CNN-3	$ Y_0 + \cos(\text{IPD}) + \sin(\text{IPD}) + \text{Angle}$	9.71M	N=257 × 15	gLN
F-BLSTM-1	$ Y_0 $	67.05M	4 × BLSTM-896	-
F-BLSTM-2	$ Y_0 + \cos(\text{IPD}) + \sin(\text{IPD})$	89.15M	4 × BLSTM-896	-
F-BLSTM-3	$ Y_0 + \cos(\text{IPD}) + \sin(\text{IPD}) + \text{Angle}$	92.84M	4 × BLSTM-896	-
T-CNN-1	y_0	8.76M	L/N=40/256	gLN
T-CNN-2	$y_0 + \cos(\text{IPD}) + \sin(\text{IPD})$	8.83M	L/N=40/256	BN

Table 4: *Comparing spectrogram and waveform separation for both separation and ASR tasks.*

# Channels	Domain	Model	Si-SNR					SDR					PESQ	WER Reduc. (%)
			0-15°	15-45°	45-90°	90-180°	AVG	0-15°	15-45°	45-90°	90-180°	AVG		
1-ch	time	T-BLSTM	-	-	-	-	-	-	-	-	-	-	-	-
		T-CNN-1	9.02	9.33	9.59	9.71	9.47	9.57	9.83	10.09	10.2	9.97	1.95	45.53
	freq	F-BLSTM-1	7.54	7.80	7.72	7.81	7.74	8.14	8.39	8.29	8.38	8.32	1.77	32.21
		F-CNN-1	7.08	7.48	7.45	7.48	7.42	7.70	8.06	8.02	8.06	8	1.77	35.17
m-ch	time	T-BLSTM	-	-	-	-	-	-	-	-	-	-	-	-
		T-CNN-2	7.70	11.63	12.33	12.62	11.55	8.31	12.07	12.74	13.03	11.99	2.10	59.11
	freq	F-BLSTM-2	5.41	9.37	10.13	10.65	9.38	6.13	9.89	10.62	11.13	9.91	1.92	45.32
		F-CNN-2	6.88	10.27	11.02	11.54	10.36	7.5	10.75	11.47	11.99	10.84	2.00	51.73
Oracle IBM			11.56	11.51	11.53	11.53	11.53	11.93	11.86	11.88	11.88	11.88	2.01	50.37
Oracle IAM			11.05	11.03	11.05	11.03	11.04	11.33	11.3	11.31	11.29	11.30	2.23	71.45
Oracle IRM			11.01	10.96	10.98	10.97	10.98	11.45	11.39	11.39	11.39	11.40	2.22	70.28
Oracle IPSM			13.68	13.6	13.64	13.63	13.63	14.04	13.94	13.98	13.97	13.98	2.28	71.10
Reference													2.35	73.01

Multi-Channel ASR

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1-ch	time	T-BLSTM	-	-	-	-	-	-	-	-	-	-	-	-
		T-CNN-1	9.02	9.33	9.59	9.71	9.47	9.57	9.83	10.09	10.2	9.97	1.95	45.53
	freq	F-BLSTM-1	7.54	7.80	7.72	7.81	7.74	8.14	8.39	8.29	8.38	8.32	1.77	32.21
		F-CNN-1	7.08	7.48	7.45	7.48	7.42	7.70	8.06	8.02	8.06	8	1.77	35.17
m-ch	time	T-BLSTM	-	-	-	-	-	-	-	-	-	-	-	-
		T-CNN-2	7.70	11.63	12.33	12.62	11.55	8.31	12.07	12.74	13.03	11.99	2.10	59.11
	freq	F-BLSTM-2	5.41	9.37	10.13	10.65	9.38	6.13	9.89	10.62	11.13	9.91	1.92	45.32
		F-CNN-2	6.88	10.27	11.02	11.54	10.36	7.5	10.75	11.47	11.99	10.84	2.00	51.73
	Oracle IBM		11.56	11.51	11.53	11.53	11.53	11.93	11.86	11.88	11.88	11.88	2.01	50.37
	Oracle IAM		11.05	11.03	11.05	11.03	11.04	11.33	11.3	11.31	11.29	11.30	2.23	71.45
	Oracle IRM		11.01	10.96	10.98	10.97	10.98	11.45	11.39	11.39	11.39	11.40	2.22	70.28
	Oracle IPSM		13.68	13.6	13.64	13.63	13.63	14.04	13.94	13.98	13.97	13.98	2.28	71.10
	Reference												2.35	73.01

- 1-ch보다 m-ch이 성능이 좋다
- CNN이 BLSTM 보다 성능이 좋다
- Time domain + IPD가 성능이 좋다 → Inter-channel relation을 잘 고려해야 한다.

Multi-Channel ASR

- DASFORMER: DEEP ALTERNATING SPECTROGRAM TRANSFORMER FOR MULTI/SINGLE-CHANNEL SPEECH SEPARATION, ICASSP23

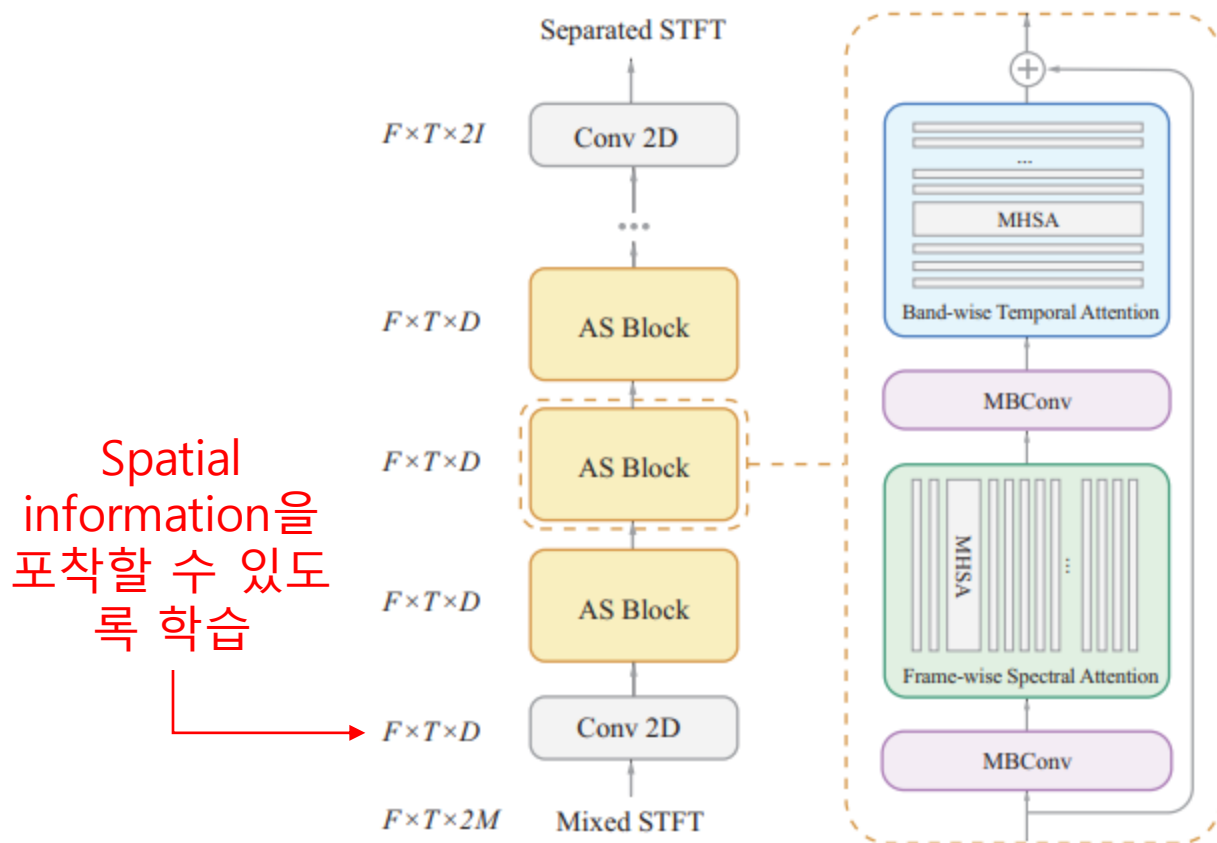


Fig. 1: The architecture of the proposed DasFormer.

Model	Params. (M)	PESQ	SDRi (dB)
RIR settings as [16, 5]			
Mixture	—	1.80	0.0
FaSNet-TAC [21]	2.8	2.90	11.7
NBC [4]	2.0	2.95	13.3
Beam-TasNet [16]	—	—	16.8
BeamGuided-TasNet [5]	5.4	—	21.5
DasFormer (ours)	2.2	4.33	25.9
RIR settings as [4]			
Mixture	—	1.80	0.0
NBC [4]	2.0	3.53	15.3
DasFormer (ours)	2.2	4.11	20.5

Table 1: Experiment results on spatialized WSJ0-2Mix.

Multi-Channel ASR

- SpatialNet: Extensively Learning Spatial Information for Multichannel Joint Speech Separation, Denoising and Dereverberation, ASRU19

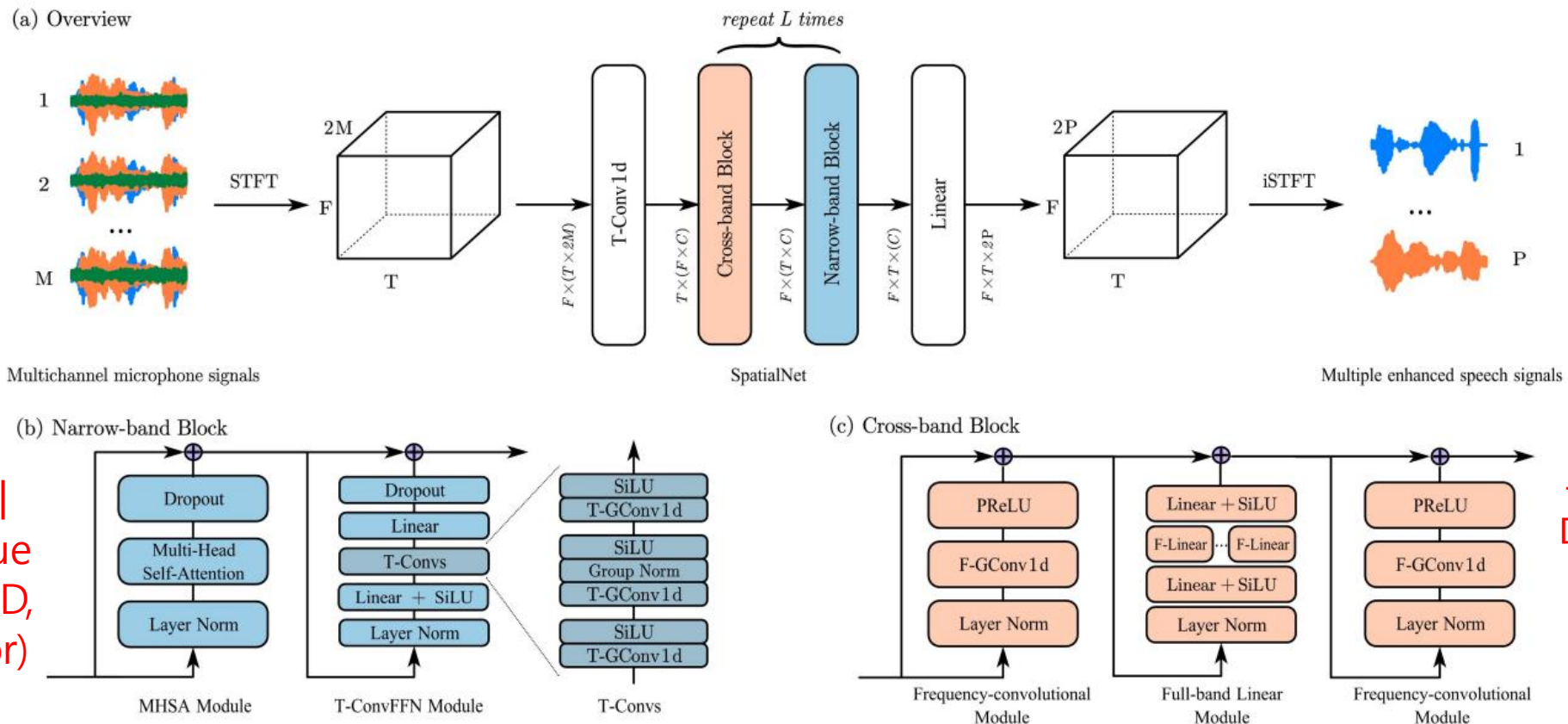


Fig. 1. The proposed SpatialNet. (a) The system overview. The input dimensions of neural blocks are presented before each of them in the form “batch dimension \times (dimension of one sample in batch)”. (b) The narrow-band block. (c) The cross-band block.

Multi-Channel ASR

- SpatialNet: Extensively Learning Spatial Information for Multichannel Joint Speech Separation, Denoising and Dereverberation, ASRU19

TABLE IV
RESULTS ON 2-CHANNEL AND 6-CHANNEL SMS-WSJ DATASET. *
DENOTE THAT THE SCORES ARE QUOTED FROM [17].

Method	SISDR (dB)	SDR (dB)	NB-PESQ	eSTOI	WER (%)
unproc.	-5.45	-0.38	1.50	0.441	78.7
oracle direct-path	∞	∞	4.5	1.0	6.31
Results for 2-channel SMS-WSJ					
FasNet+TAC * [12]	6.9	-	2.27	0.731	34.84
Multi-TasNet * [61]	5.8	-	2.16	0.720	45.72
MISO ₁ -BF-MISO ₃ [17]	12.7	-	3.43	0.907	10.67
Convolutional Prediction [62]	15.8	-	3.71	-	8.60
TFGridNet [26]	20.3	22.0	3.81	0.967	7.41
SpatialNet-small (prop.)	19.4	21.0	3.80	0.957	8.10
SpatialNet-large (prop.)	23.3	24.6	4.03	0.975	7.20
Results for 6-channel SMS-WSJ					
FasNet+TAC * [12]	8.60	-	2.37	0.771	29.8
Multi-TasNet * [61]	10.8	-	2.78	0.844	23.1
MISO ₁ -BF-MISO ₃ [17]	15.6	-	3.76	0.942	8.28
MC-CSM with LBT [63]	13.2	14.8	3.33	0.910	9.62
TFGridNet [26]	22.8	24.9	4.08	0.980	6.76
SpatialNet-small (prop.)	21.3	23.2	3.99	0.974	7.05
SpatialNet-large (prop.)	25.1	27.1	4.17	0.986	6.70