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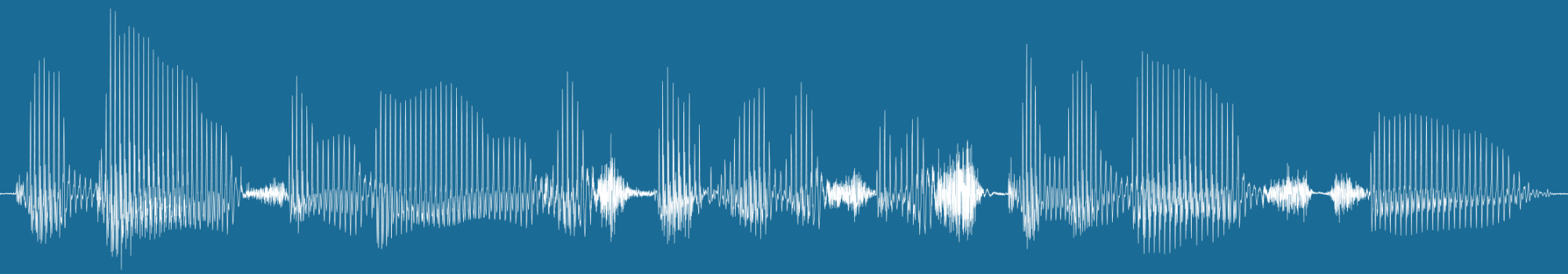

A POLYNOMIAL EIGENVALUE DECOMPOSITION MUSIC APPROACH FOR BROADBAND SOUND SOURCE LOCALIZATION

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Sound source localization is an important task for a multitude of applications, including robot audition and voice-controlled smart devices.

Direction of arrival estimates are essential in providing angular positional information for localization.

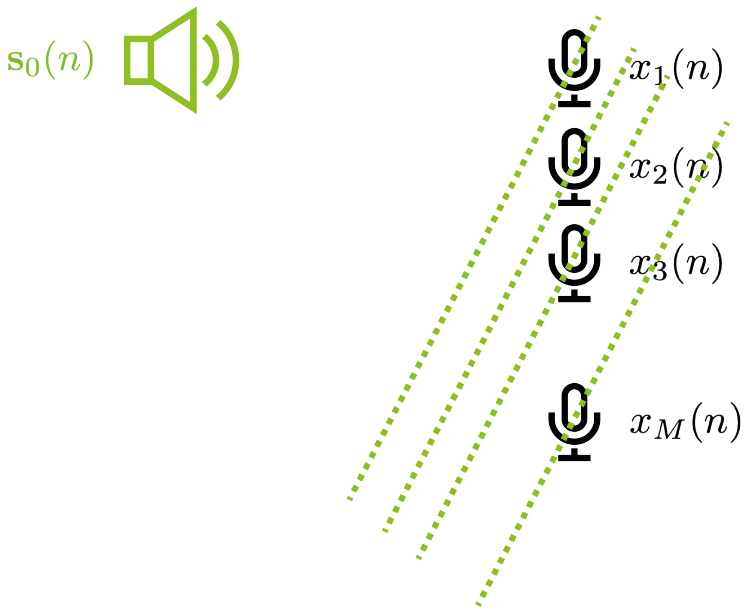
MUSIC ALGORITHM

Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector .

Data model:

$$\mathbf{x}(n) =$$

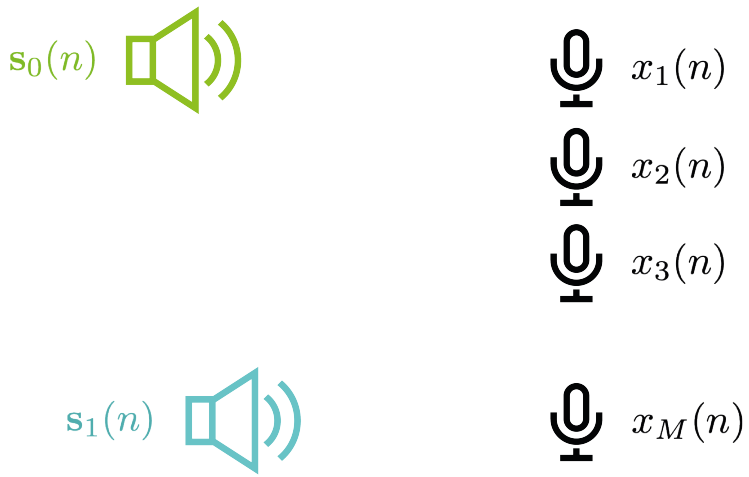
Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector $\mathbf{a}_0$.



Data model:

$$\mathbf{x}(n) = s_0(n) \cdot \mathbf{a}_0$$

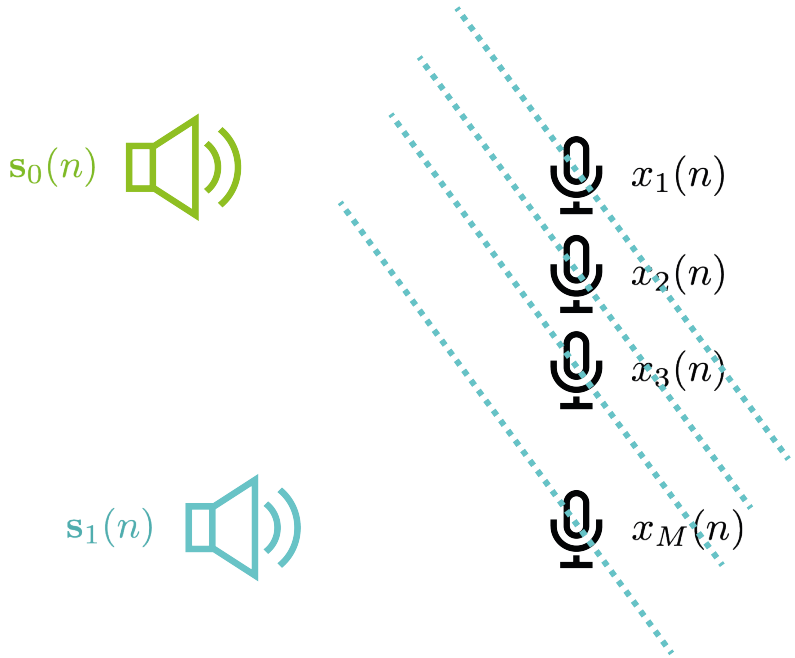
Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector $\mathbf{a}_0$.



Data model:

$$\mathbf{x}(n) = s_0(n) \cdot \mathbf{a}_0$$

Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector $\mathbf{a}_0$, $\mathbf{a}_1$.



Data model:

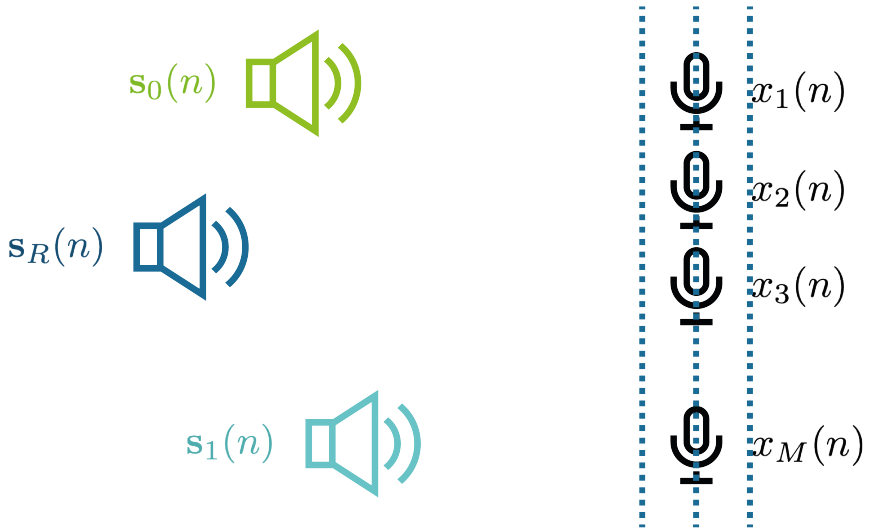
$$\mathbf{x}(n) = s_0(n) \cdot \mathbf{a}_0 + s_1(n) \cdot \mathbf{a}_1$$

Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector $\mathbf{a}_0$, $\mathbf{a}_1$.

Data model:

$$\mathbf{x}(n) = s_0(n) \cdot \mathbf{a}_0 + s_1(n) \cdot \mathbf{a}_1$$

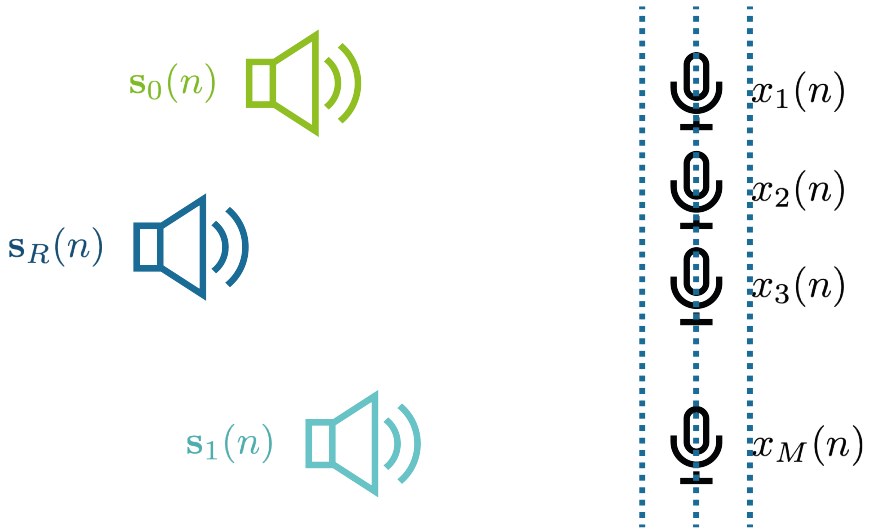
Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector $\mathbf{a}_0, \mathbf{a}_1, \cdots, \mathbf{a}_R$.



Data model:

$$\mathbf{x}(n) = s_0(n) \cdot \mathbf{a}_0 + s_1(n) \cdot \mathbf{a}_1 + \cdots + s_R(n) \cdot \mathbf{a}_R$$

Consider the scenario with a microphone array and a far-field sound source. The sound source signals arrive with delays expressed in a steering vector $\mathbf{a}_0, \mathbf{a}_1, \cdots, \mathbf{a}_R$.



Data model:

$$\mathbf{x}(n) = \sum_{r=0}^{R} s_r(n) \cdot \mathbf{a}_r$$

A signal $s(n)$ arriving at the array can be characterised by the delays of its wavefront (neglecting attenuation):

$$\begin{bmatrix} x_1(n) \\ x_2(n) \\ \vdots \\ x_M(n) \end{bmatrix} = \begin{bmatrix} s(n-\tau_1) \\ s(n-\tau_2) \\ \vdots \\ s(n-\tau_M) \end{bmatrix} = \begin{bmatrix} \delta(n-\tau_1) \\ \delta(n-\tau_2) \\ \vdots \\ \delta(n-\tau_M) \end{bmatrix} * s(n) \xrightarrow{\mathscr{Z}} \mathbf{a}_\theta(z) S(z) \quad (1)$$

If evaluated at a narrowband normalised angular frequency Ω_i, the time delays τ_m in the broadband steering vector $\mathbf{a}_\theta(z)$ collapse to phase shifts in the narrowband steering vector $\mathbf{a}_{\theta,\Omega_i}(z)$,

$$\mathbf{a}_{\theta,\Omega_i}(z) = \mathbf{a}_\theta(z)|_{z=e^{-j\Omega_i}} = \begin{bmatrix} e^{-j\tau_1\Omega_i} \\ e^{-j\tau_2\Omega_i} \\ \vdots \\ e^{-j\tau_M\Omega_i} \end{bmatrix}. \quad (2)$$

If the noisy and reverberant signal, $x_m(n)$, at the m-th microphone for discrete-time sample $n = 0, 1, \ldots, N$, is

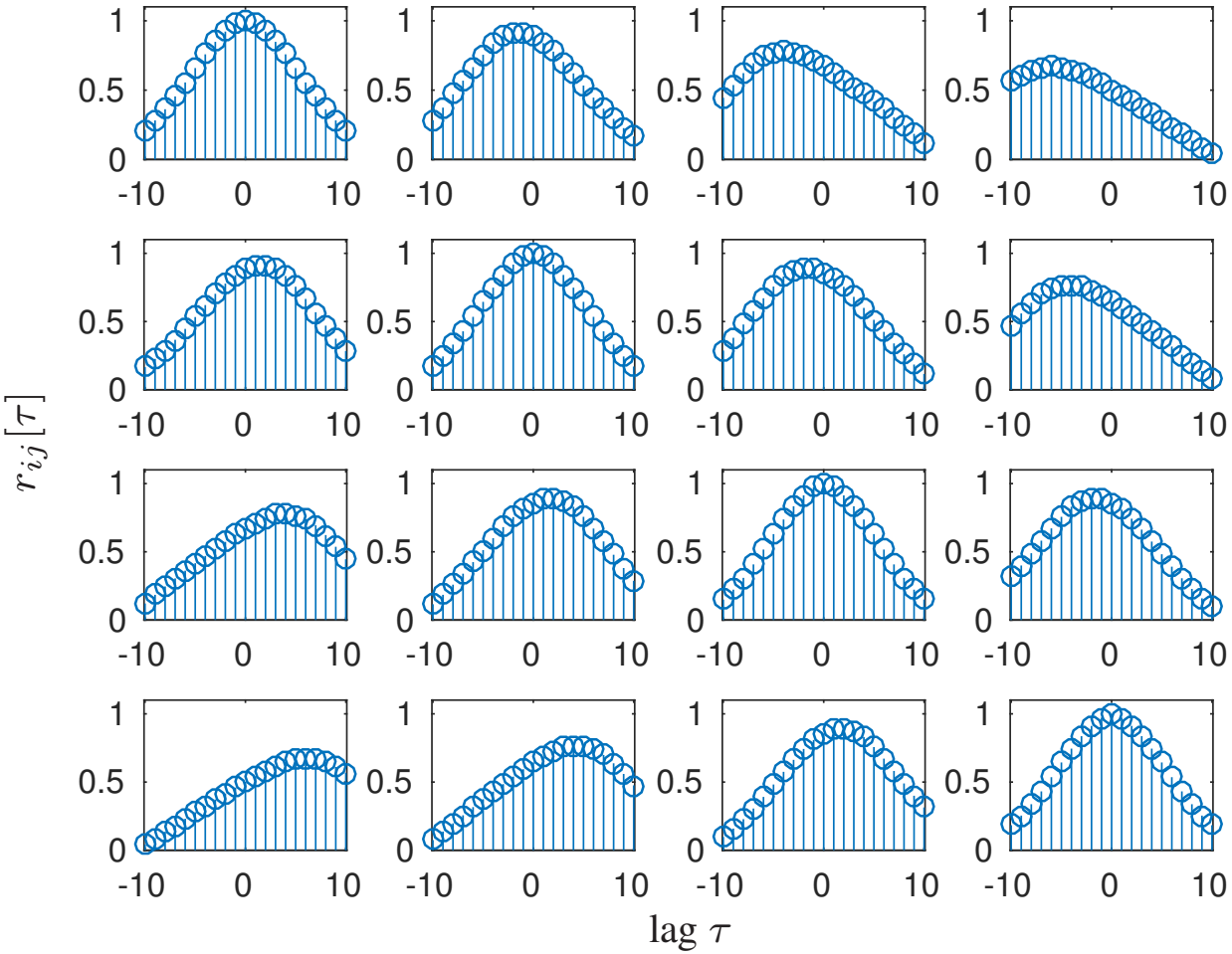
$$x_m(n) = \mathbf{h}_m^T \mathbf{s}_0(n) + v_m(n) \tag{3}$$

To capture the temporal correlations of the speech signals at different microphones, we require a space-time covariance matrix

$$\boldsymbol{\mathcal{R}}_{\mathbf{xx}}(\tau) = \mathbb{E}\{\mathbf{x}(n)\mathbf{x}^T(n-\tau)\}, \tag{4}$$

where

$$\mathbf{x}^T(n) = [x_1(n), x_2(n) \cdots x_M(n)] \tag{5}$$

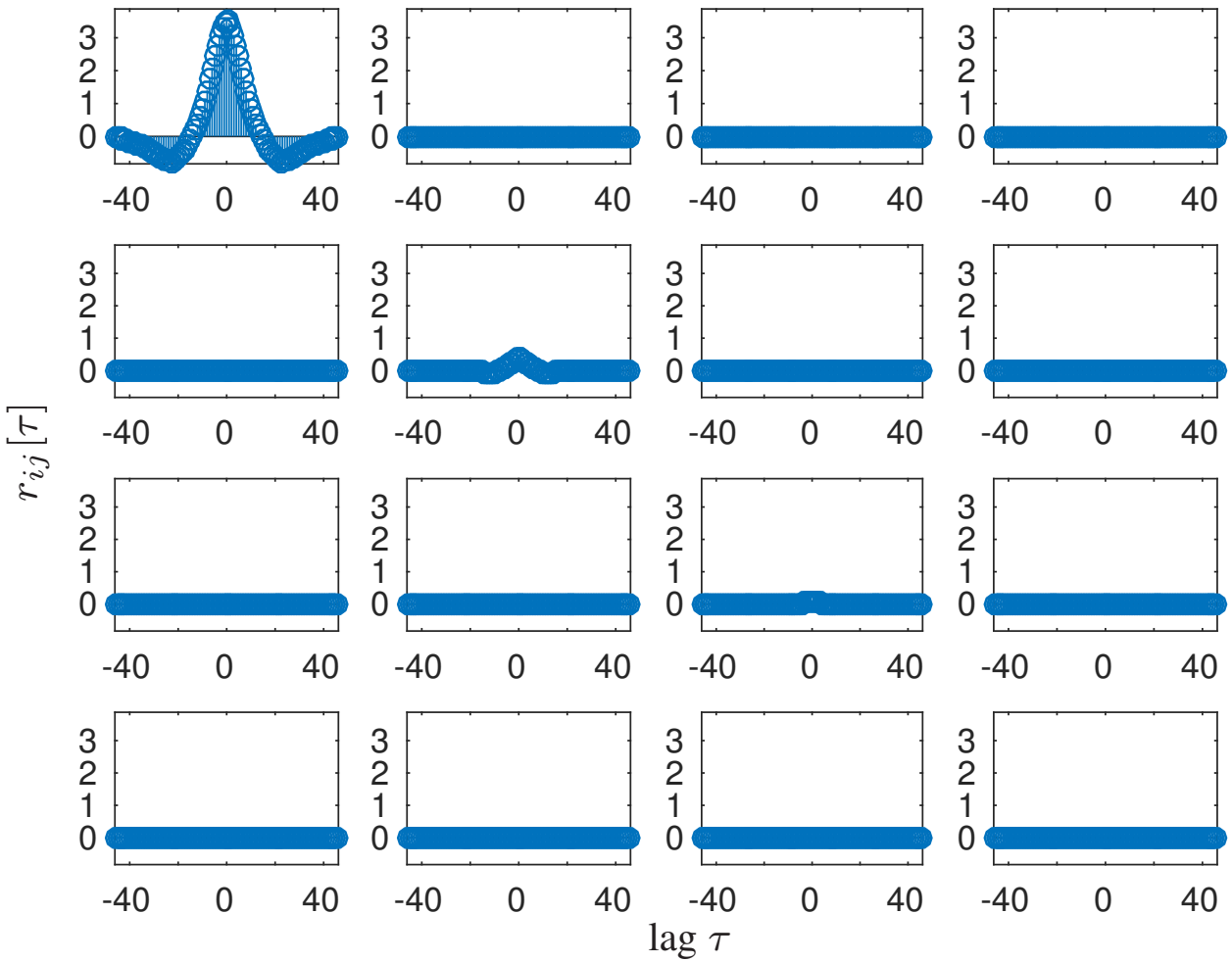


Space-time covariance matrix $\boldsymbol{\mathcal{R}}_{\mathbf{xx}}(z)$ on the interval $|\tau| \leq 10$.

Due to the signal model the PEVD of (4) can be partitioned by thresholding the eigenvalues to give

$$\tilde{\boldsymbol{\mathcal{R}}}_{\mathbf{xx}}(z) \approx \left[\boldsymbol{\mathcal{U}}_{\tilde{s}}(z) \middle| \boldsymbol{\mathcal{U}}_{\tilde{v}}(z)\right] \left[\begin{array}{c|c} \boldsymbol{\Lambda}_{\tilde{s}}(z) & \mathbf{0} \\ \hline \mathbf{0} & \boldsymbol{\Lambda}_{\tilde{v}}(z) \end{array}\right] \left[\begin{array}{c} \boldsymbol{\mathcal{U}}^{P}{}_{\tilde{s}}(z) \\ \hline \boldsymbol{\mathcal{U}}^{P}{}_{\tilde{v}}(z) \end{array}\right],$$

where $\{.\}_{\tilde{s}}$ and $\{.\}_{\tilde{v}}$ represent the orthogonal signal and noise subspace components.



Diagonalised matrix of eigenvectors produced by the SMD algorithm for the noise-free case with a direction of arrival (DoA) of 45°.

Generalising from MUSIC, the following quantity is defined

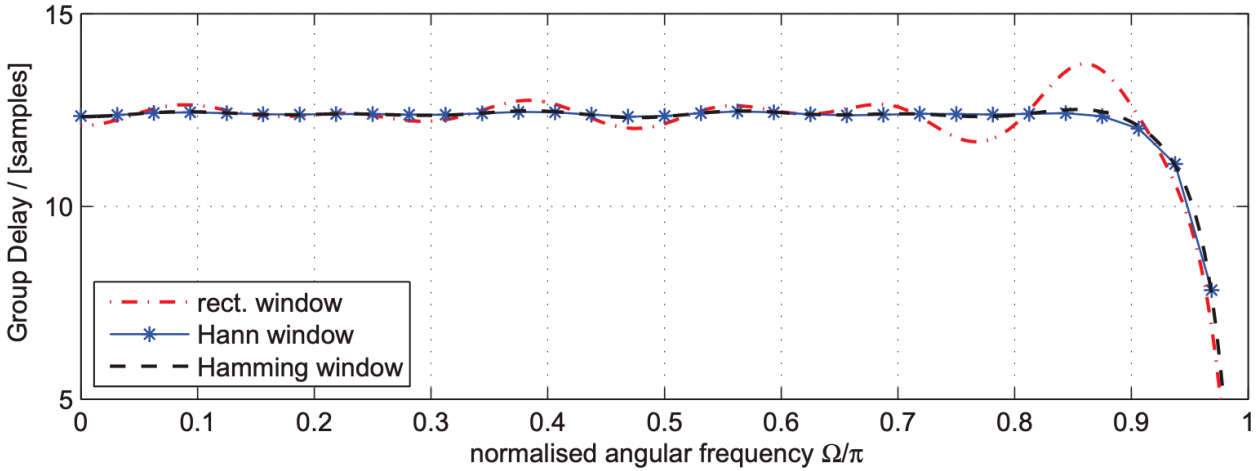
$$\Gamma_\theta(z) = \mathbf{a}_\theta^P(z)\boldsymbol{\mathcal{U}}_v(z)\boldsymbol{\mathcal{U}}_v^P(z)\mathbf{a}_\theta(z), \tag{6}$$

which is used to compute the pseudo-spectrogram for Spatio-Spectral P-MUSIC,

$$\mathbf{P}_{SSP-MU}(\theta,\Omega) = \left.\frac{1}{\Gamma_\theta(z)}\right|_{z=e^{-j\Omega}}, \tag{7}$$

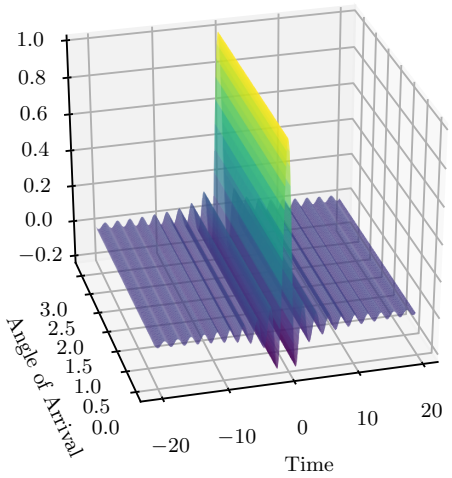
where frequency Ω is obtained by evaluating z on the unit circle.

CONTRIBUTIONS

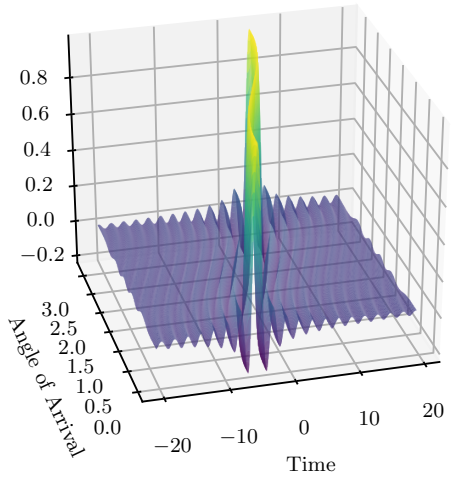


(Group delay response for tapered windows & rectangular window for fractional delay τ = 2.4)

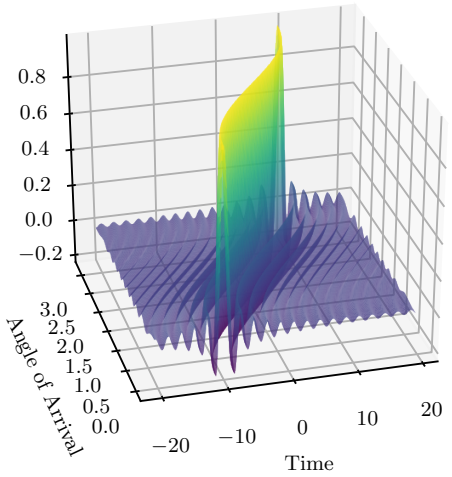
- A broadband steering vector can be realised by sampling a sinc function which leads to an infinite impulse response
- truncation results in inaccuracies towards $\frac{f_s}{2}$ (Alrmah, 2015)
- using a tapered window to create a finite length filter dramatically improves the accuracy (Selva, IEEE TSP 2008)



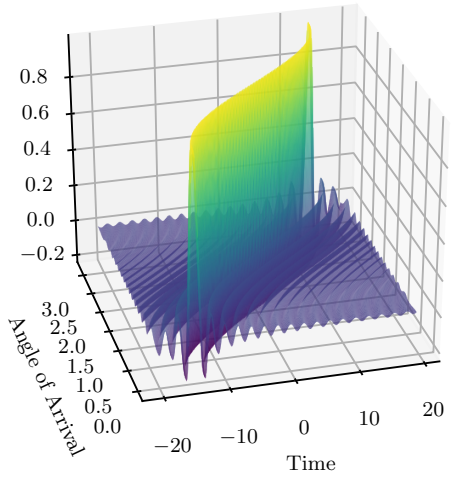
Broadband Steering Vector for m=0



Broadband Steering Vector for m=1



Broadband Steering Vector for m=2

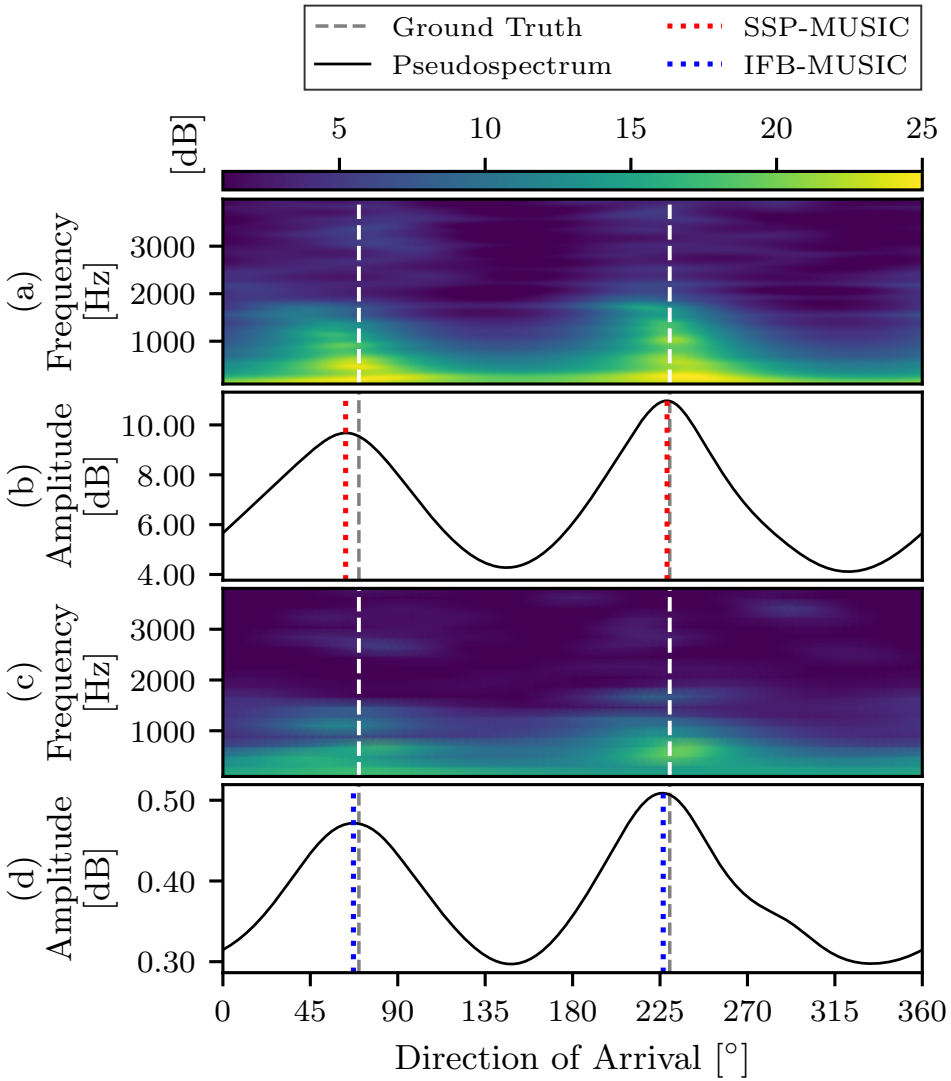


Broadband Steering Vector for m=3

We modify SSP-MUSIC to only include the direct-path to reduce the impact of reverberation on the direction of arrival (DoA) estimation.

If microphone array geometry is known, the largest direct-path delay can inform the choice of the temporal lag support, W.

$$\tilde{\boldsymbol{\mathcal{R}}}_{\mathbf{xx}}(z) \approx \sum_{\tau=-W}^{W} \mathbf{R}_{\mathbf{xx}}(\tau) z^{-\tau}, \tag{8}$$



Illustrative example of a 100 ms frame from Exp-2 (SNR: -10 dB, T60: 0.25 s).

(a) pseudo-spectrogram of SSP-MUSIC,

(b) pseudo-spectrum of SSP-MUSIC,

(c) pseudo-spectrogram of IFB-MUSIC,

(d) pseudo-spectrum of IFB-MUSIC.

RESULTS

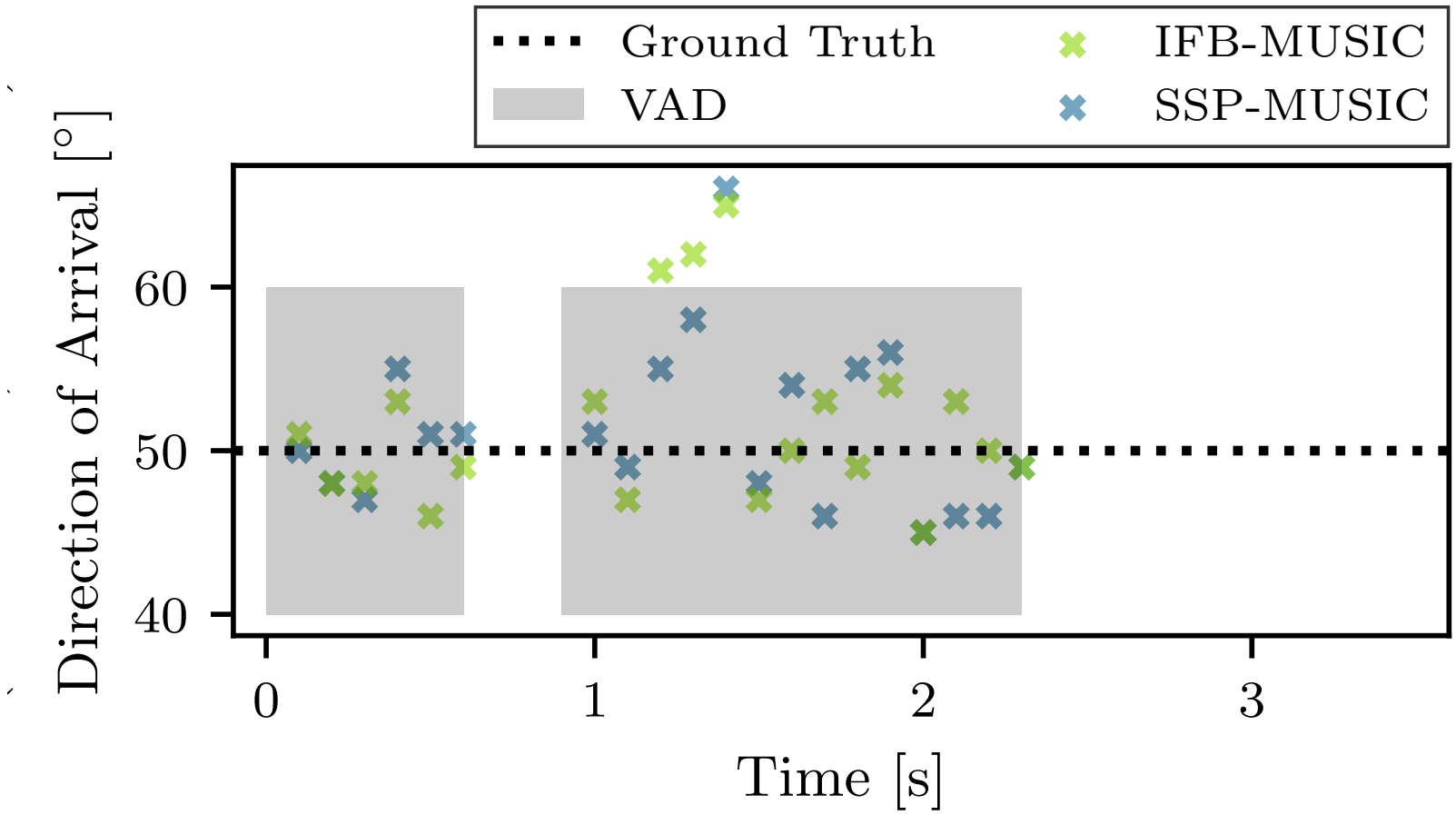
A '**HIT**' is when a sound source (speaker) has been detected once within a $\pm 15^{\circ}$ collar applied around the ground-truth azimuth.

A '**MISS**' is when a speaker change has not been detected within this collar and a FA is when a detection falls outside of a ground-truth azimuth collar.

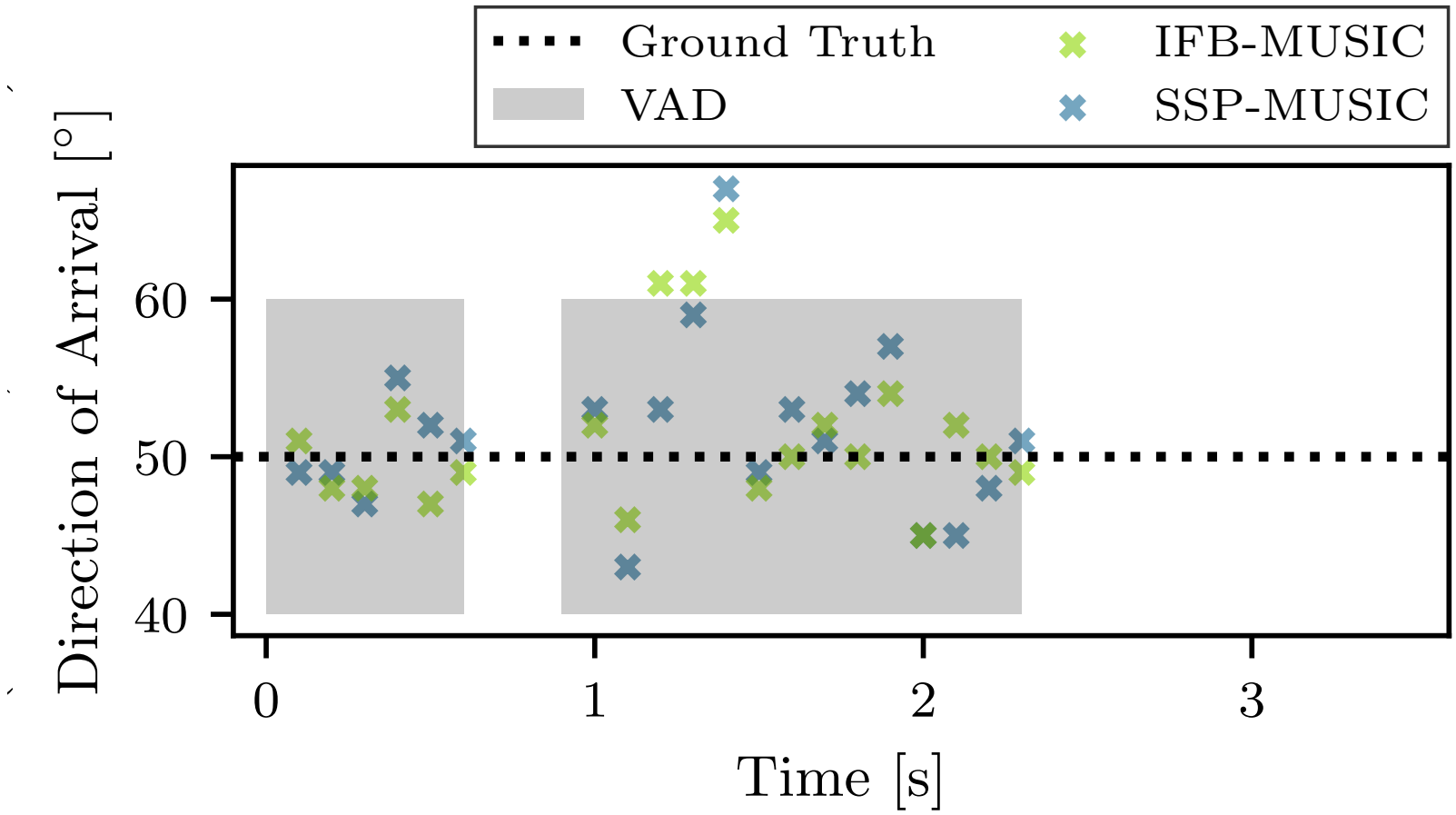
The **Hit Rate** and **False Alarm Rate** are, therefore, defined as,

$$\text{Hit Rate} = \frac{\text{HITs}}{\text{HITs} + \text{MISSs}}\%, \quad \text{False Alarm Rate} = \frac{\text{FAs}}{\text{HITs} + \text{FAs}}\% \tag{9}$$

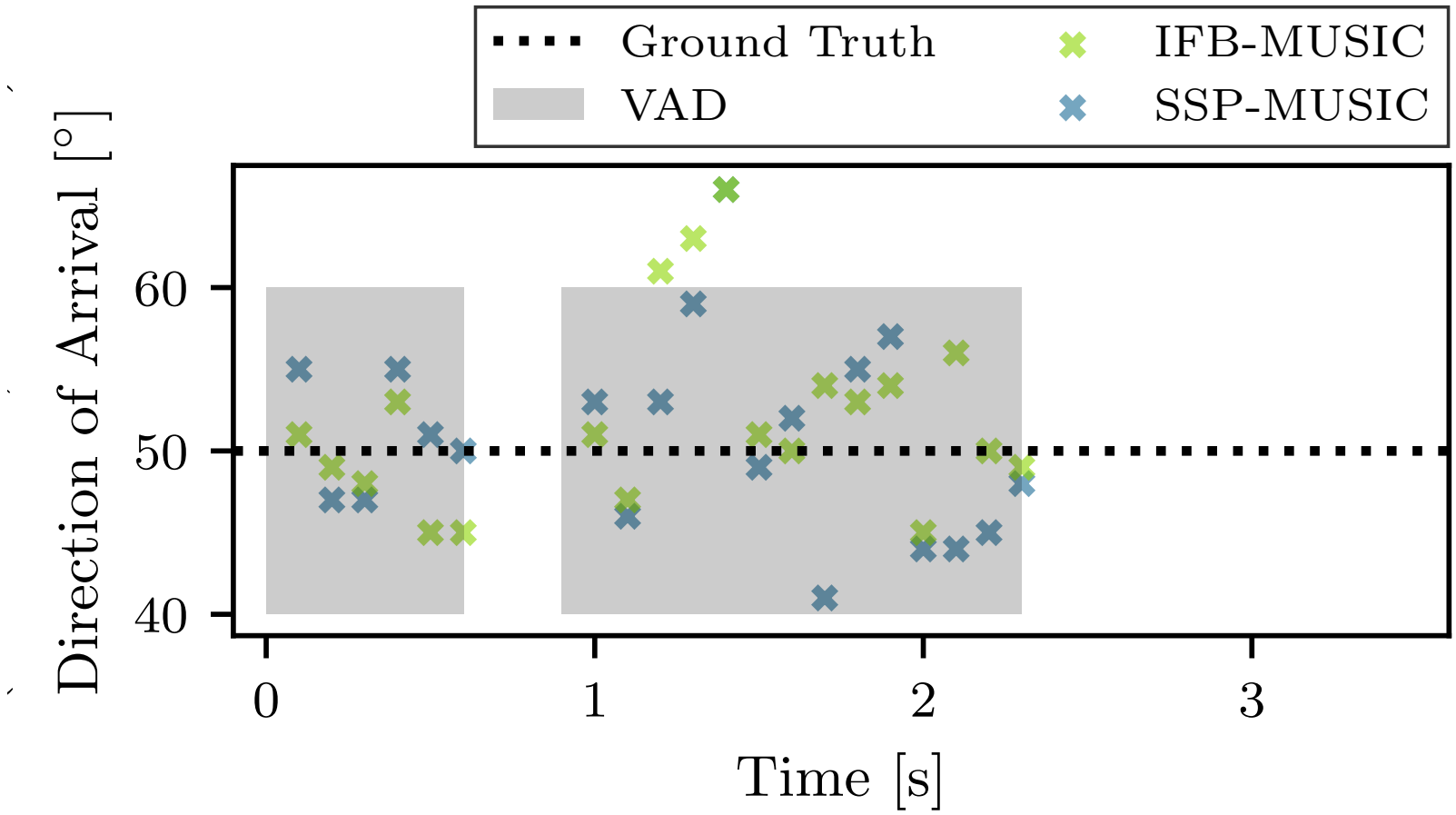
SNR: 25 dB, T60: 0.25 s



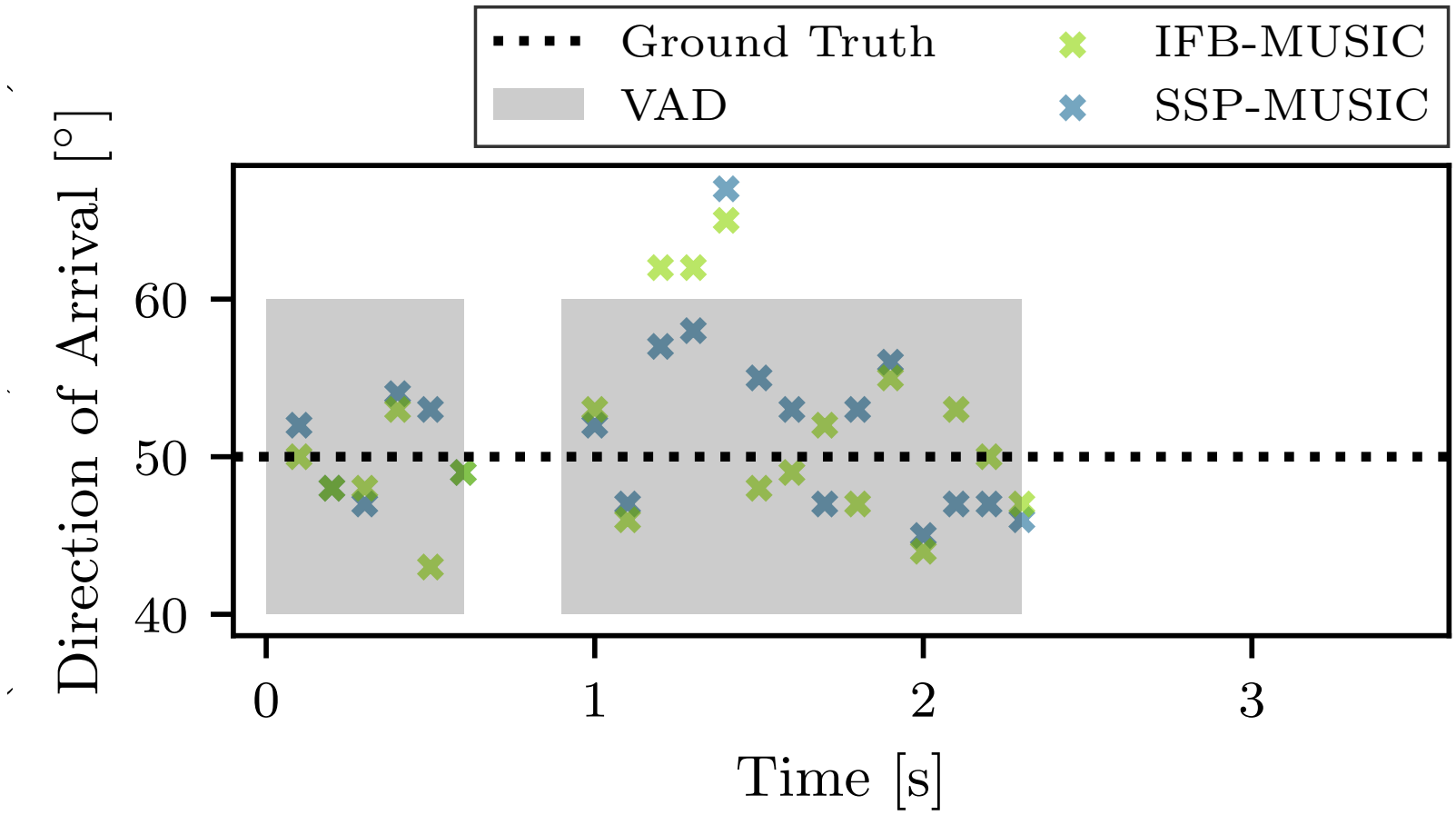
SNR: 20 dB, T60: 0.25 s



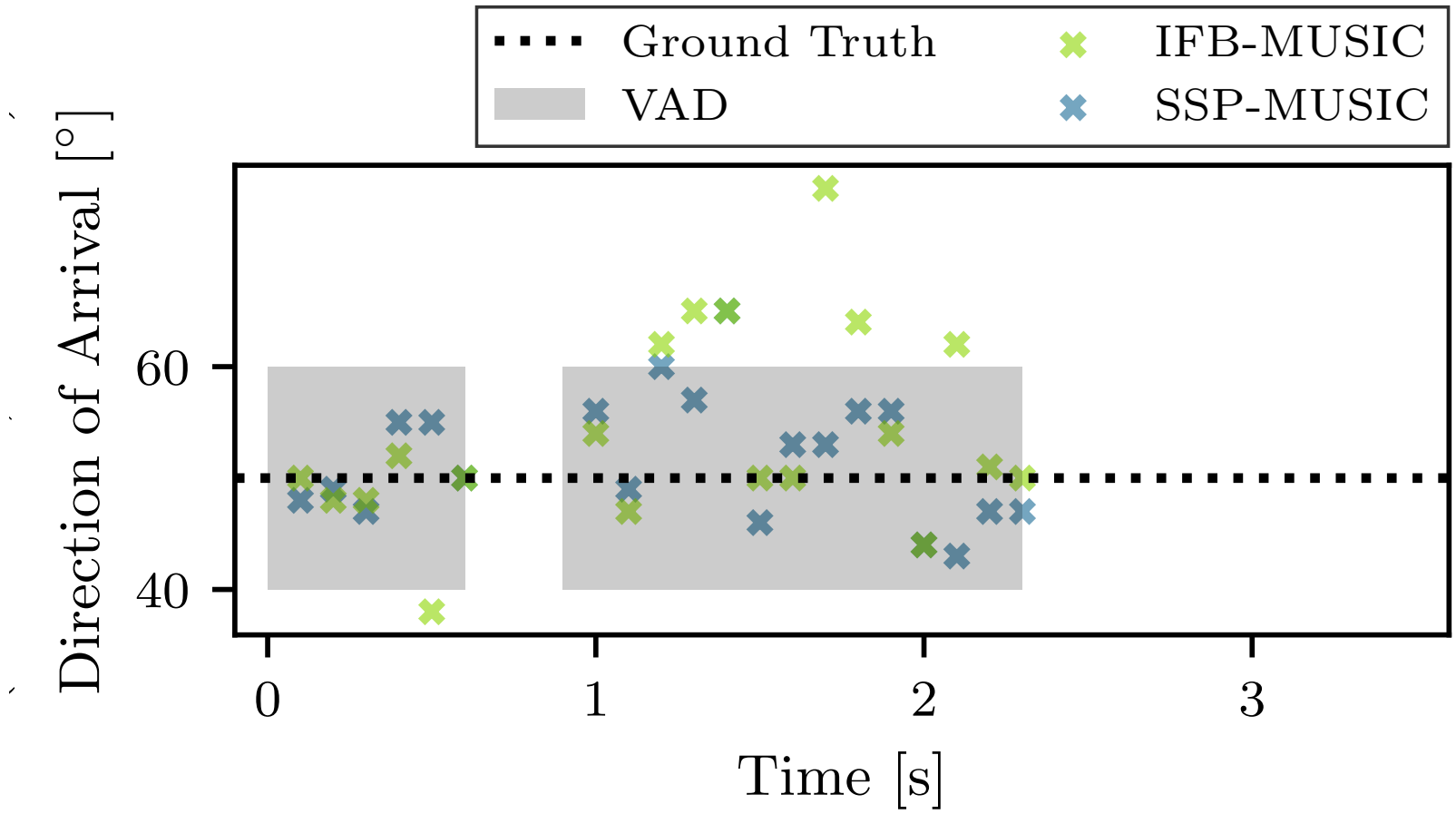
SNR: 15 dB, T60: 0.25 s



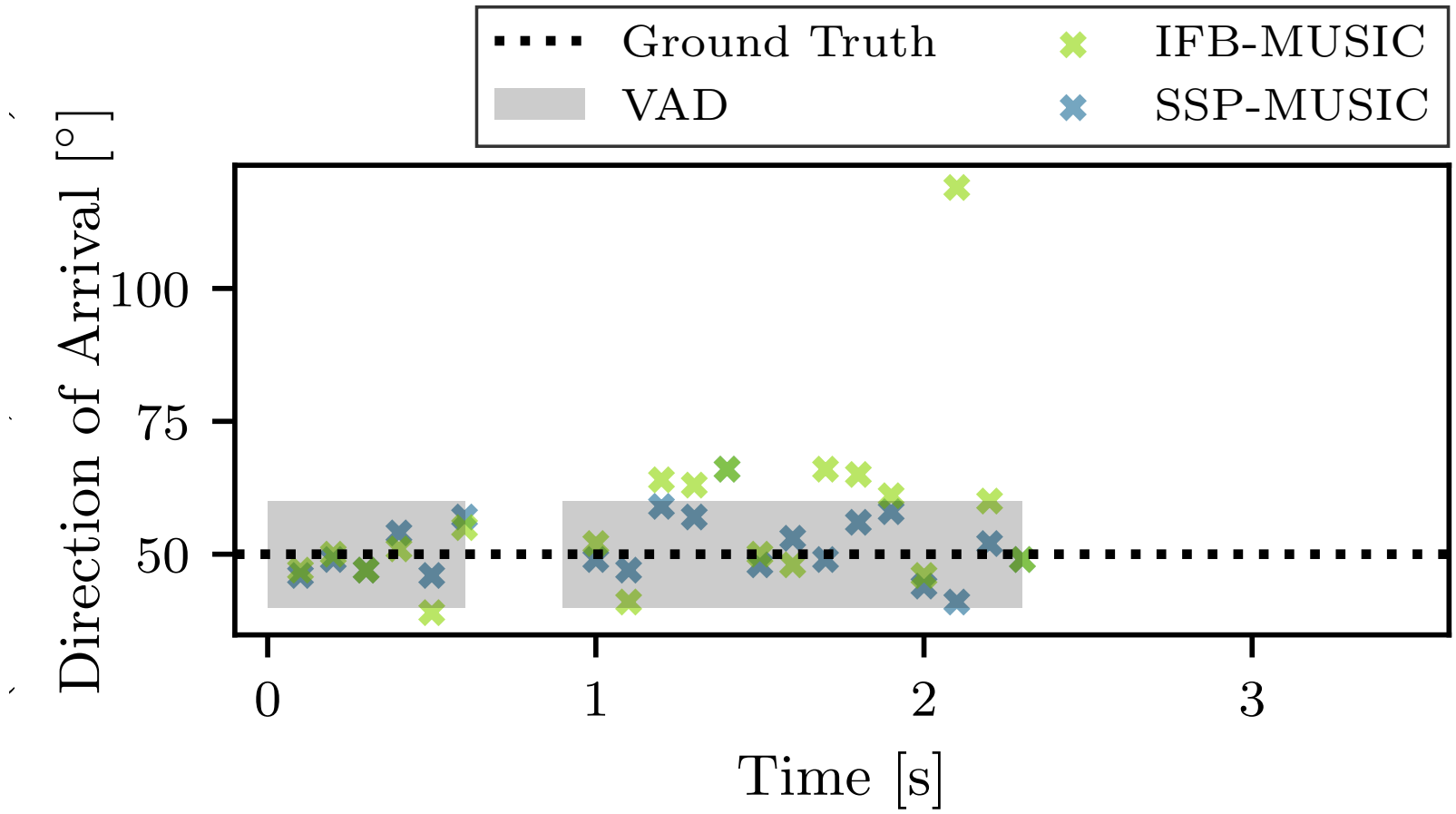
SNR: 10 dB, T60: 0.25 s



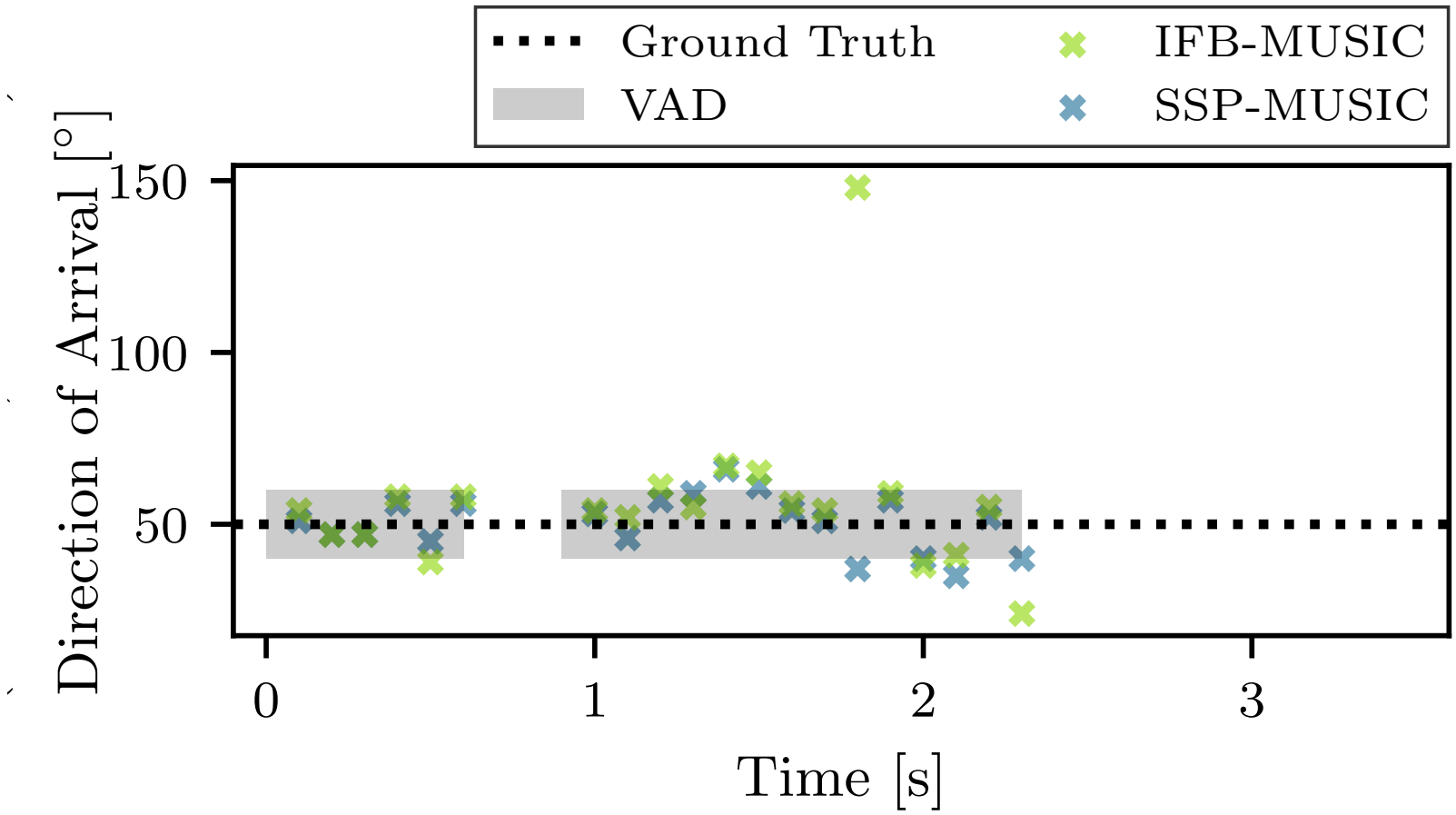
SNR: 5 dB, T60: 0.25 s



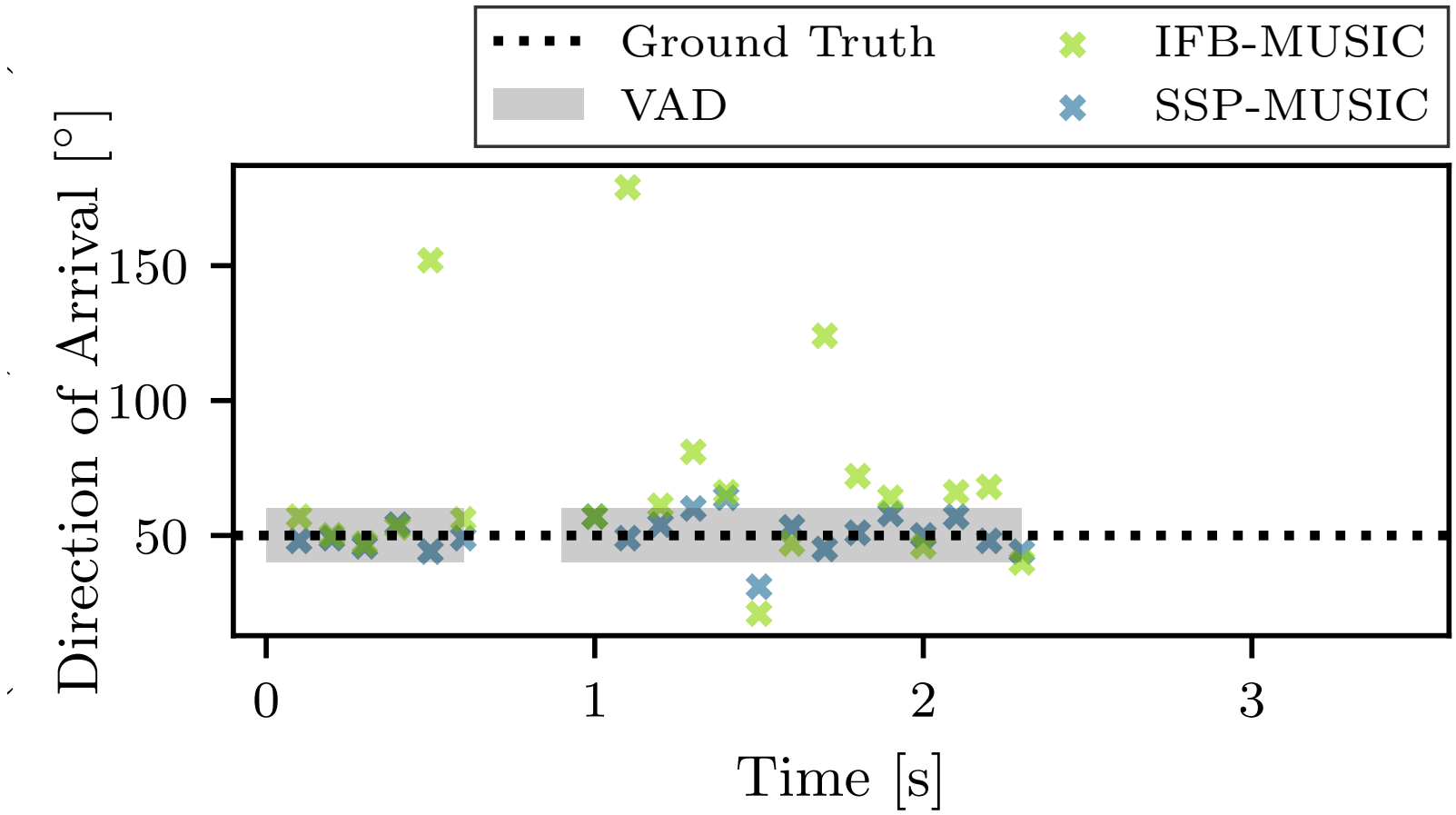
SNR: 0 dB, T60: 0.25 s



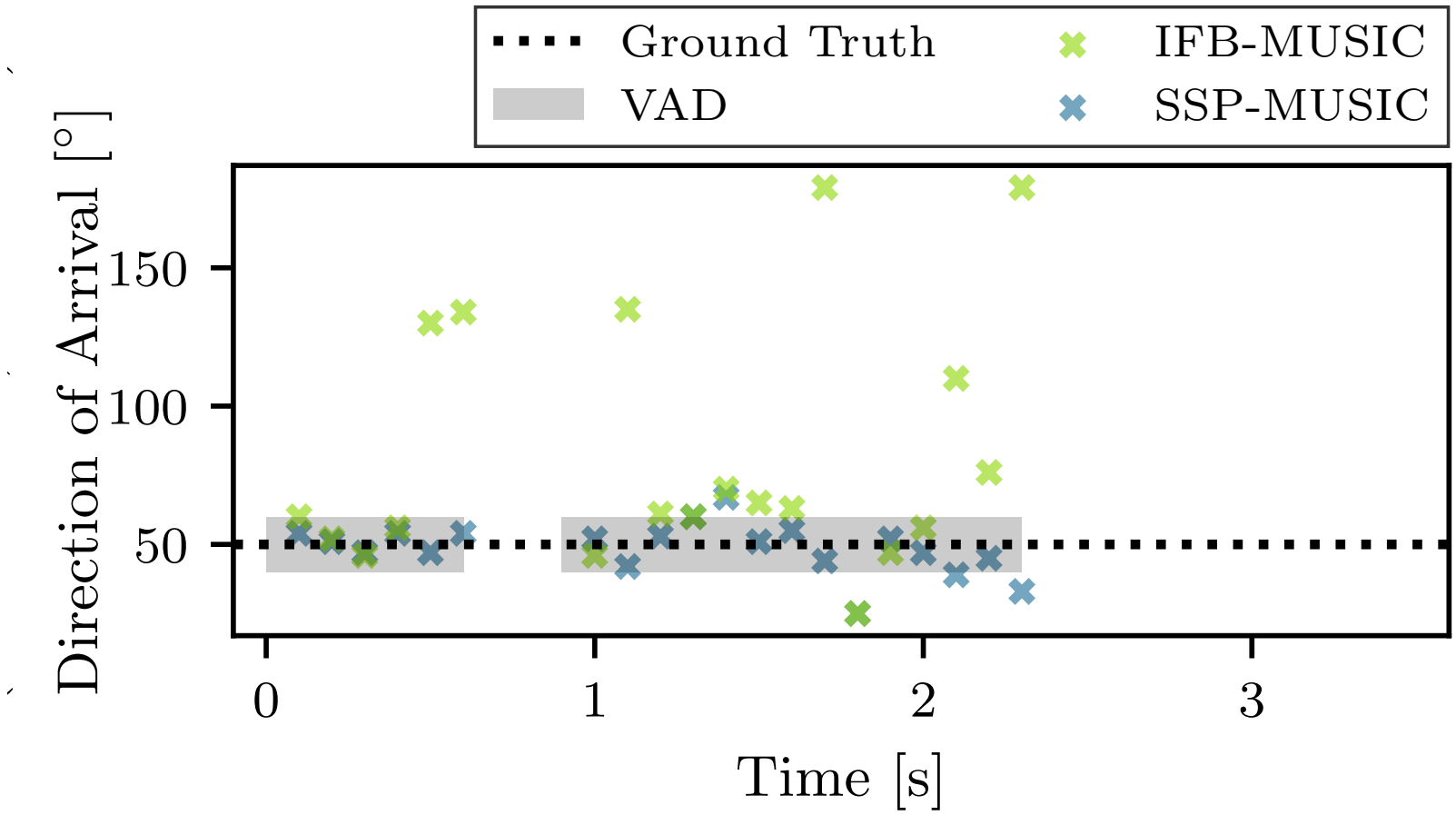
SNR: -5 dB, T60: 0.25 s



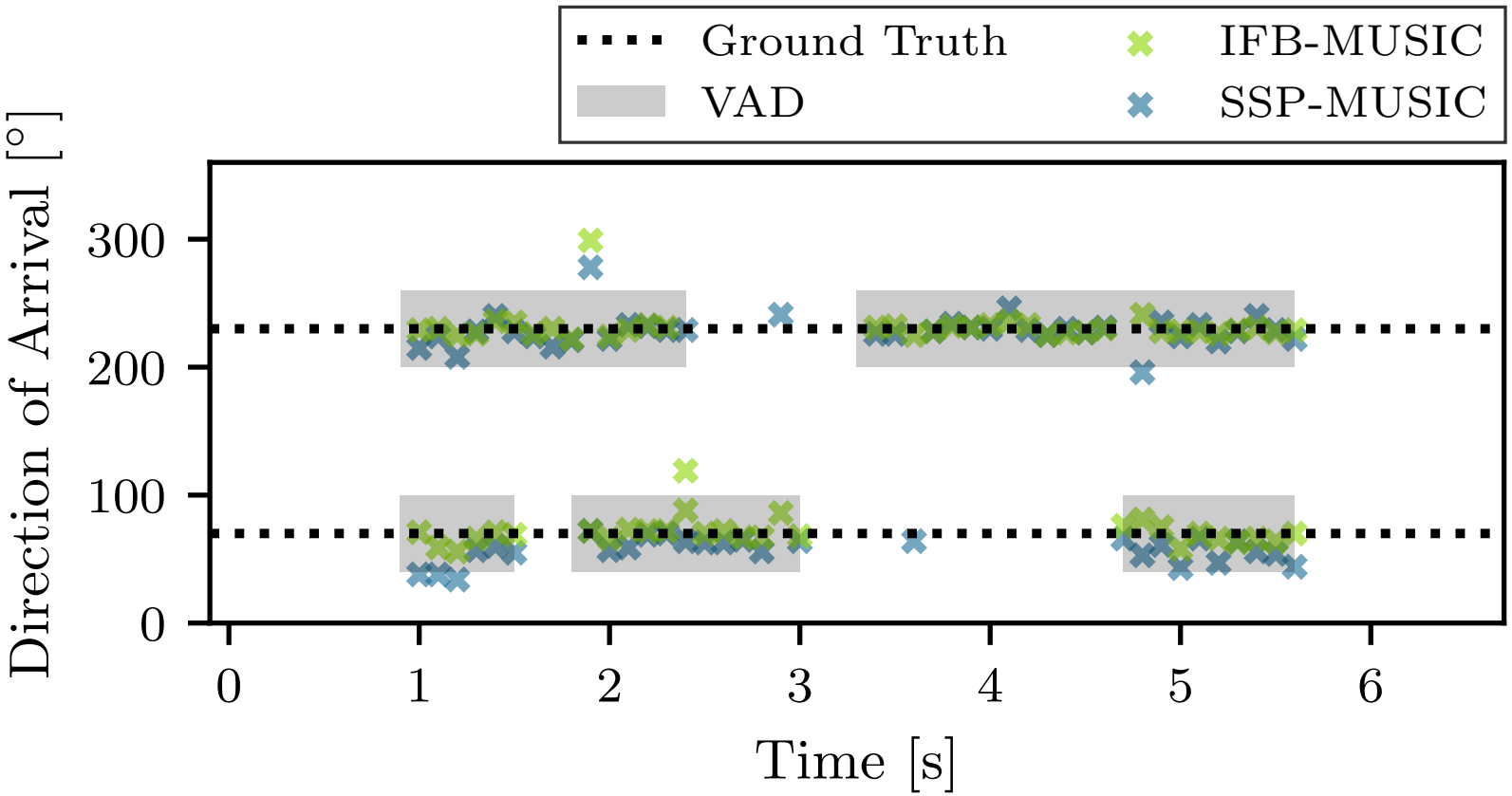
SNR: -10 dB, T60: 0.25 s



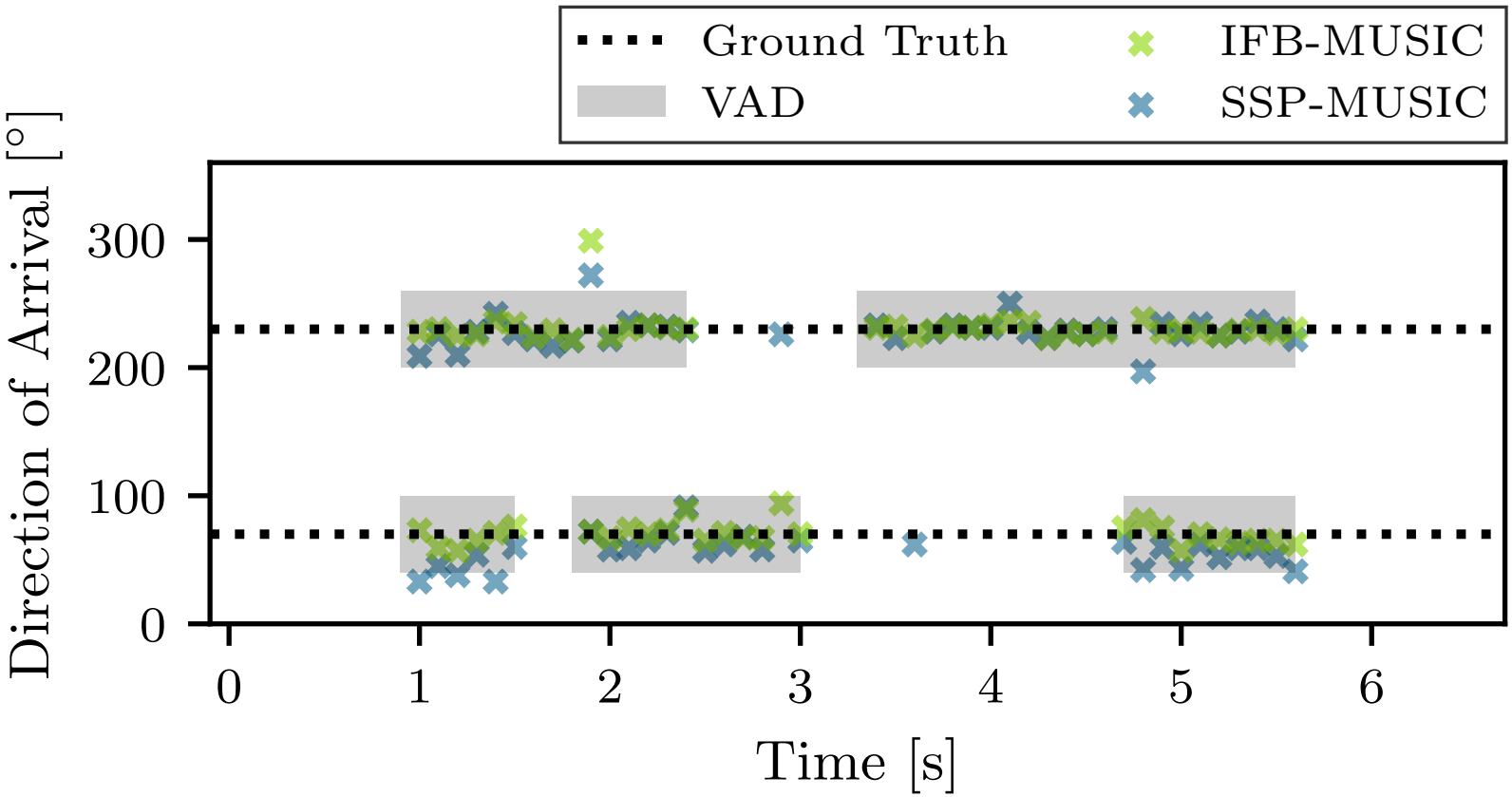
SNR: -15 dB, T60: 0.25 s



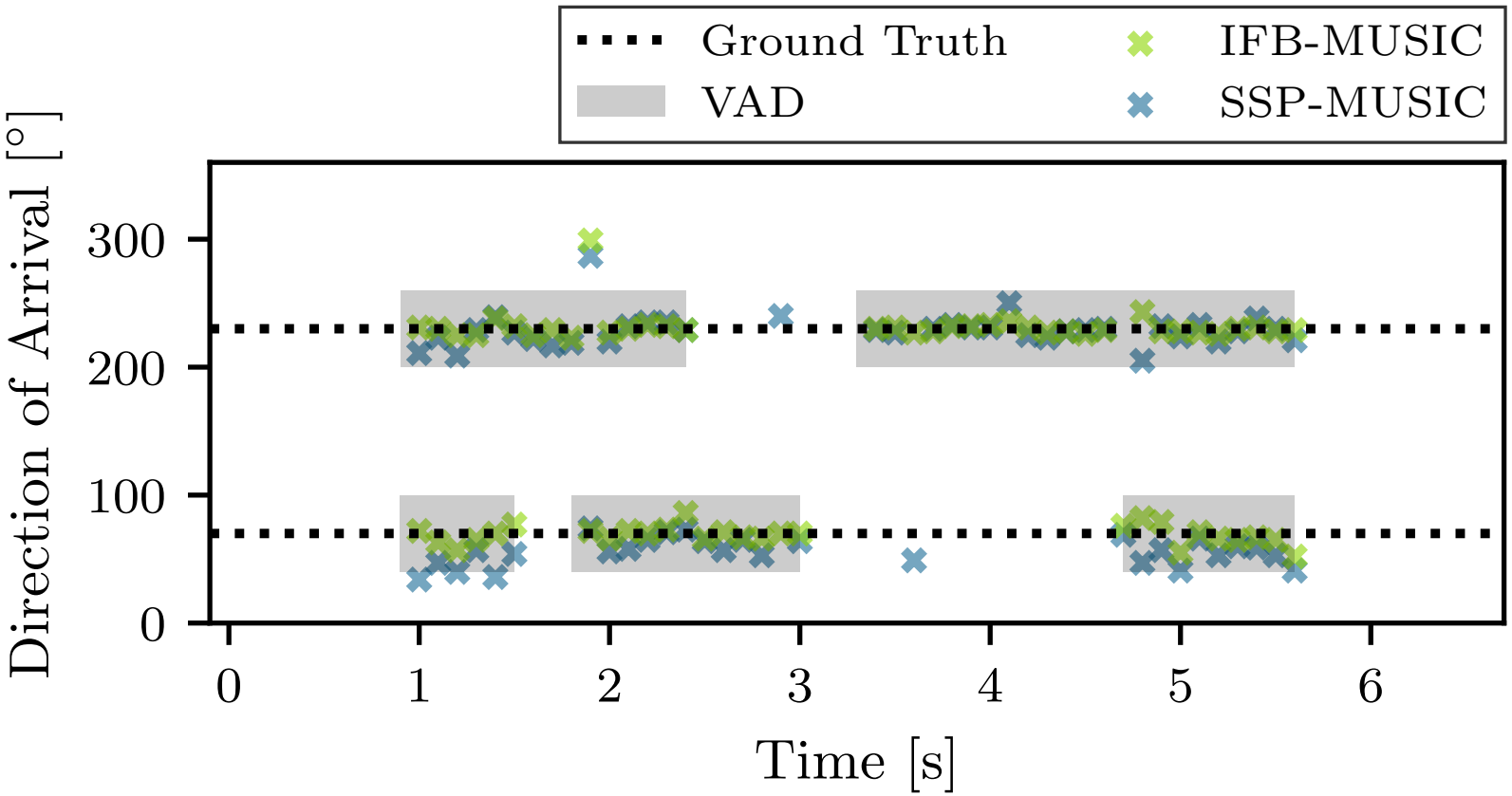
SNR: 25 dB, T60: 0.25 s



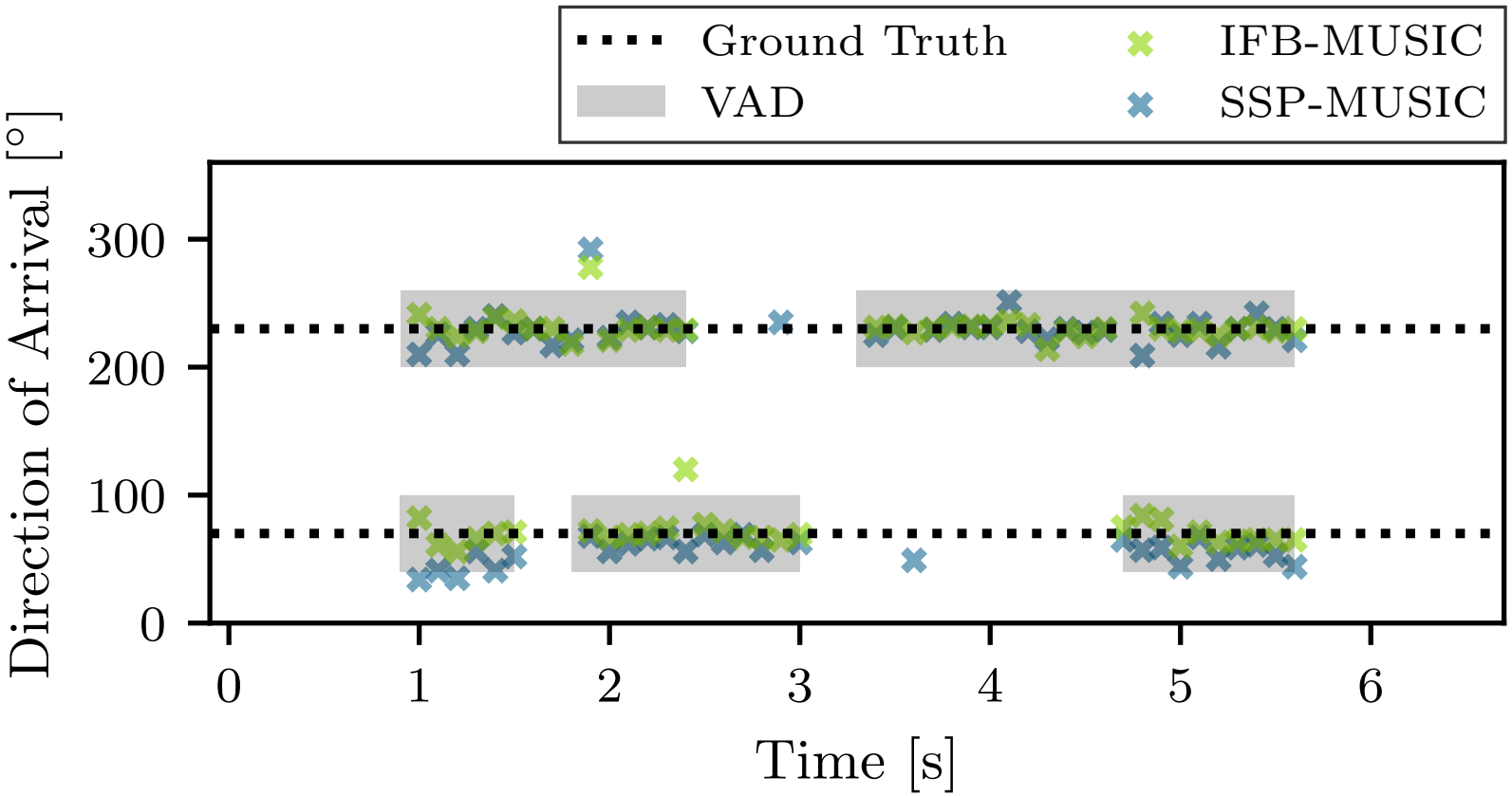
SNR: 20 dB, T60: 0.25 s



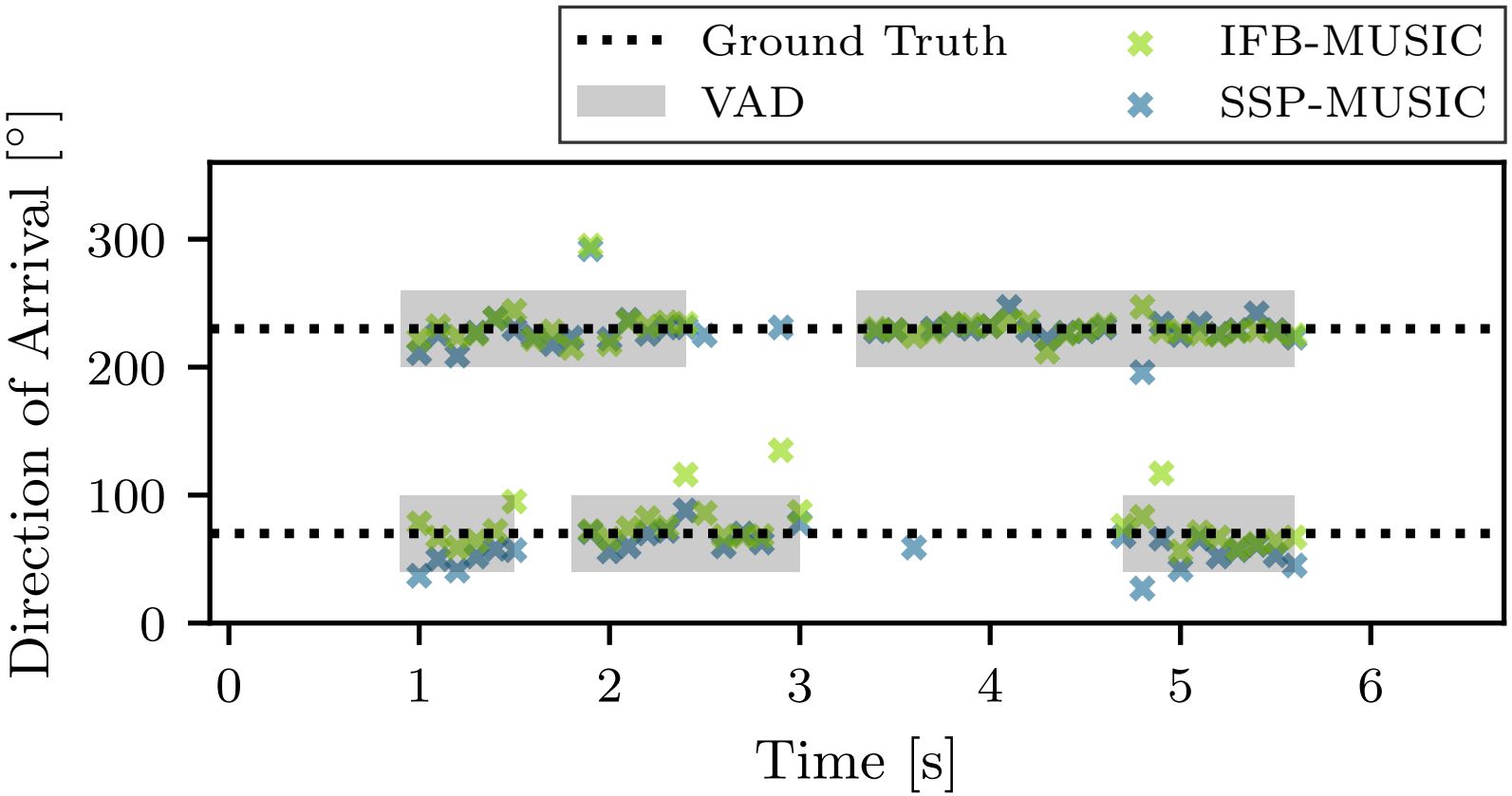
SNR: 15 dB, T60: 0.25 s



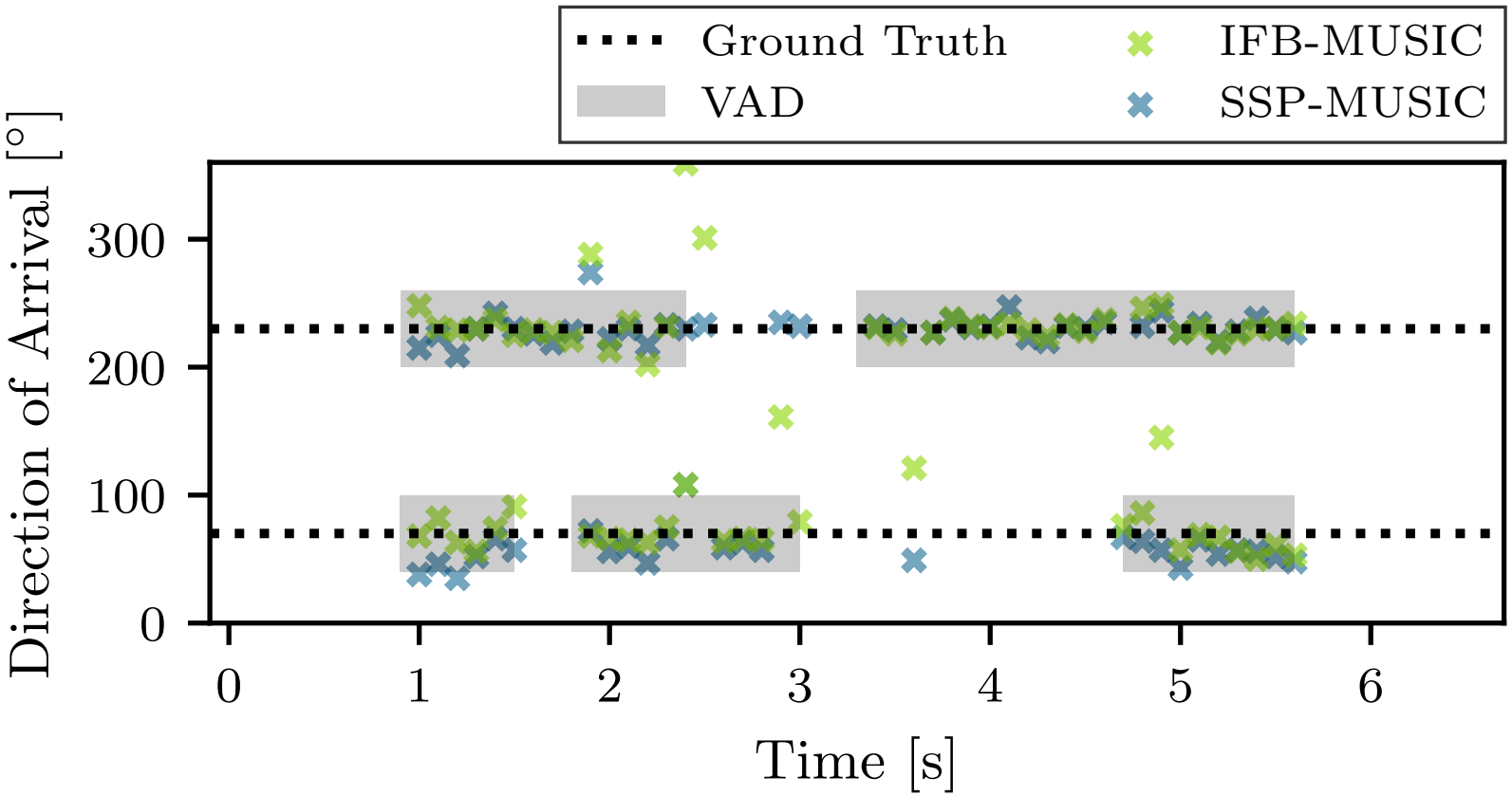
SNR: 10 dB, T60: 0.25 s



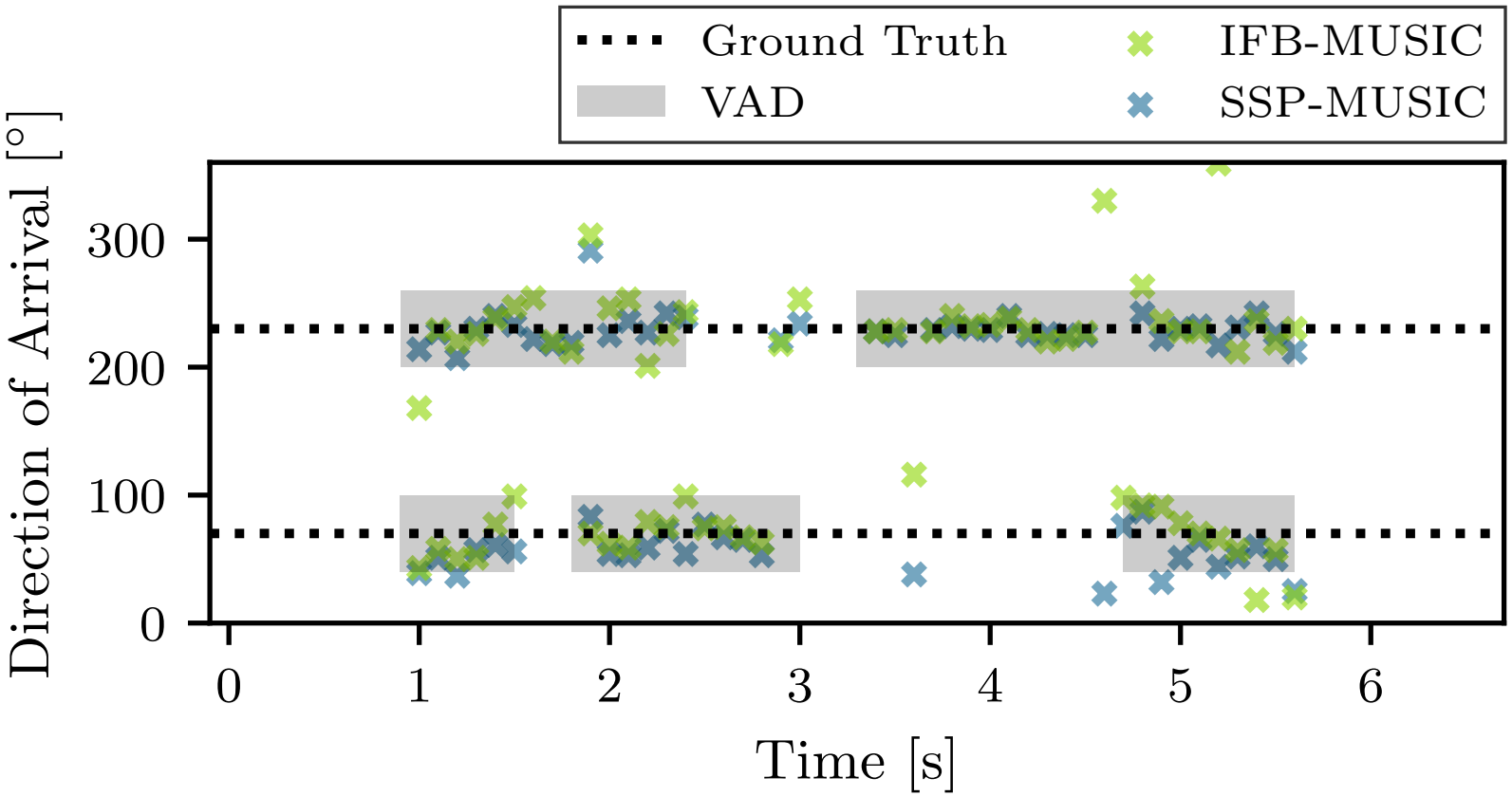
SNR: 5 dB, T60: 0.25 s



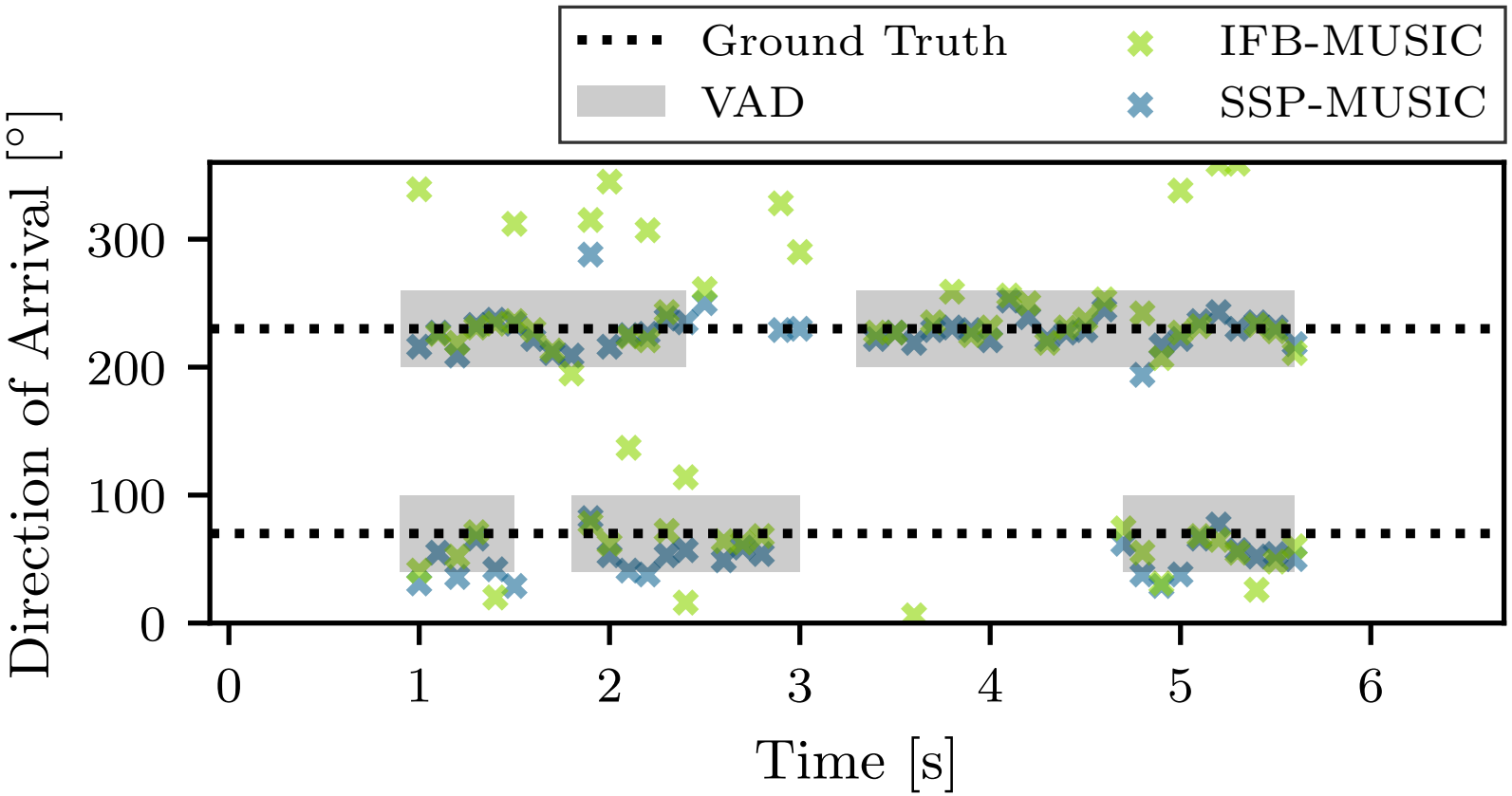
SNR: 0 dB, T60: 0.25 s



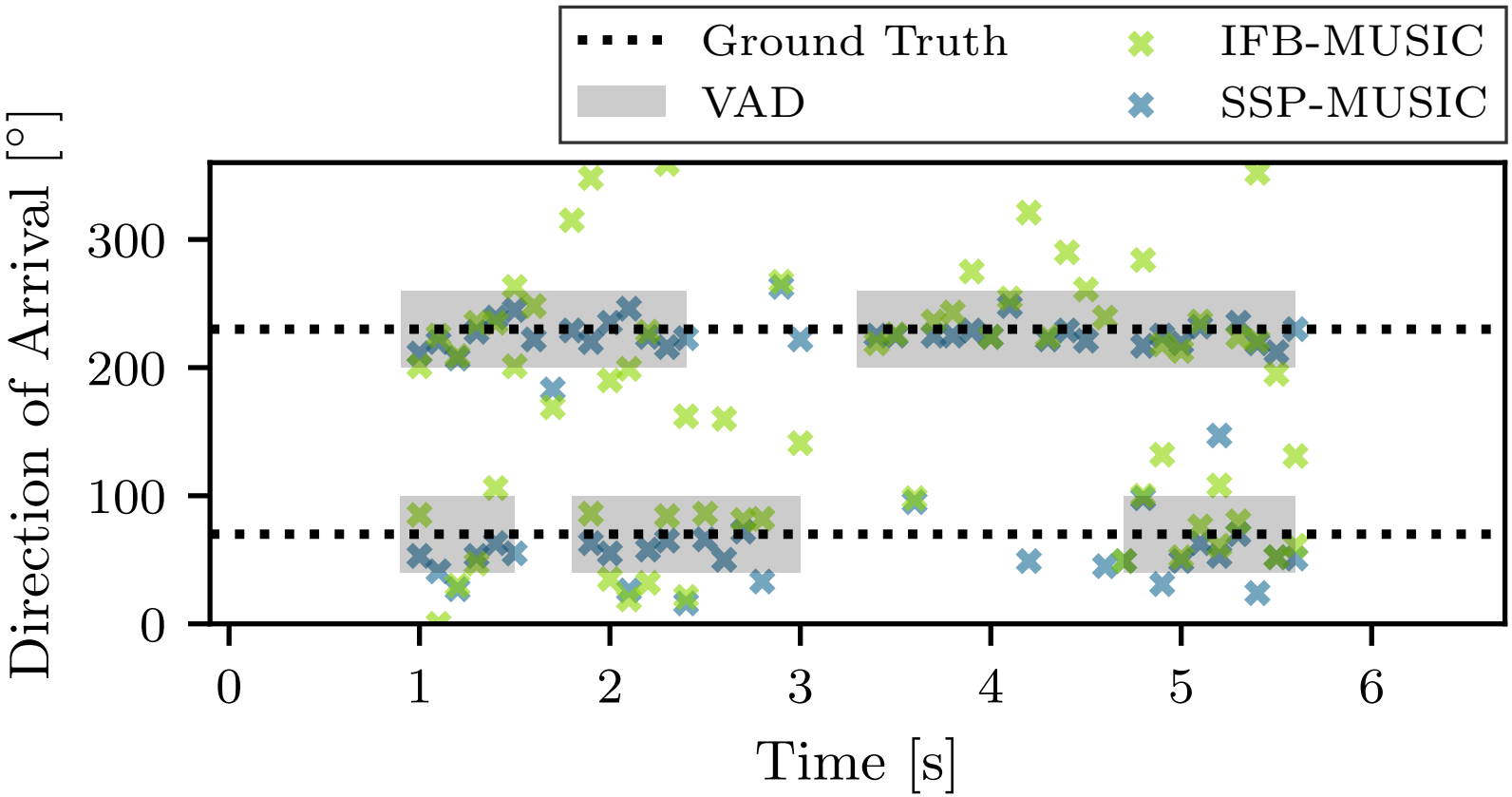
SNR: -5 dB, T60: 0.25 s



SNR: -10 dB, T60: 0.25 s

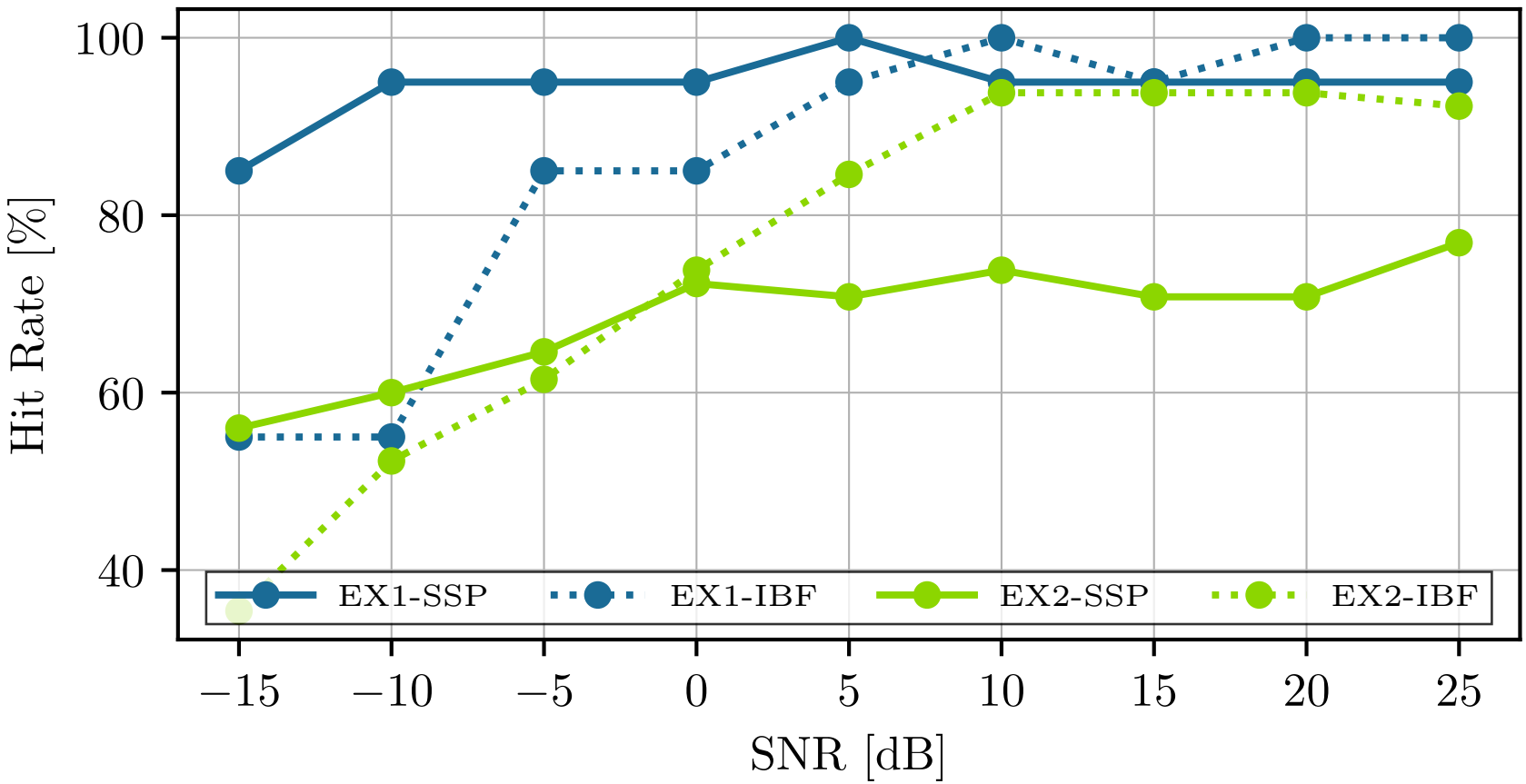


SNR: -15 dB, T60: 0.25 s

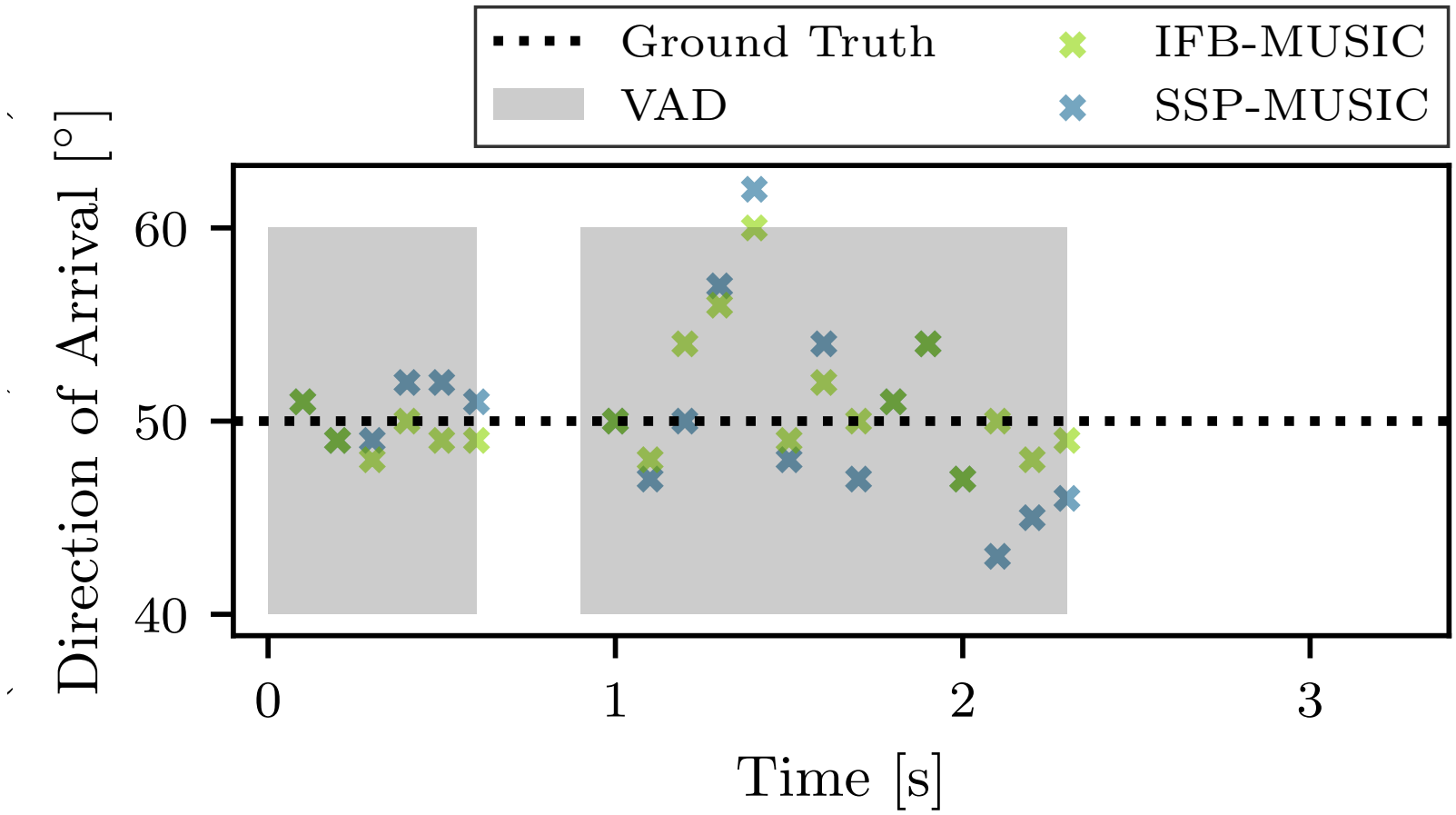


EX1: SINGLE SOUND SOURCE IN A SIMULATED ROOM

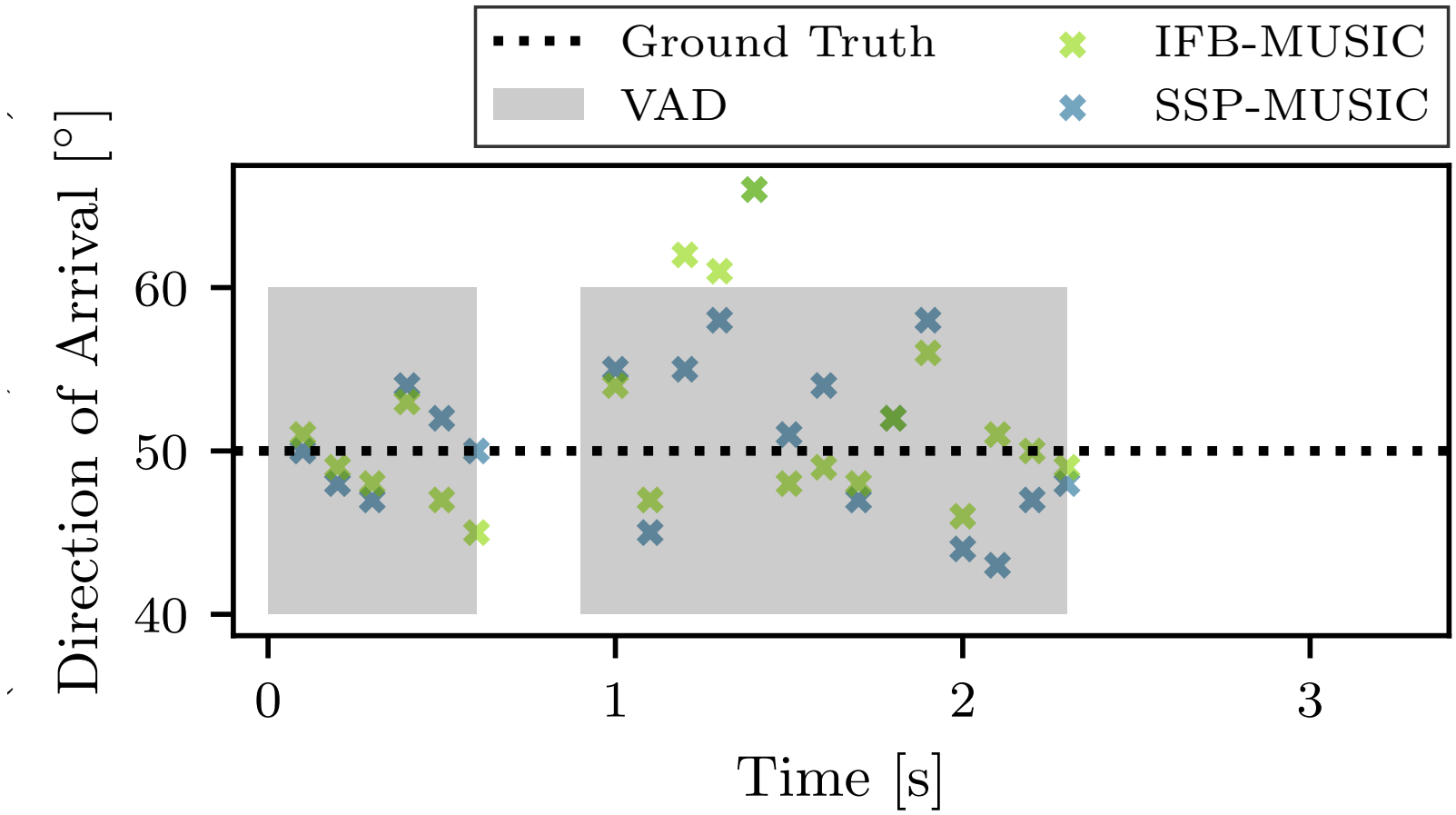
EX2: TWO SOUND SOURCES IN A SIMULATED ROOM



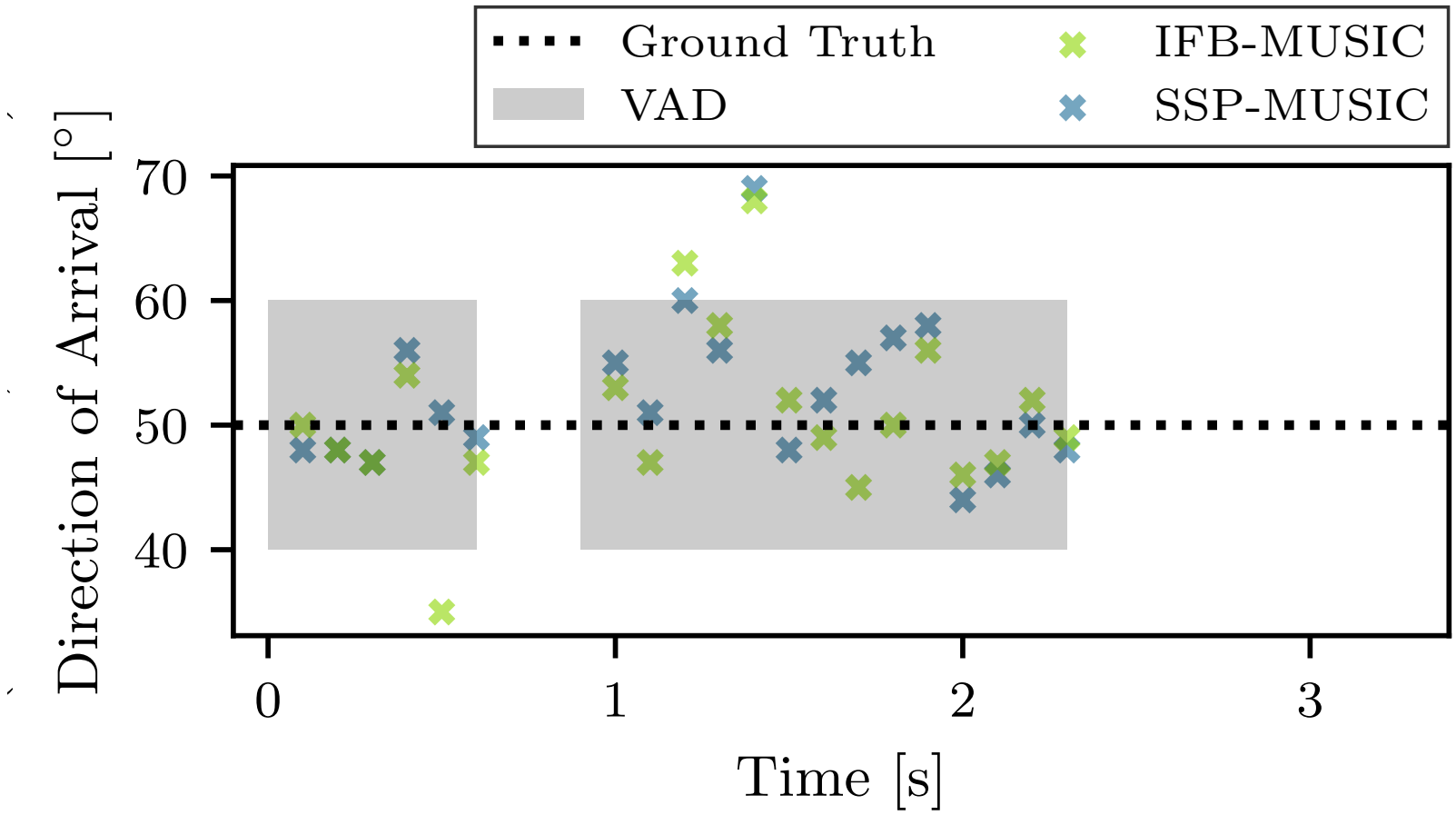
SNR: 15 dB, T60: 0.1 s



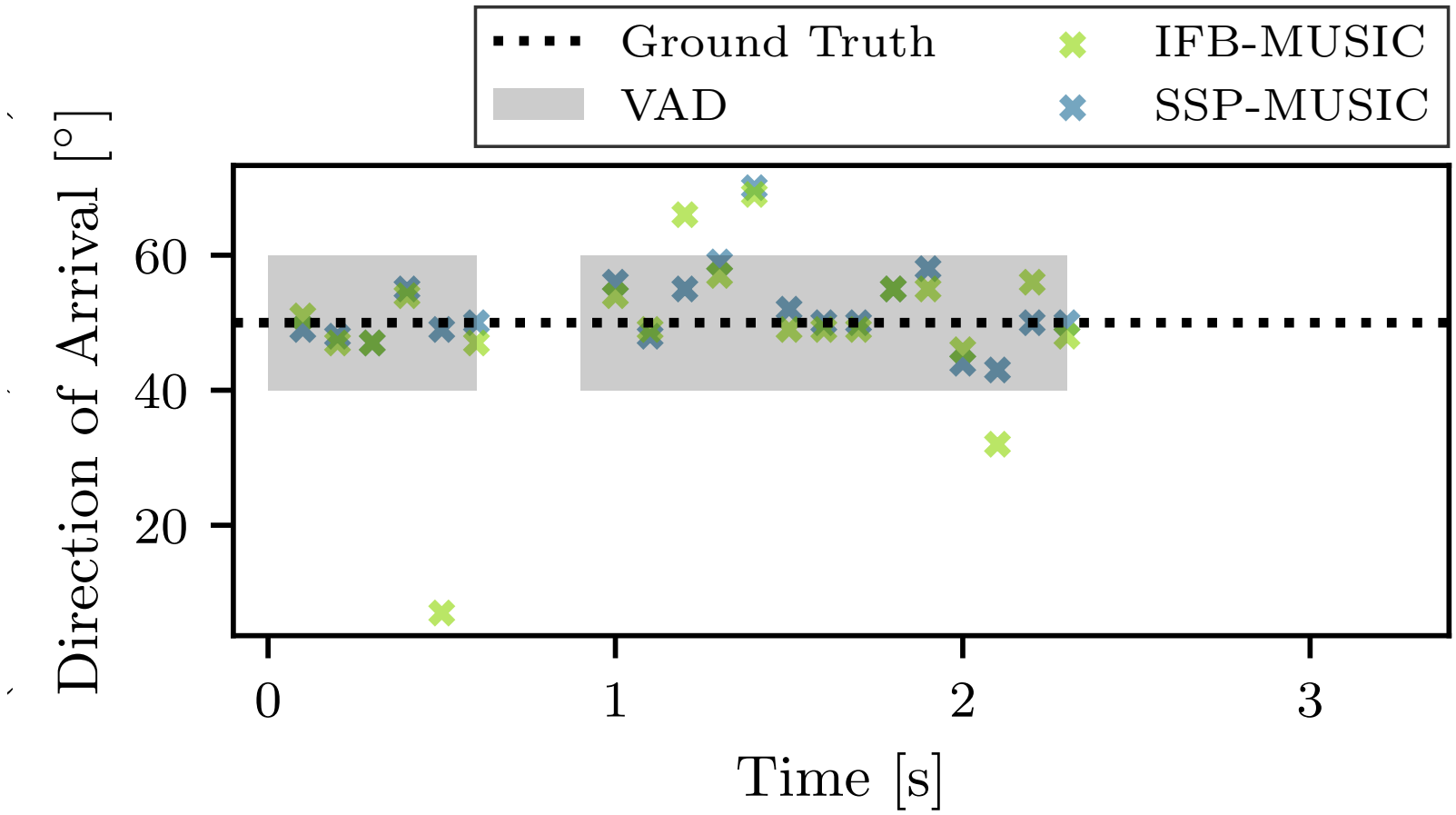
SNR: 15 dB, T60: 0.3 s



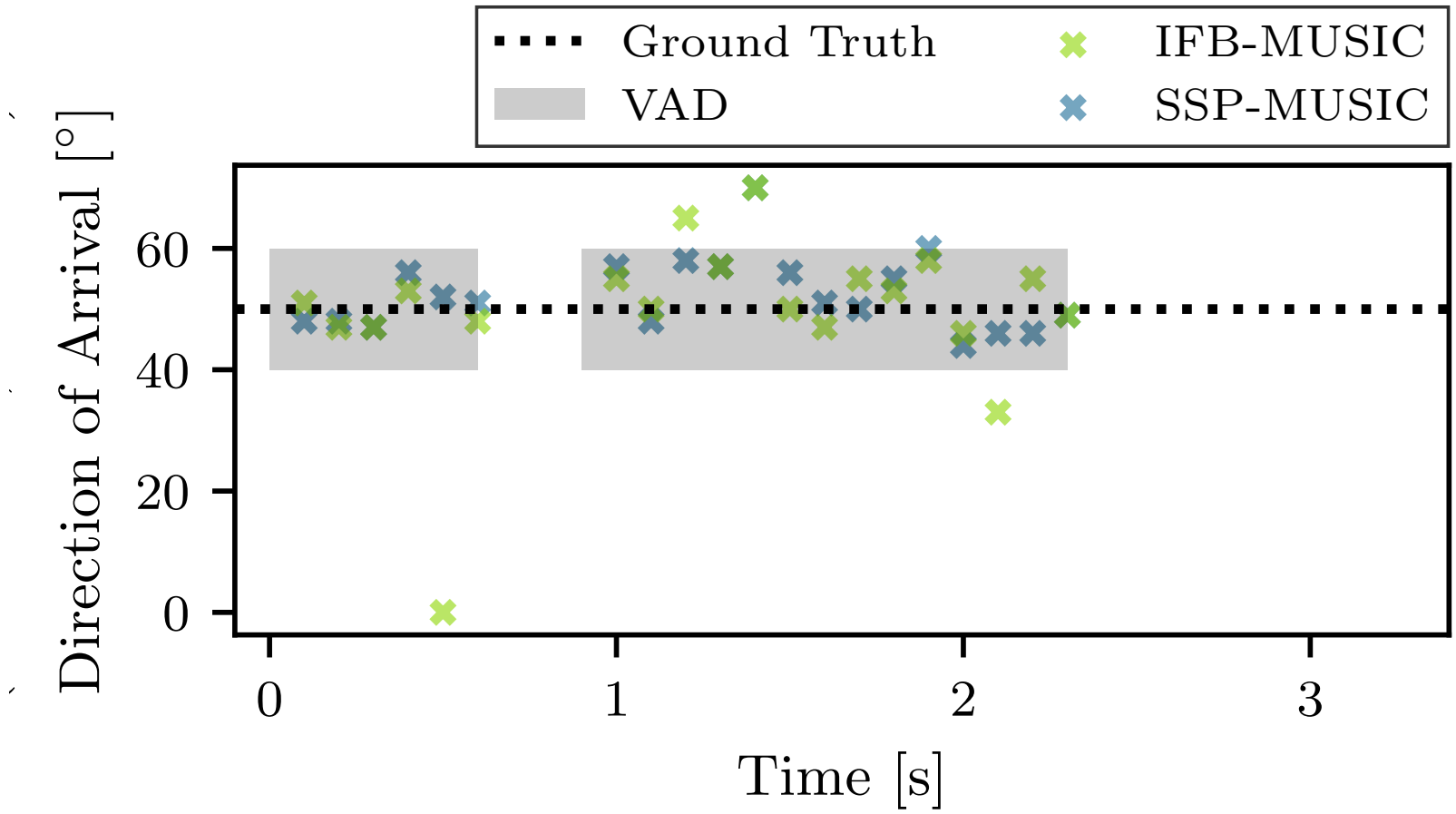
SNR: 15 dB, T60: 0.5 s



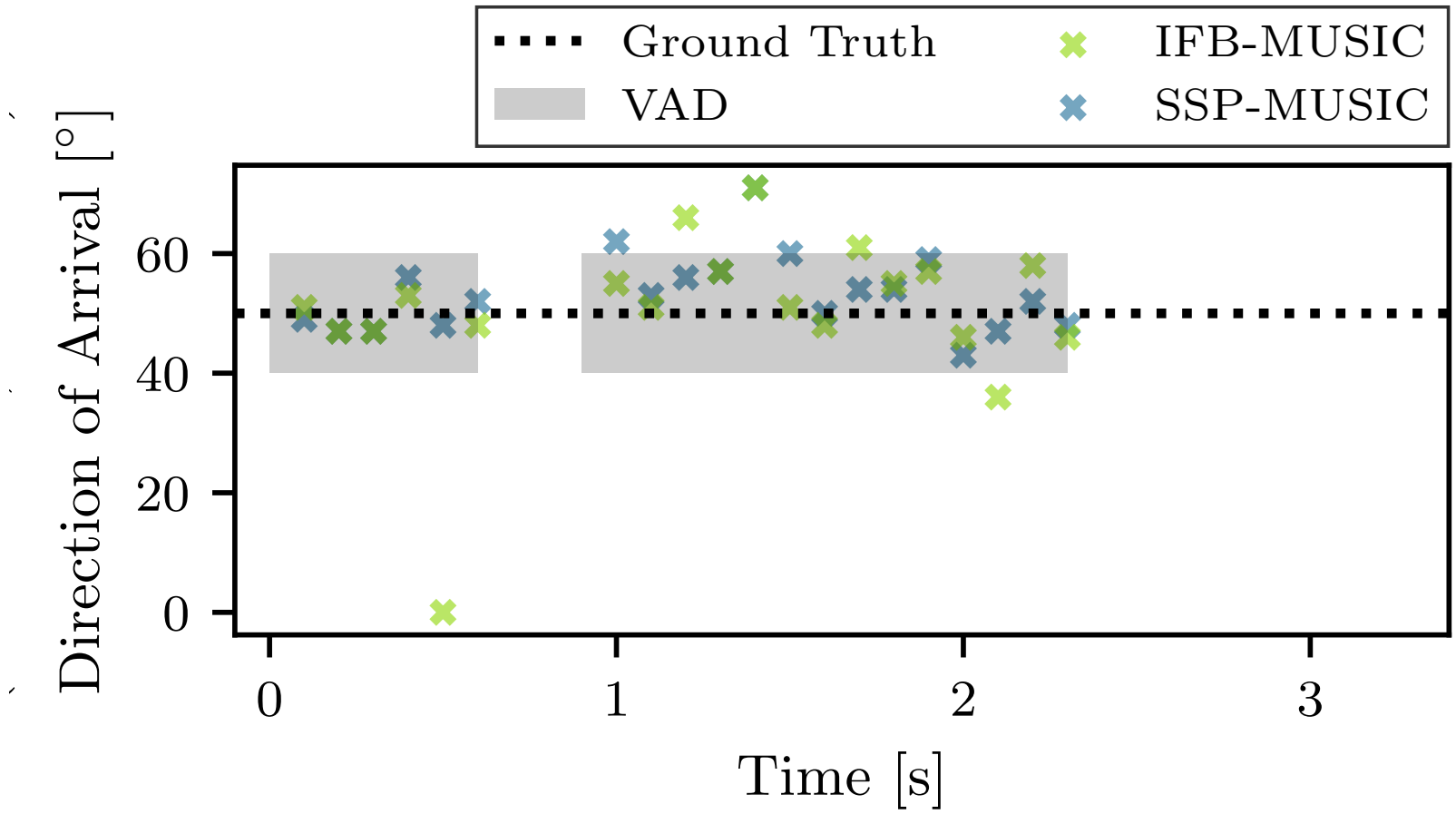
SNR: 15 dB, T60: 0.7 s



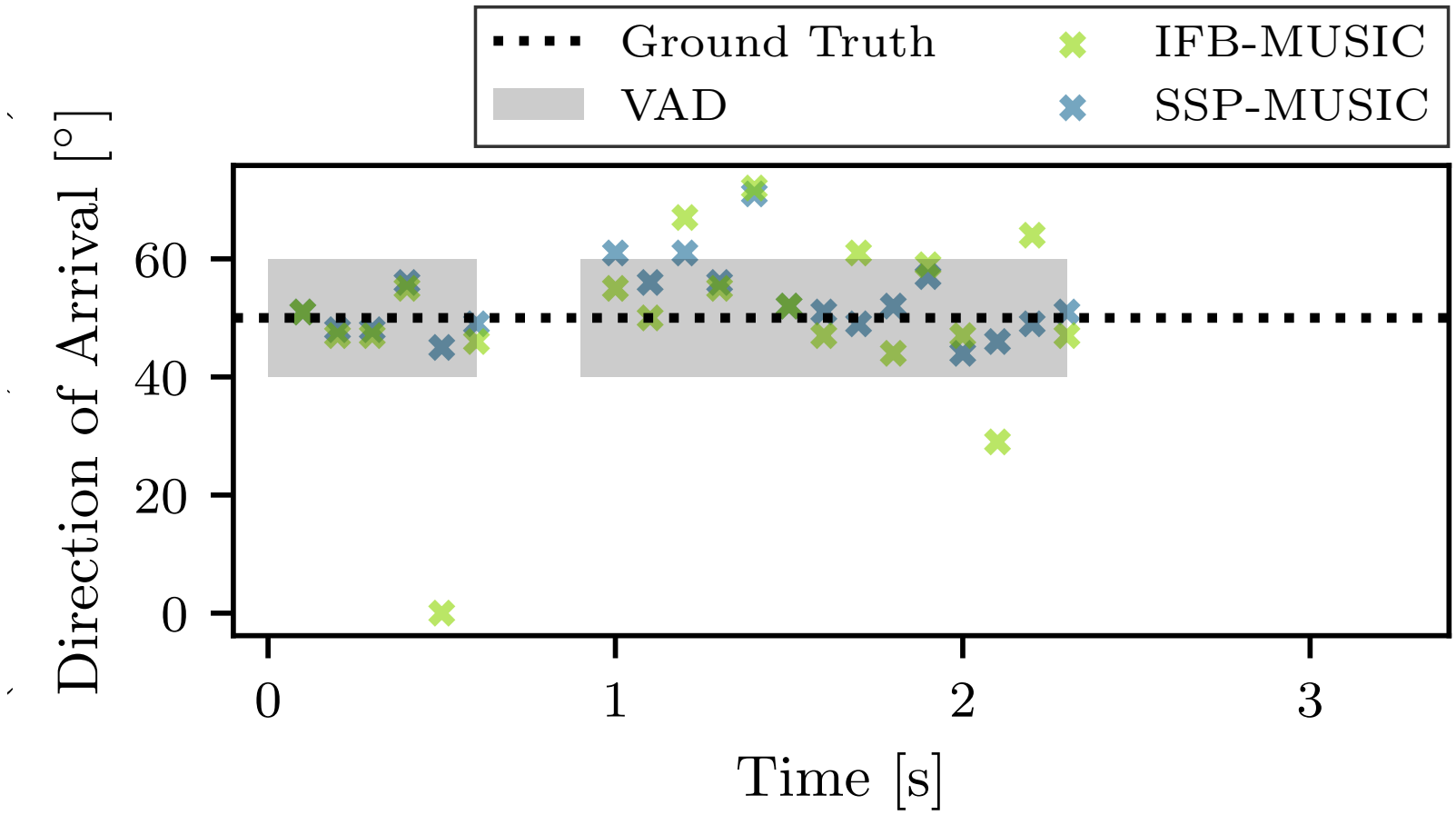
SNR: 15 dB, T60: 0.9 s



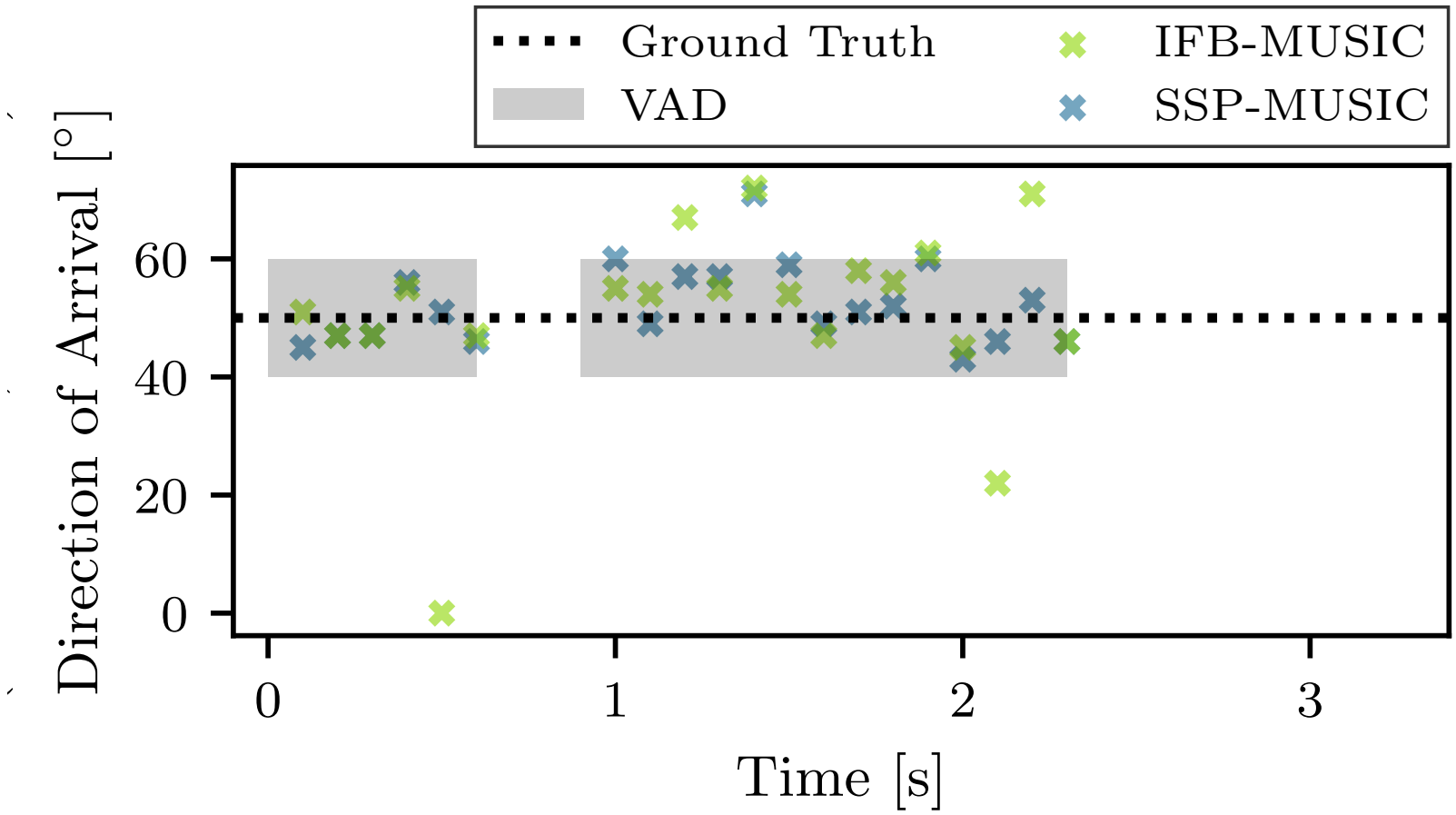
SNR: 15 dB, T60: 1.1 s



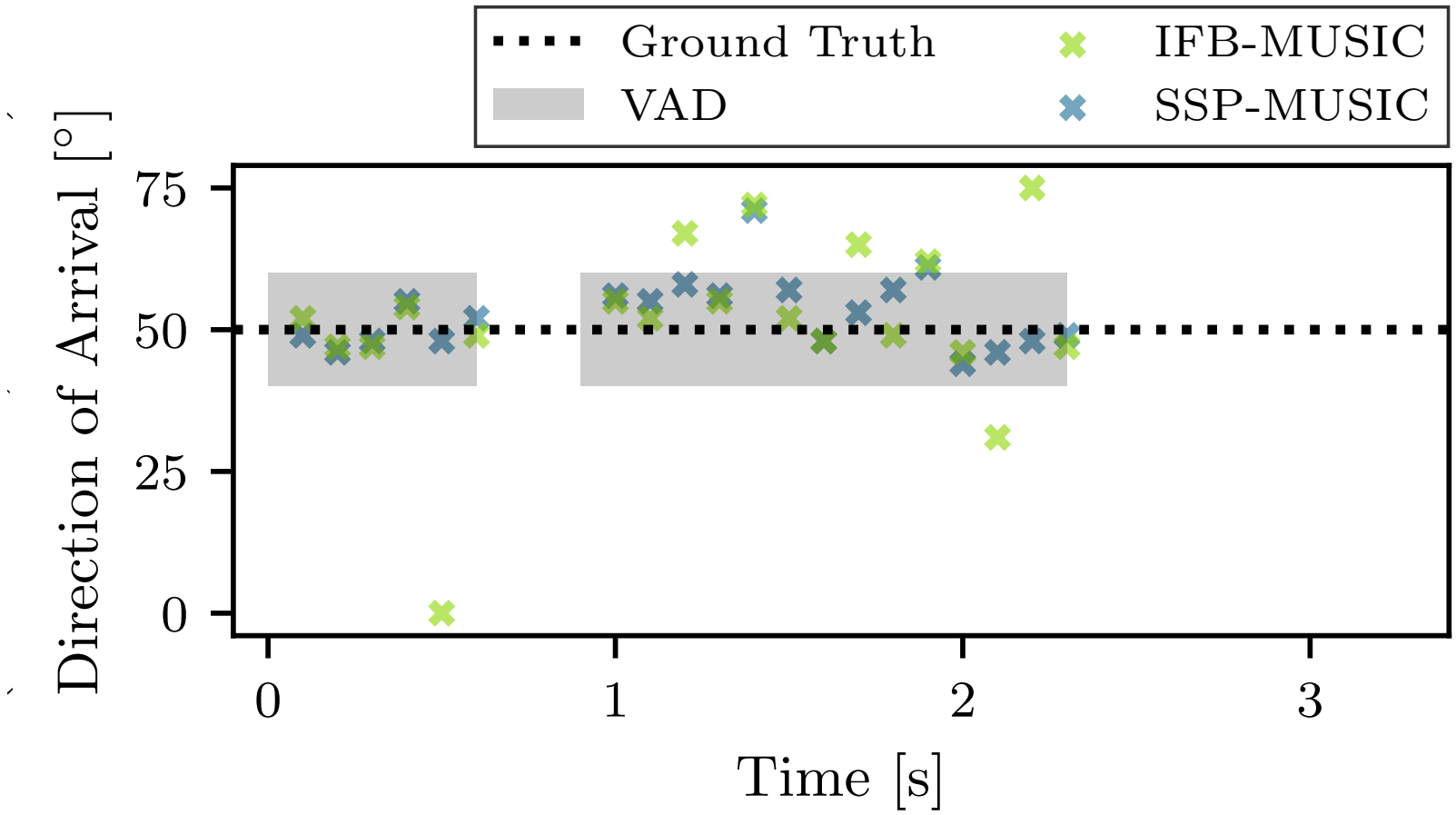
SNR: 15 dB, T60: 1.3 s



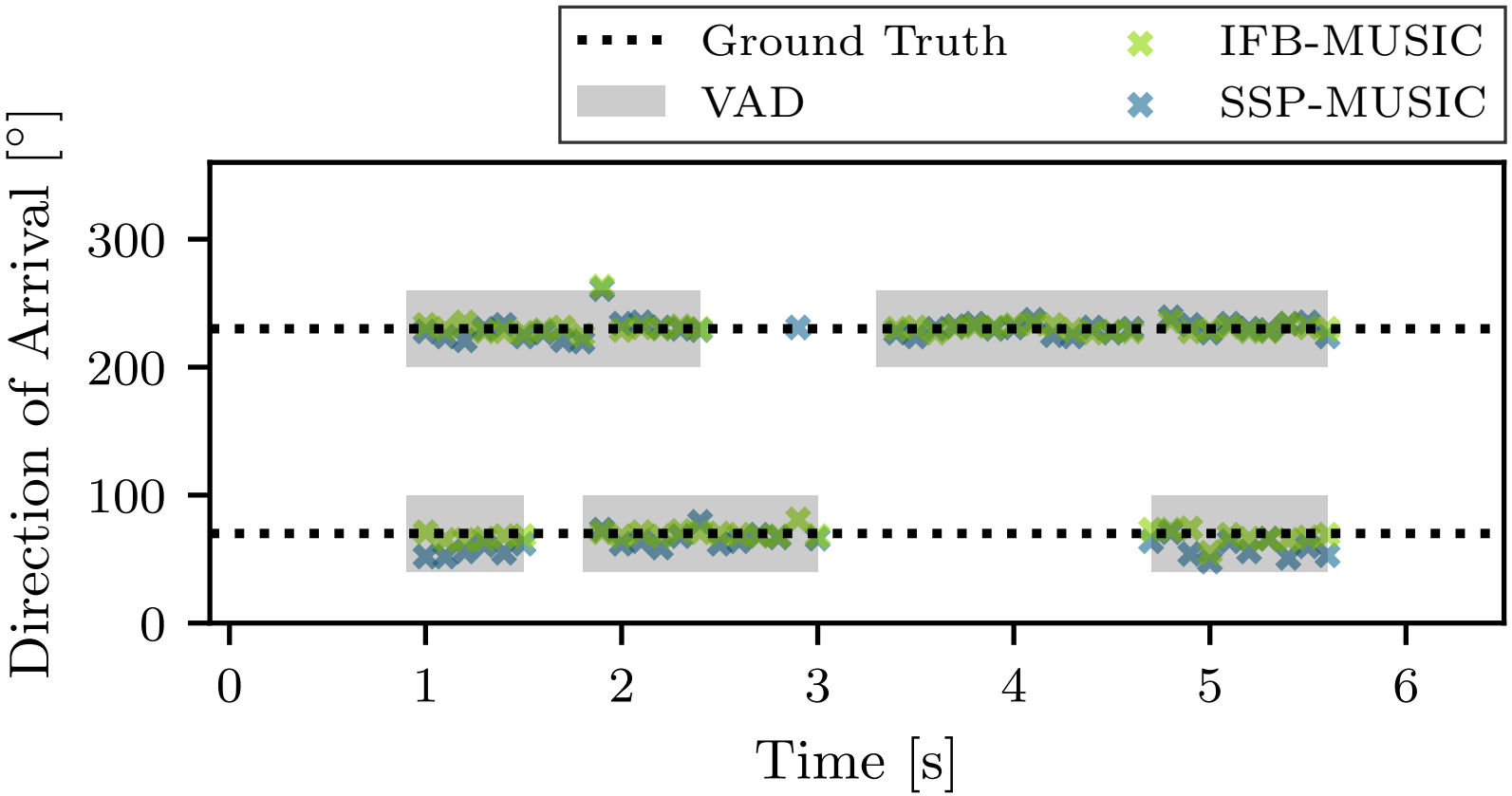
SNR: 15 dB, T60: 1.5 s



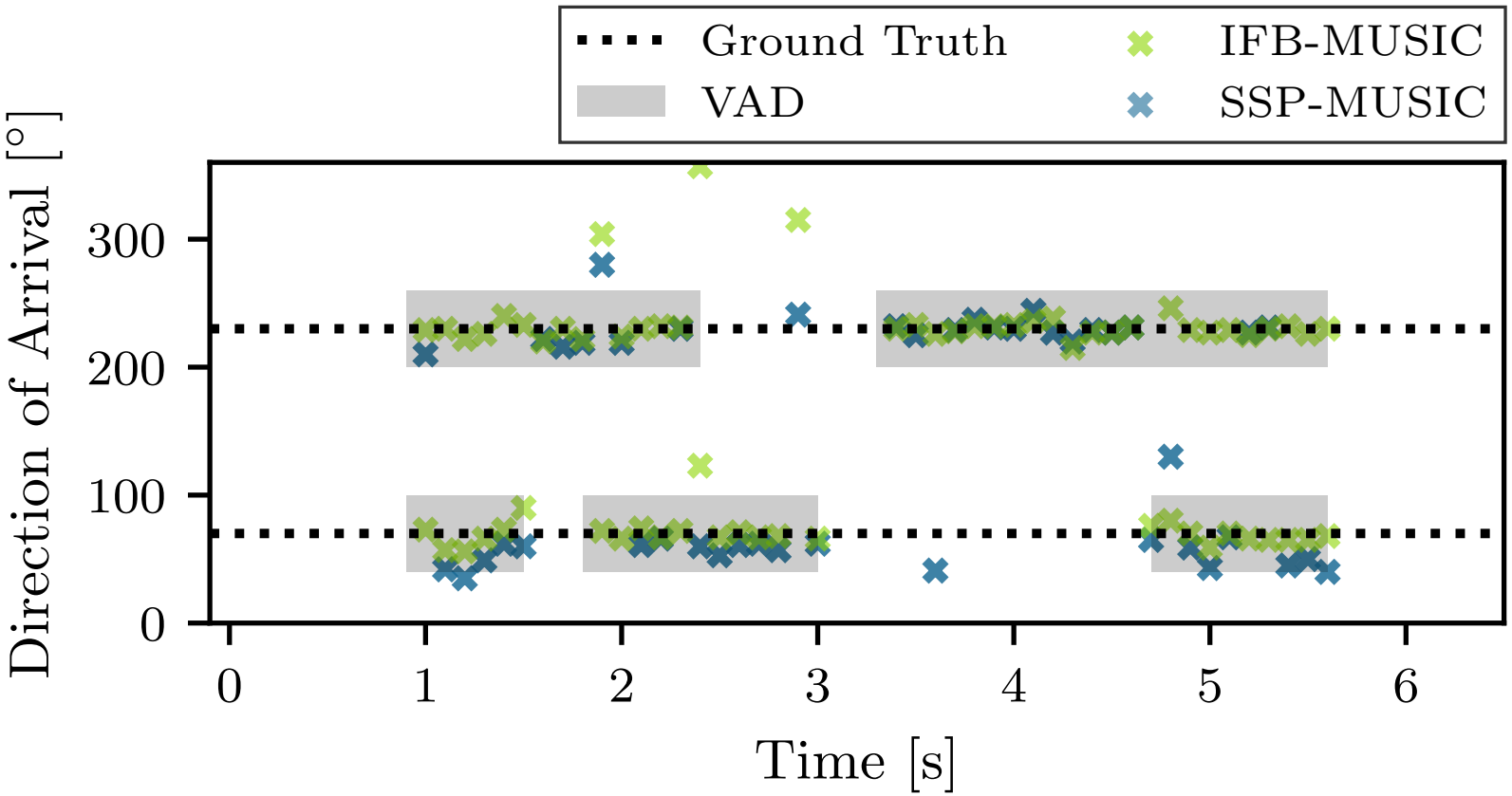
SNR: 15 dB, T60: 1.7 s



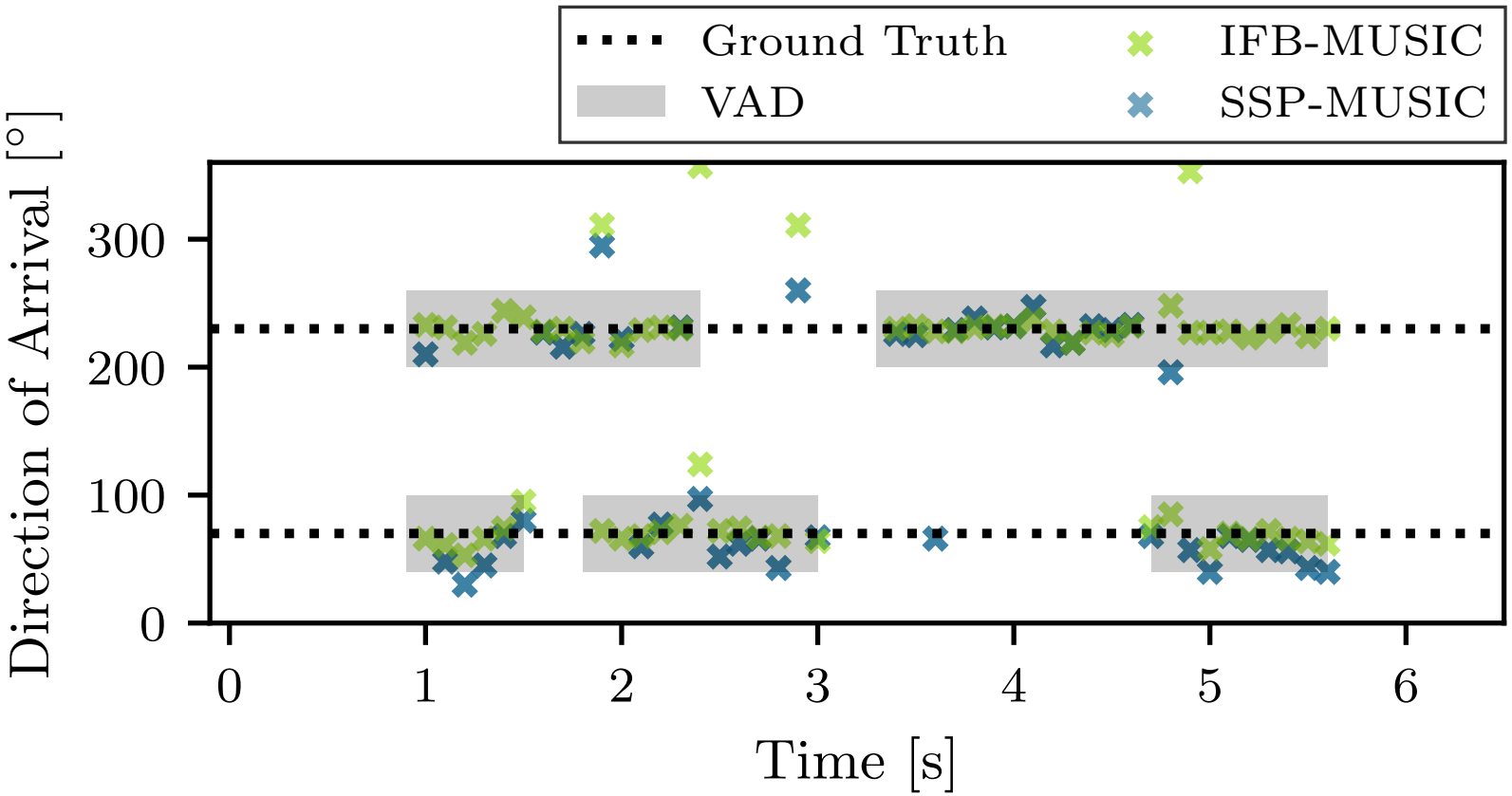
SNR: 15 dB, T60: 0.1 s



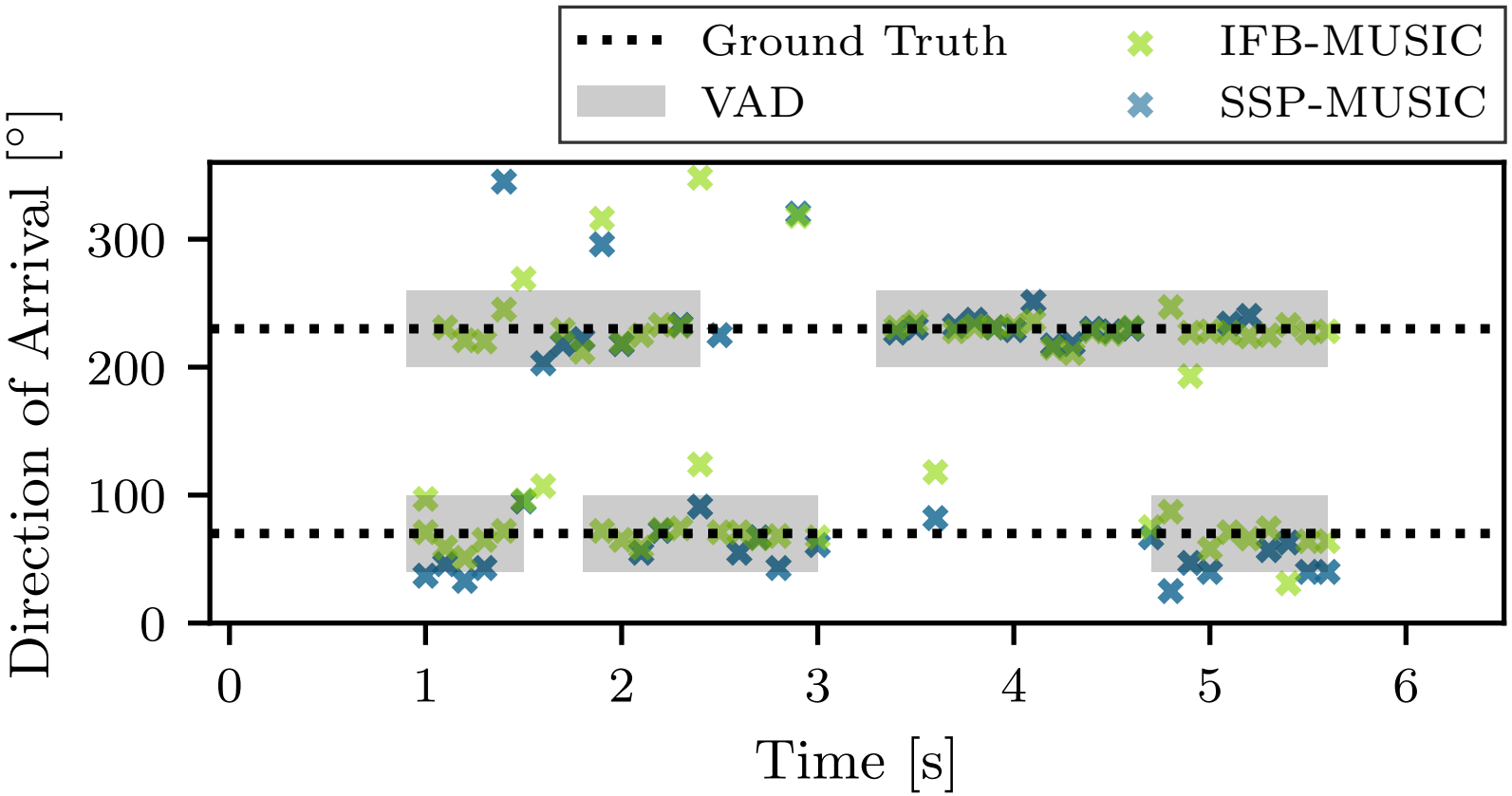
SNR: 15 dB, T60: 0.3 s



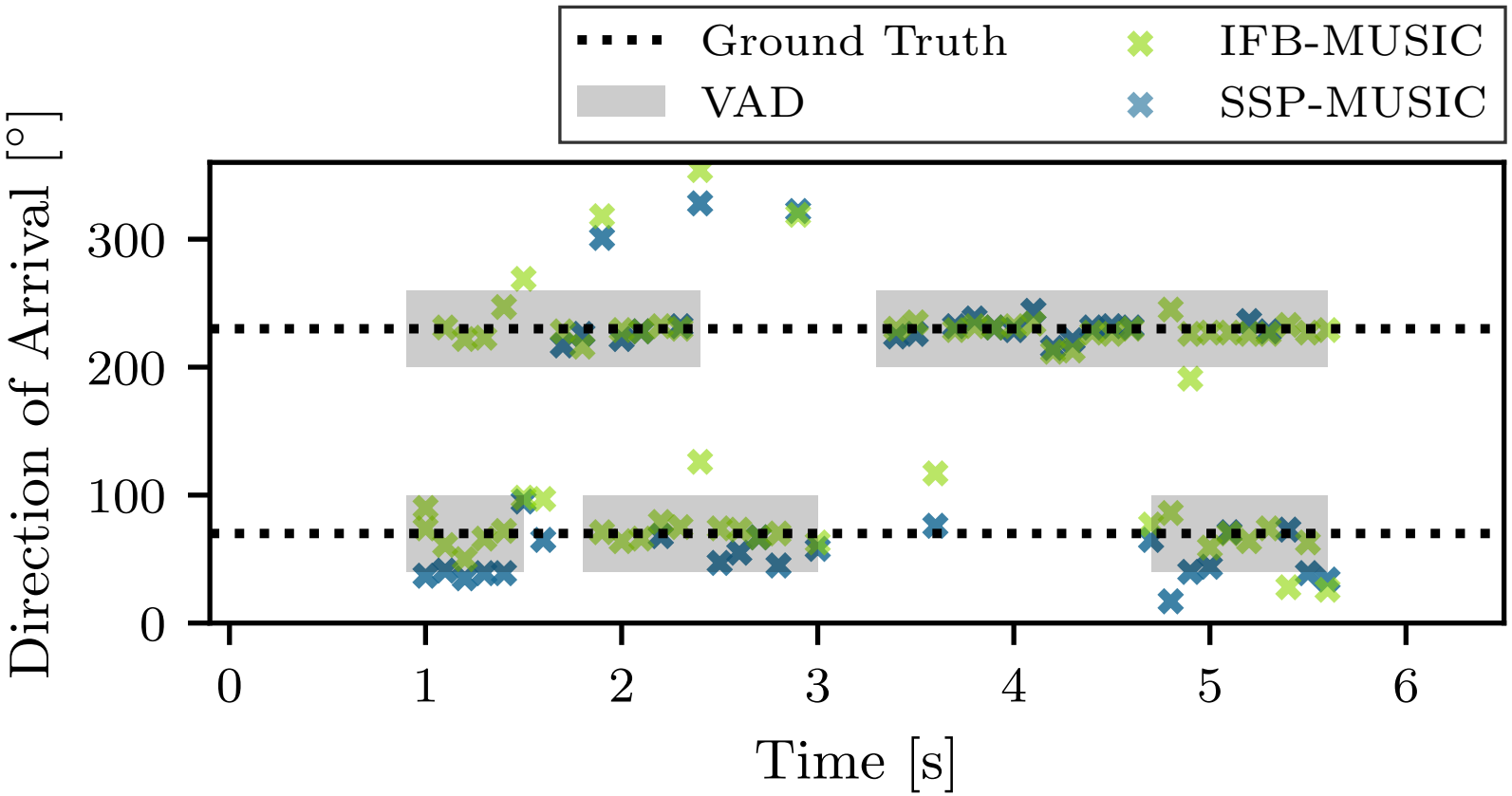
SNR: 15 dB, T60: 0.5 s



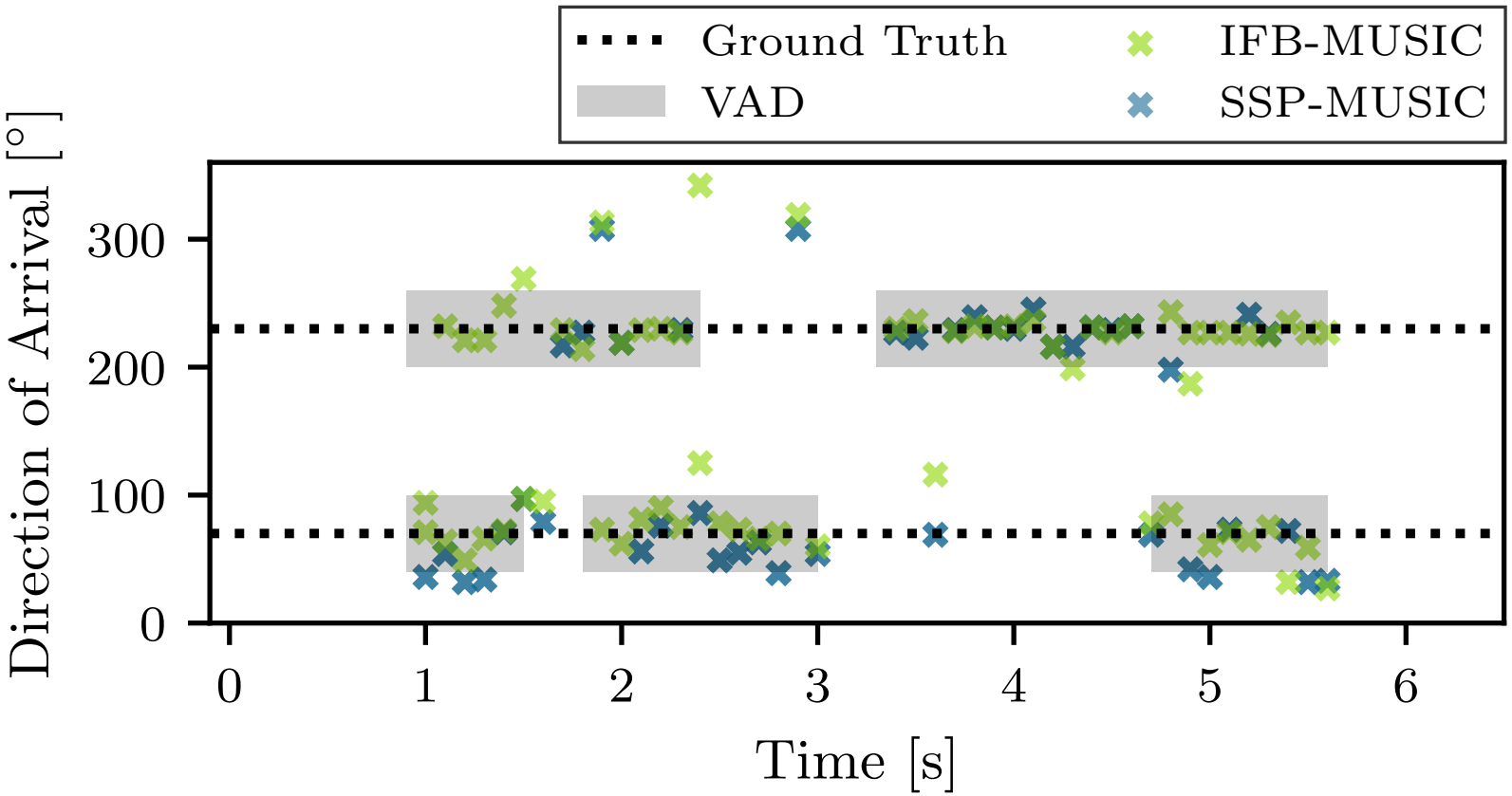
SNR: 15 dB, T60: 0.7 s



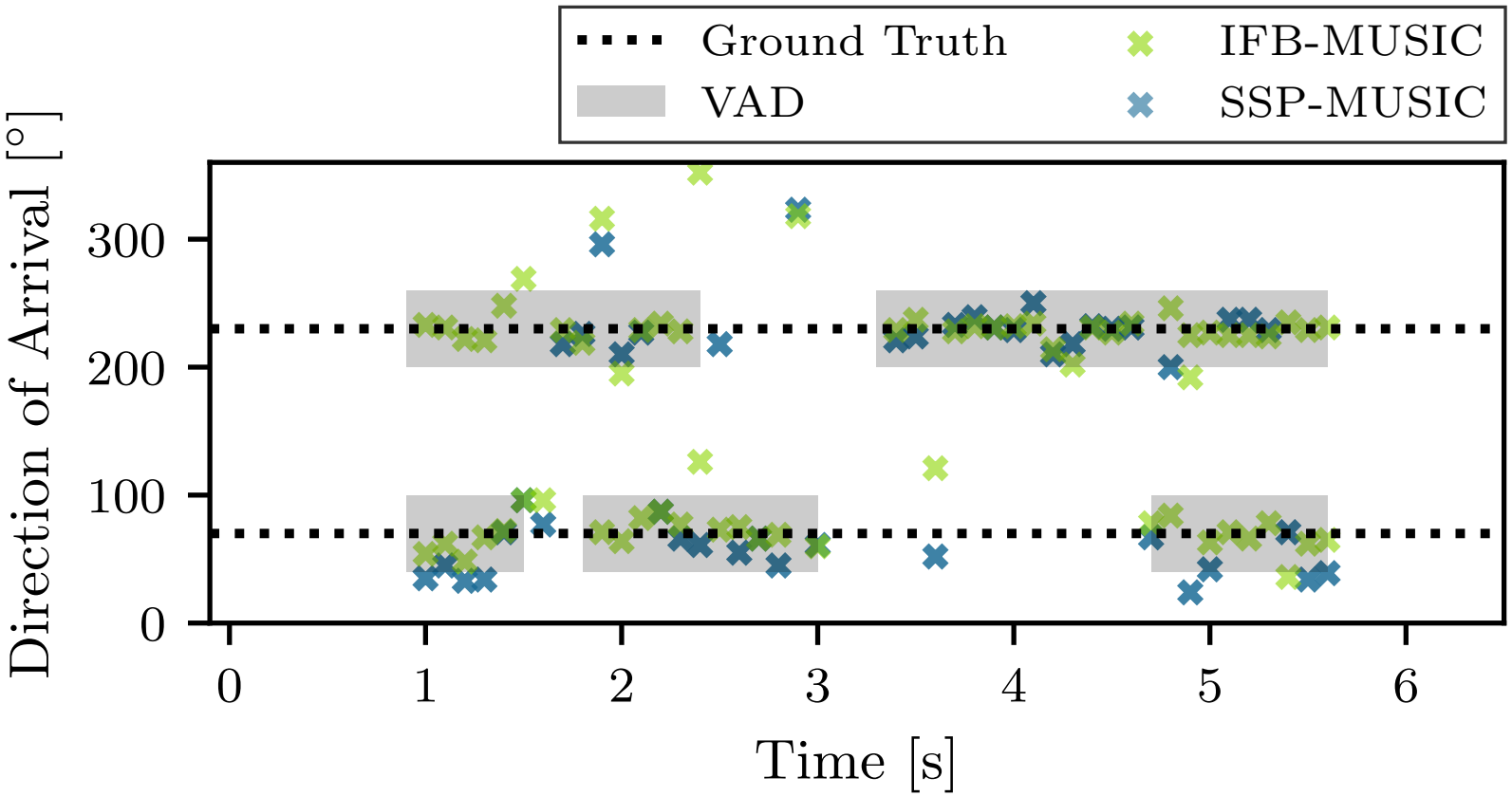
SNR: 15 dB, T60: 0.9 s



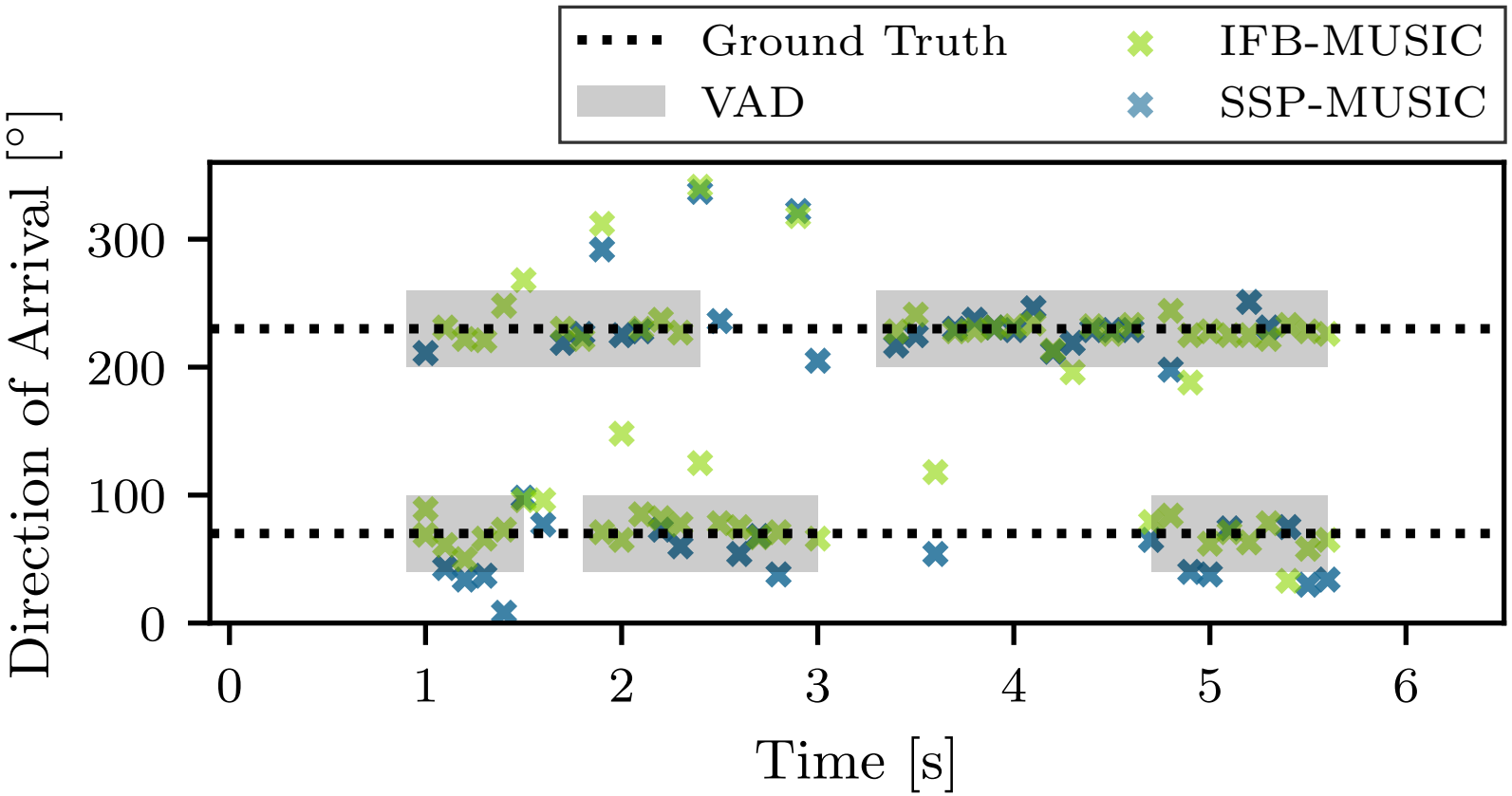
SNR: 15 dB, T60: 1.1 s



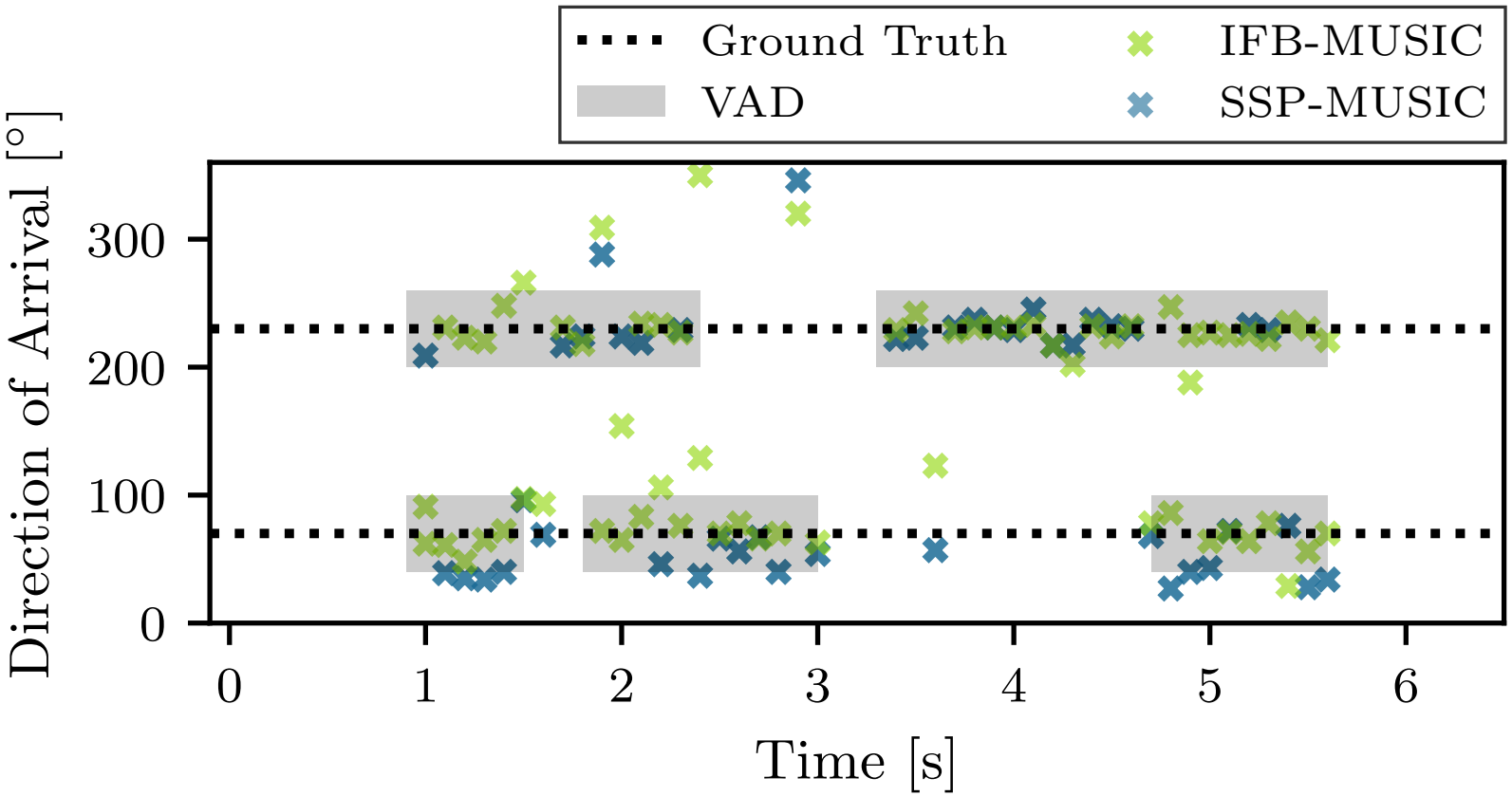
SNR: 15 dB, T60: 1.3 s



SNR: 15 dB, T60: 1.5 s

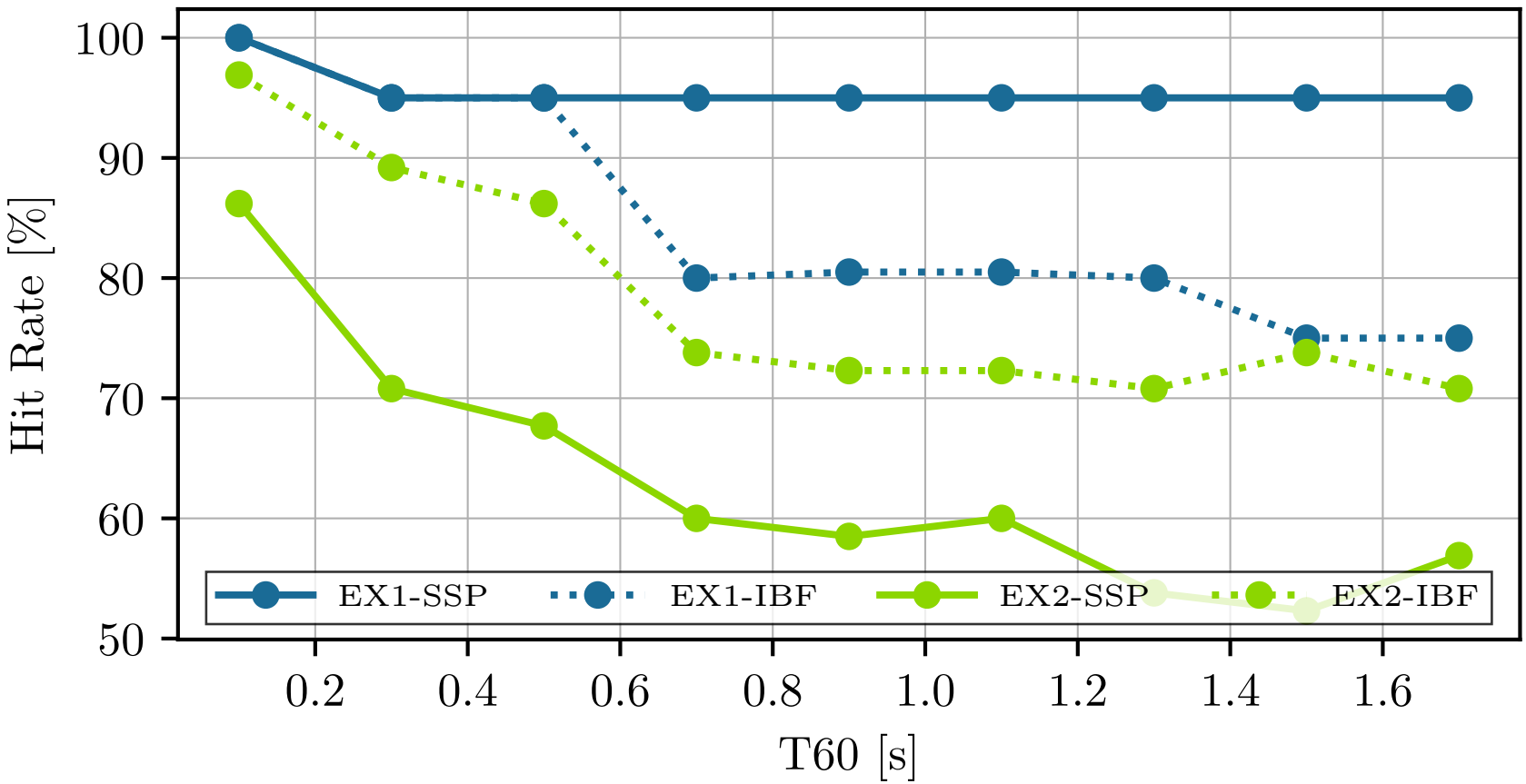


SNR: 15 dB, T60: 1.7 s



EX1: SINGLE SOUND SOURCE IN A SIMULATED ROOM

EX2: TWO SOUND SOURCES IN A SIMULATED ROOM



Method		SSP-MUSIC		IFB-MUSIC	
Metric		HR	FAR	HR	FAR
Recording	1	90.0	10.0	95.0	5.0
	2	64.7	35.3	67.6	32.4
	3	95.1	4.9	75.6	24.4

Comparison of SSP-MUSIC against IFB-MUSIC on the first 3 LOCATA recordings for Task 1.

CONCLUSION

In this talk, we have...

... developed and explored the potential of SSP-MUSIC, which is a polynomial extension of MUSIC.

... proposed some enhancements for sound source localization using SSP-MUSIC.

... demonstrated that SSP-MUSIC is more robust to noise and reverberation.

Questions?

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