Challenges in microphone array processing for hearing aids

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Why directional microphones in hearing aids?

Sensorineural hearing loss

Reduced time and frequency resolution leads to a decreased speech intelligibility in noisy environment: SNR-loss typically 4-10 dB (Dillon, 2001). Up to now, no direct compensation method exists => **Increase SNR !**

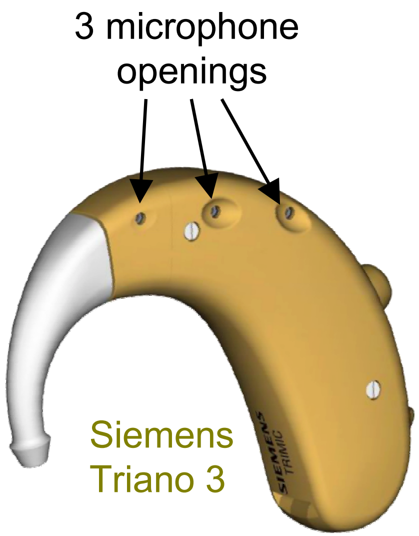
Use of indirect methods

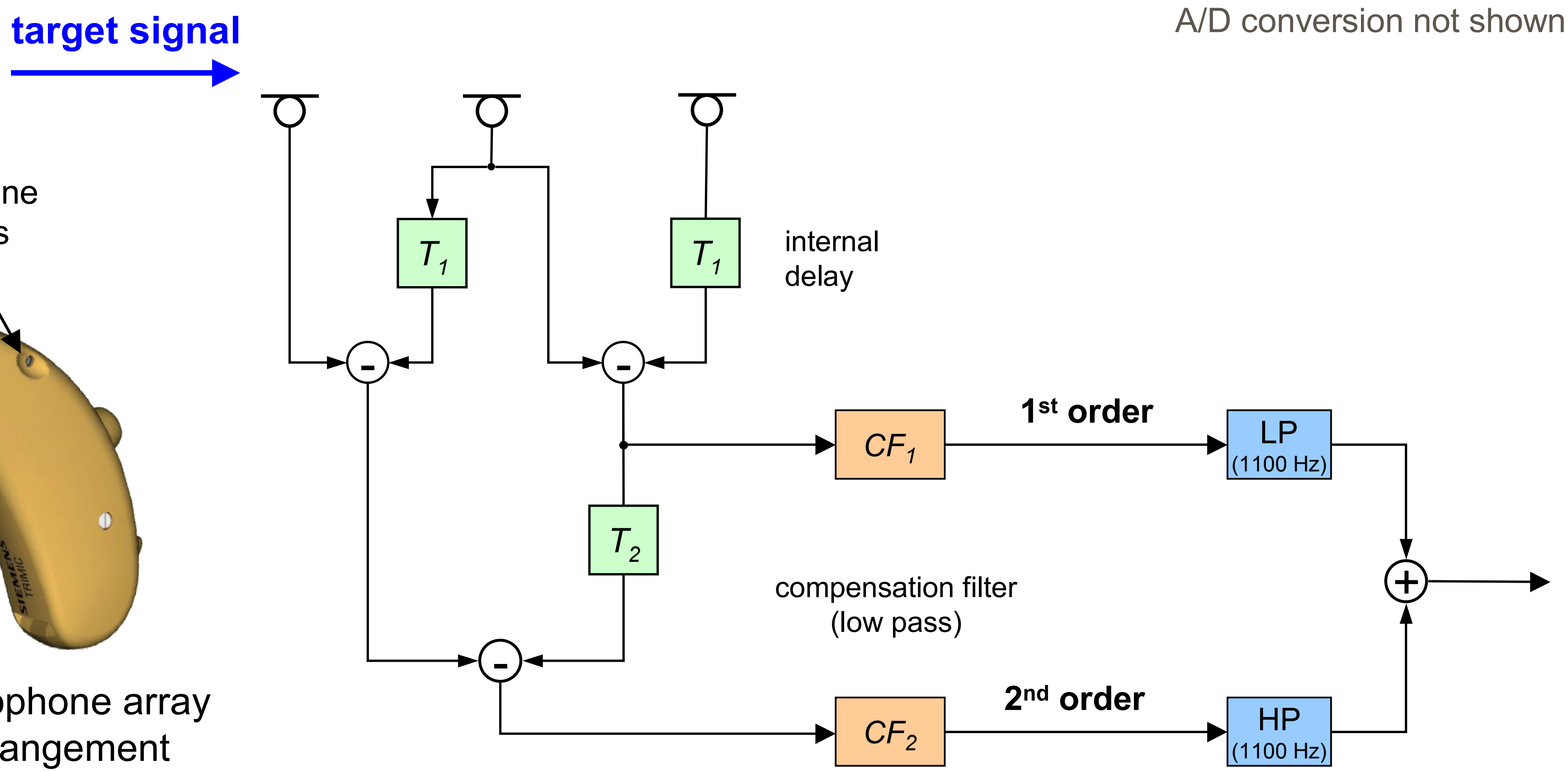
Single-microphone noise reduction algorithms have not been shown to significantly improve speech intelligibility!

Directional microphones implemented in hearing aids can improve SNR in a way that leads to improved speech intelligibility (e.g. Valente, 1995).

Three-microphone array

combination of 1st and 2nd order differential processing *(TriMic™)*
AI-DI (KEMAR) approx. 6.2 dB





- three-microphone array
- end-fire arrangement
- 8 mm microphone spacing

Performance Challenge

SNR-Effect

Dir.-Mic. vs. Omni	+4.0 dB
Omni vs. Open Ear	appox. -0.5 dB
Net improvement	**3.5 dB**

In many cases todays directional microphones do not provide enough SNR benefit to compensate for the 4-10 dB SNR-loss caused by the hearing impairment.

=> Improve SNR by advanced adaptive microphone array processing algorithms

Robustness Challenge

Additional performance loss in real-life use

- microphone mismatch
- array misalignment (e.g. non-0° or moving targets)
- earmold venting

Improve robustness by

- adaptive microphone matching (amplitude and phase)
- beamforming design procedures with robustness against microphone mismatch (e.g. Doclo, 2003)
- adaptive target source tracking algorithms
- active noise control
- ...

Advanced multi-microphone noise reduction algorithms

Approaches known from the literature

- Beamforming for microphone arrays taking head shading effects into account (e.g. Meyer 2001)
- Adaptive beamforming (e.g. Greenberg et al. 1992, Kompis & Dillier 1993, 2001, Herbordt et al. 2002)
- Blind source separation (e.g. Parra 2000, Anemueller 2001)
- Coherence based filtering (e.g. Allen et al. 1977)
- Binaural Spectral Subtraction (Doerbecker 1996)
- “Cocktail Party” processors (e.g. Peissig 1992, Bodden 1992, Feng et al. 2001)
- ...

Parameters defining the acoustic scene

Each algorithm is designed for <u>specific acoustic scenes</u>, described by:

Spatial parameters:

- target source position
- number of noise sources
- position of noise sources
- reverberation
- dynamics (temporal changes)
- ...

Sound source characteristics:

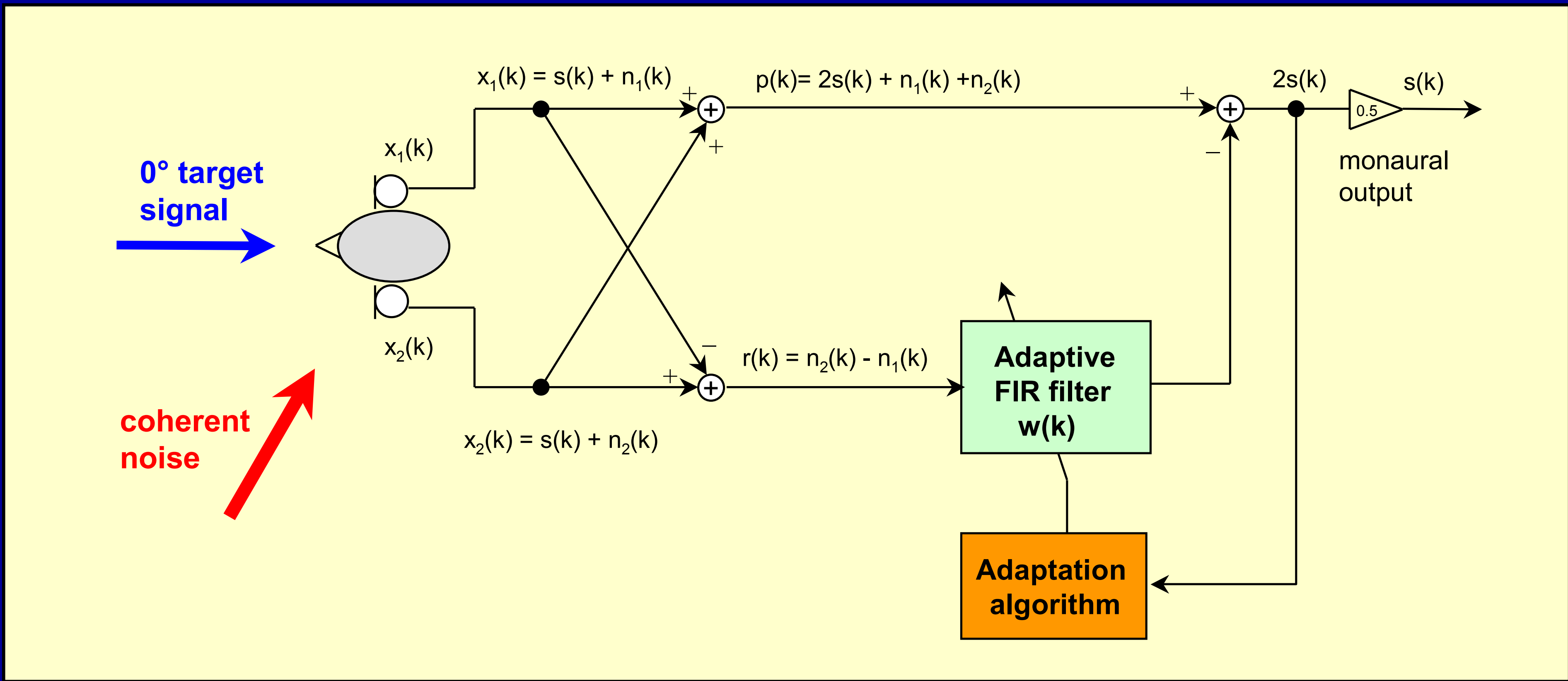
- type (speech, music, etc)
- level
- spectrum
- stationarity
- ...

☹ However, in situations other than those specified many algorithms become ineffective or even make the SNR worse!

Adaptive Beamformer for binaural hearing aids
(Greenberg et al. 1992, Dillier & Kompis 1993)

Assumptions:

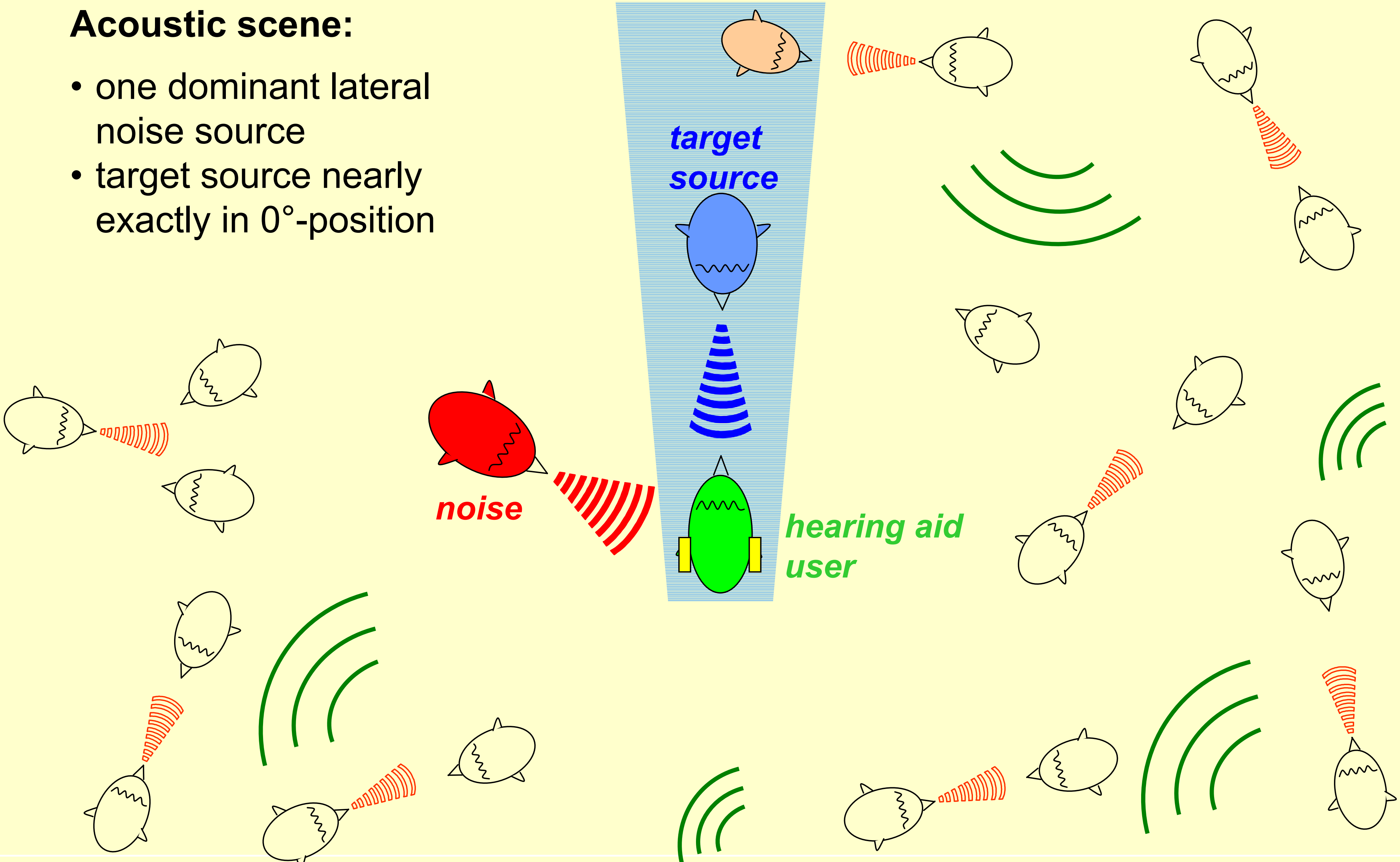
- signal source exactly in 0°-position
- coherent noise incidence, i.e. one dominant lateral noise source



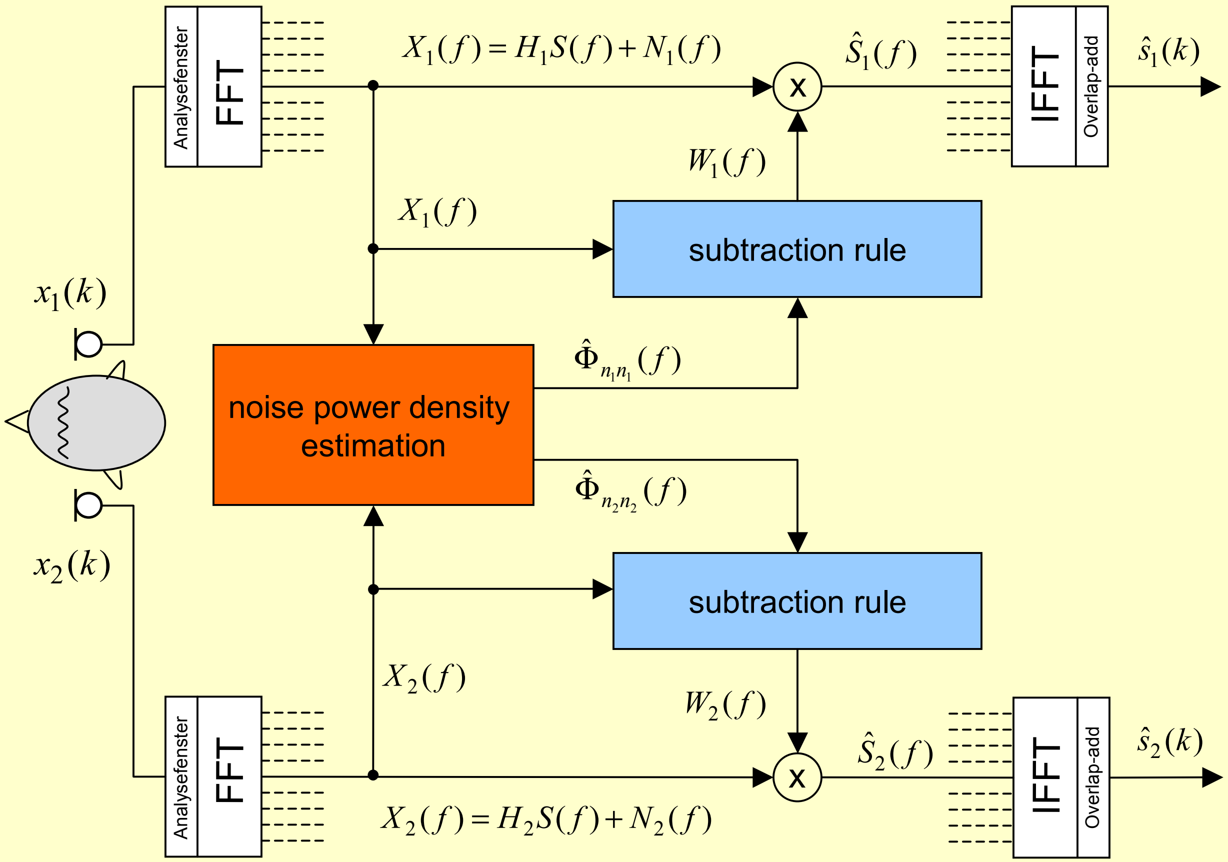
Adaptive Beamformer (cont'd)

Acoustic scene:

- one dominant lateral noise source
- target source nearly exactly in 0°-position



Binaural Spectral Subtraction (Doerbecker et al., 1996)



Assumptions

- coherent target signal, e.g. near speaker
- diffuse noise field

Noise power density estimation

based on correlation analysis

$$X_1(f) = H_1(f)S(f) + N_1(f)$$
$$X_2(f) = H_2(f)S(f) + N_2(f)$$

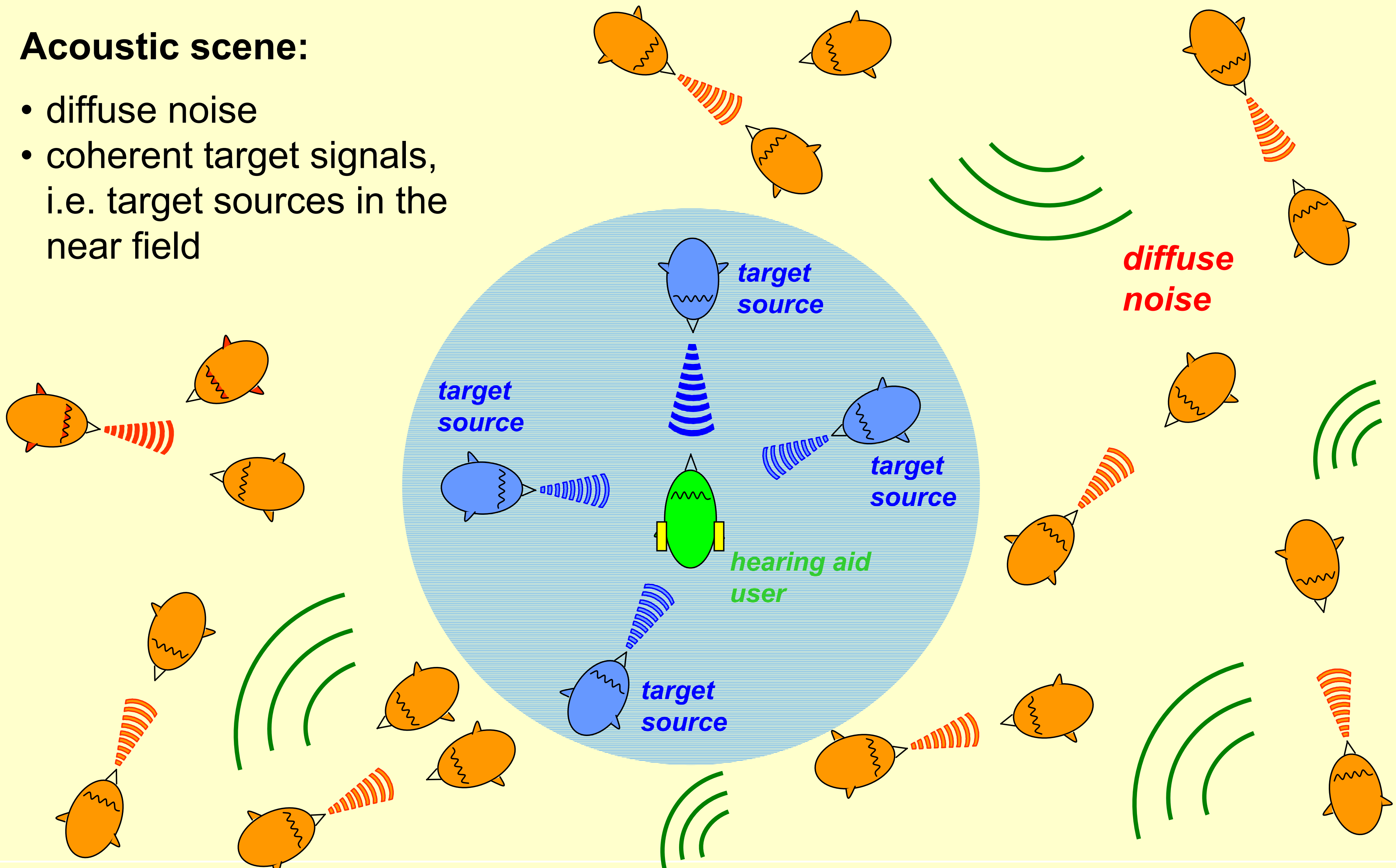
with $|H_1(f)| = |H_2(f)|$ the noise power density can be estimated:

$$\Phi_{n_1 n_1}(f) = \Phi_{x_1 x_1}(f) - \left|\Phi_{x_1 x_2}(f)\right|$$
$$\Phi_{n_2 n_2}(f) = \Phi_{x_2 x_2}(f) - \left|\Phi_{x_1 x_2}(f)\right|$$

Binaural Spectral Subtraction (cont'd)

Acoustic scene:

- diffuse noise
- coherent target signals, i.e. target sources in the near field



Control problem

Prospects for the "array-processing future":

Hearing aids will offer a set of highly specialized multi-microphone noise reduction algorithms designed for specific acoustic scenes.

Problem: probably the user will not be willing or be able to a) monitor the acoustic environment continuously, b) recognize the specific acoustic scenes, and c) activate the best algorithm.

Manual control of directional microphones is already nowadays a severe problem for many hearing aid user (e.g. Kuk 1996, Cord et al. 2002).

=> Need of intelligent control systems in hearing aids based on continuous classification of the acoustic scene and automatic activation of the appropriate algorithm.

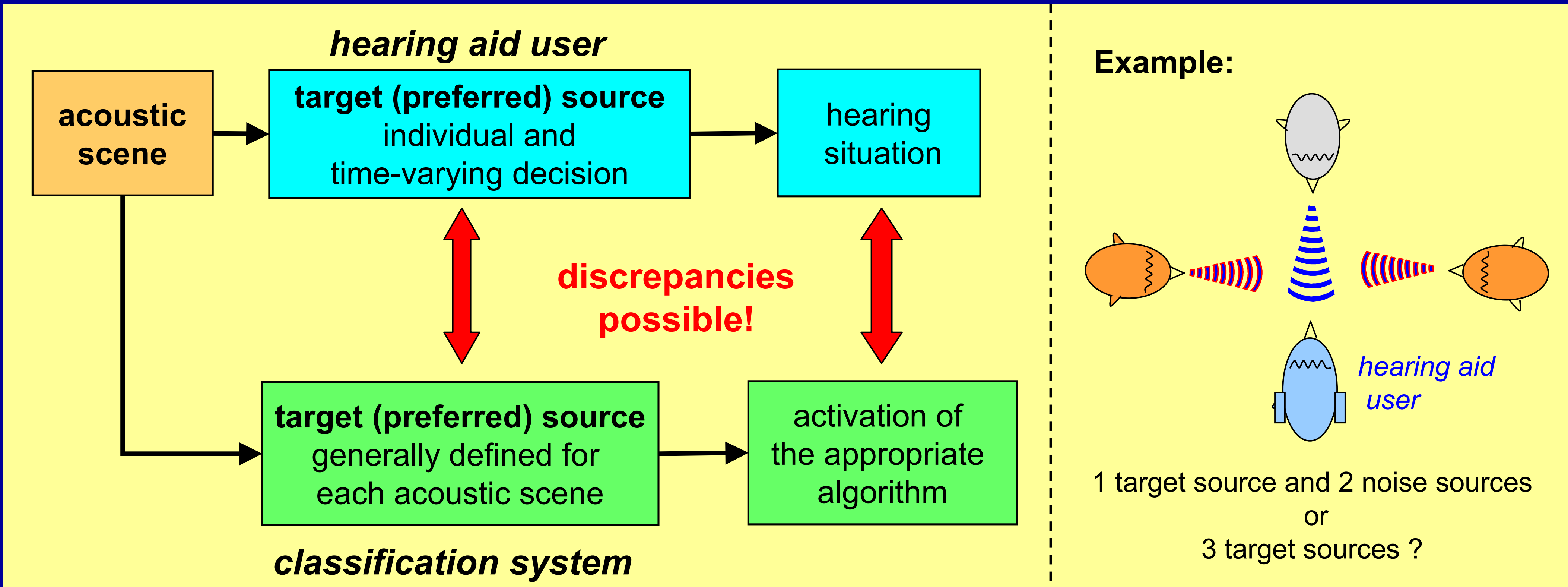
Control challenge

Todays classification & control systems rely only on statistical information derived from one microphone signal. Spatial parameters are not taken into account!

=> Appropriate classification & control systems must be developed, which determine the acoustic scene based on statistical and spatial information extracted from microphone-array analysis.

Otherwise, the SNR-improvement offered by advanced adaptive microphone-array processing will probably not fully turn into customer benefit in real life.

Limitation of classification & control



As long as the „link to the brain“ is not available, manual control of the hearing aid operation occasionally will be necessary!

Conclusions

Microphone array processing is currently the only method to substantially improve the SNR.

To fully compensate for the SNR-loss in everyday-life three groups of challenges have to be met:

- *Robustness*
- *Performance (SNR)*
- *Automatic Control*

Advanced multi-microphone noise reduction algorithms differ in their assumptions concerning the acoustic scene => Need of automatic control systems.

Classification & Control will never match 100% with listener’s decisions.

=> Manual control occasionally is neccessary, when the actual hearing situation differs from the one estimated by the classification system.

Thank you for your attention!