

# Fundamentals of Sound Source Localization

University of Pretoria – August 2018



# Fundamentals of Sound Source Localization

## Agenda



- Introduction to Sound Source Localization
- Beamforming-based localization methods
- Far-field Deconvolution methods
- Map averaging methods

# Introduction to sound source localization

## Overview of current techniques

Simple

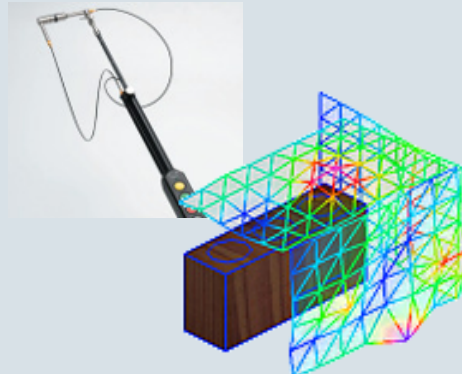
Advanced

### Stethoscope



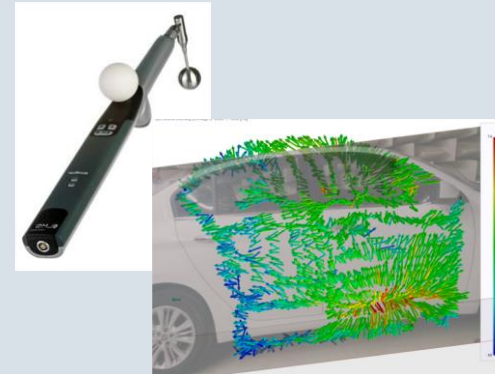
- Limited hardware
- No reportable data
- Sound pressure only
- Stationary sources only

### Sound Intensity



- ISO sound power & partial sound power
- Time-consuming
- Limited frequency range
- Stationary sources only

### SoundBrush



- Fast results
- Limited hardware
- Directivity & sound power estimation
- Stationary sources only

### Sound Camera

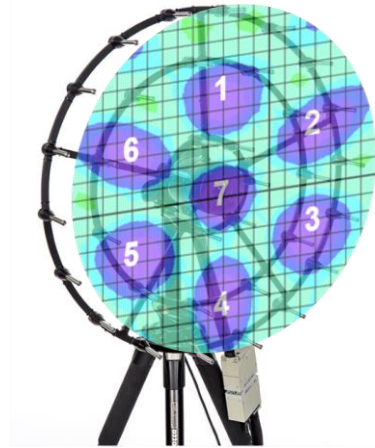


- Real-time results
- Stationary and transient sources
- Requires more advanced knowledge to operate

# Introduction to sound source localization

## Why don't we just measure sound pressure?

**Measured**  
pressure field  
(7 sources?)



**Physical propagation**

**Back-propagation**



**Source**  
pressure field  
(3 sources)

Sound pressure waves are deformed during physical propagation towards the array (interference effects, reflections, absorption, etc.)

Raw measured pressure maps do not show correct localization of sound sources

**Need for specialized back-propagation techniques**

Wide variety of techniques available today:  
Beamforming, Holography, Deconvolution, etc.

No fit-for-all solution, every technique has its own boundary conditions, strengths and weaknesses

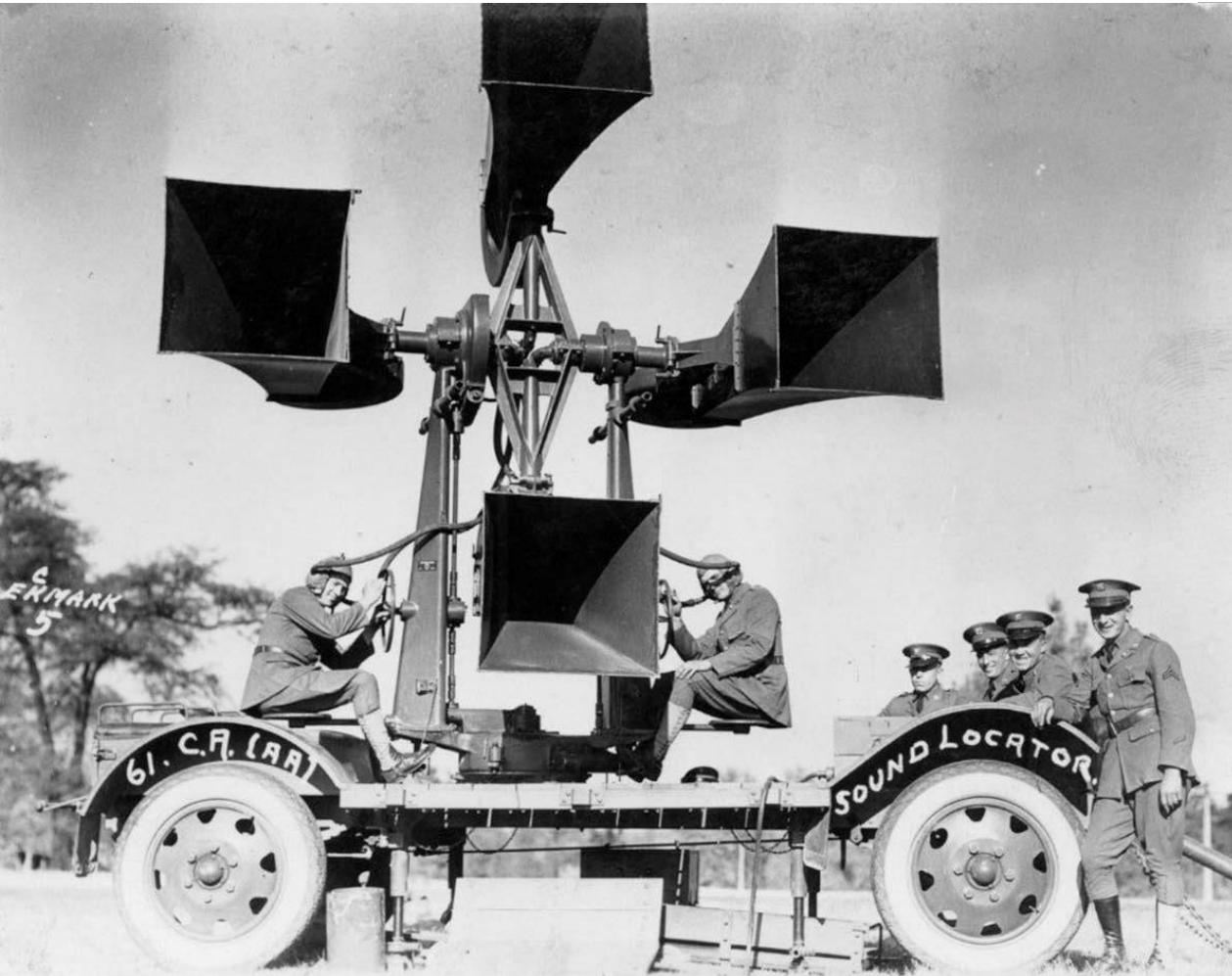
**Microphone arrays are an expert tool**



# Beamforming

A new method?

**SIEMENS**  
*Ingenuity for life*



# Fundamentals of Sound Source Localization

## Agenda

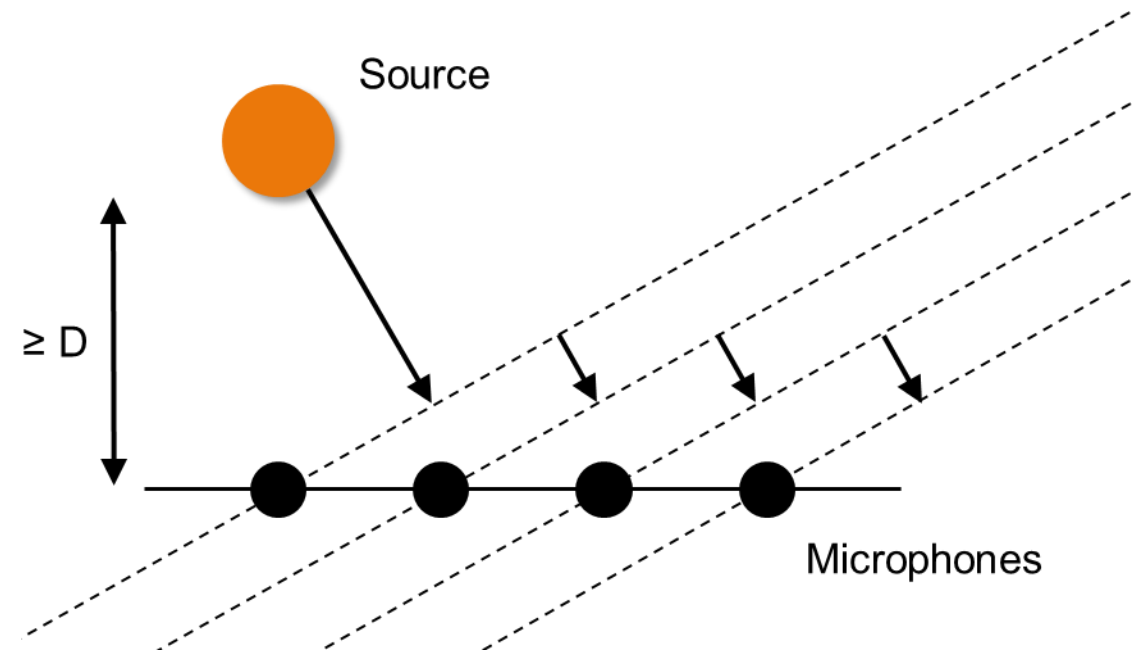


- Introduction to Sound Source Localization
- **Beamforming-based localization methods**
- Far-field Deconvolution methods
- Map averaging methods

# Beamforming

## How does it work?

- Assume source emits **planar** pressure waves
- Assumption requires **far-field** measurement conditions  
→ Distance to source  $\geq$  array diameter  $D$
- Microphone array consists of many microphones which record the sound pressure signal simultaneously
- Spatial distribution of sources causes a small time delay between the measured signals



**Beamforming core principle:**  
Sound source origin can be extracted from time delay  
(or phase delay) information between microphones

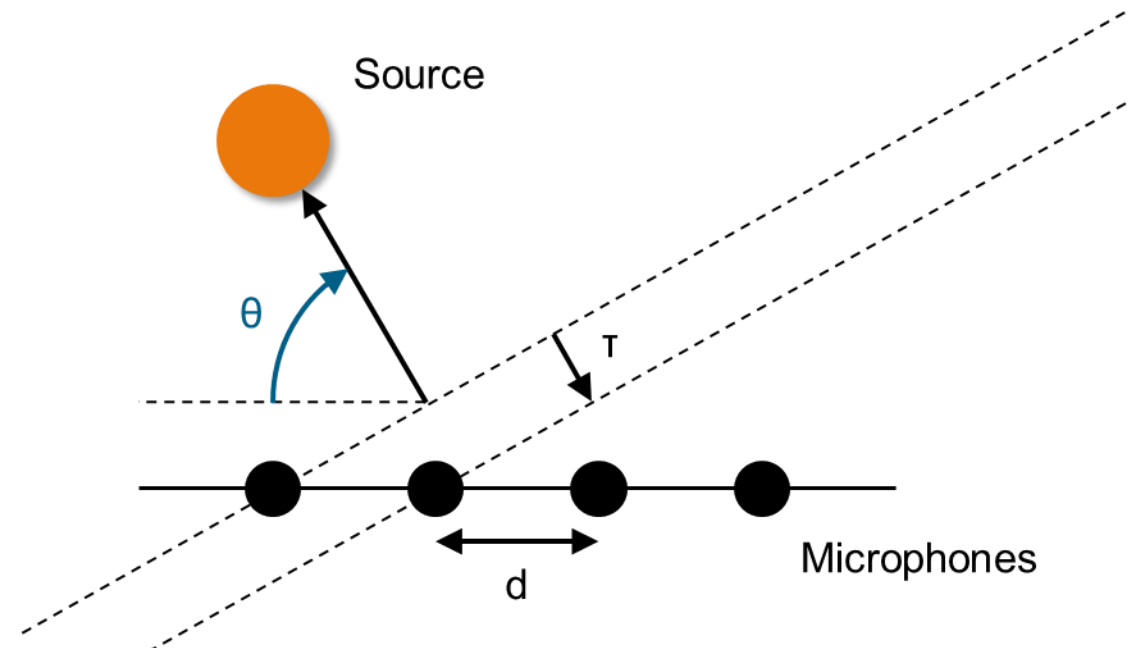
# Beamforming

## How does it work?

- Time delay  $\tau$  between microphone signals depends on:
  - Distance between microphones  $d$
  - Direction of the source vs. the microphone array  $\theta$
  - Speed of sound in ambient conditions  $c$

$$\tau = \frac{d \cdot \cos(\theta)}{c}$$

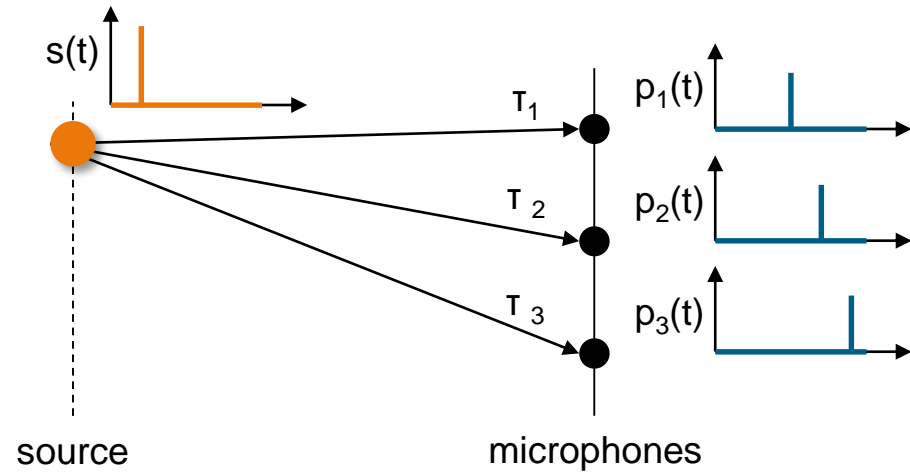
- Beamforming is a 'delay-and-sum' method:
  - Apply reverse time delay for each potential source
  - Sum microphone signals
- Reverse time delay for true source directions results in in-phase summation of microphone signals
- Summation amplitude ~ localization probability**





# Beamforming

## Time-domain example

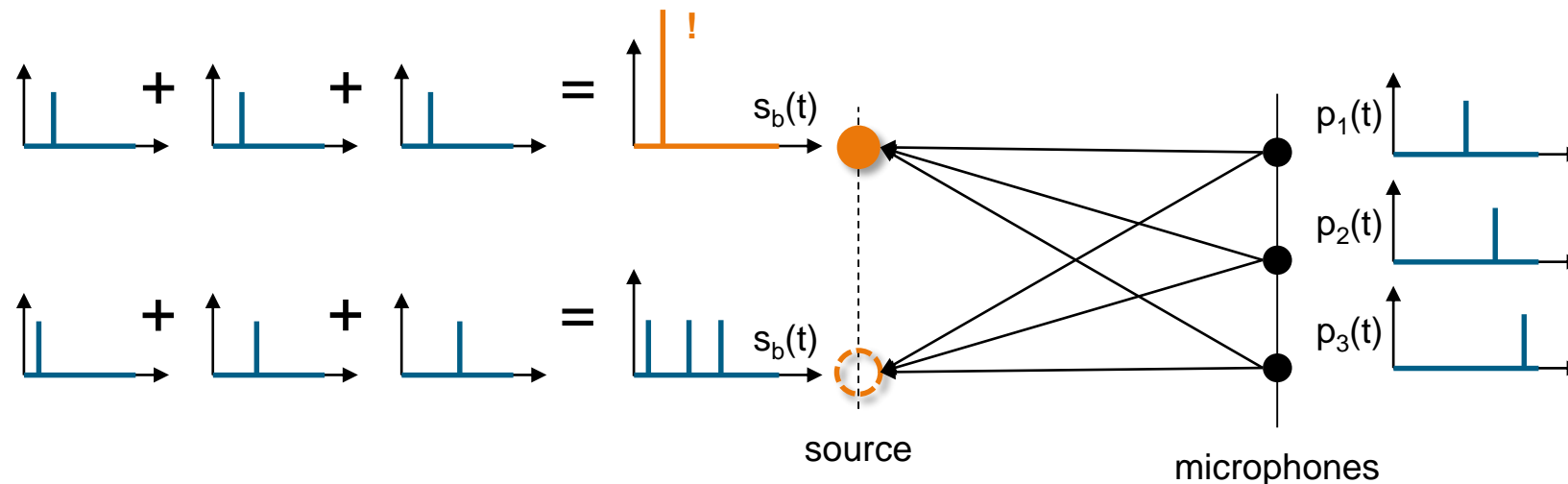


**Propagation of source signal:**

$$p_i(t) = s(t + \tau_i)$$

**Back-propagation:**

$$s_b(t) = \sum_i p_i(t - \tau_i)$$



**In practice:**

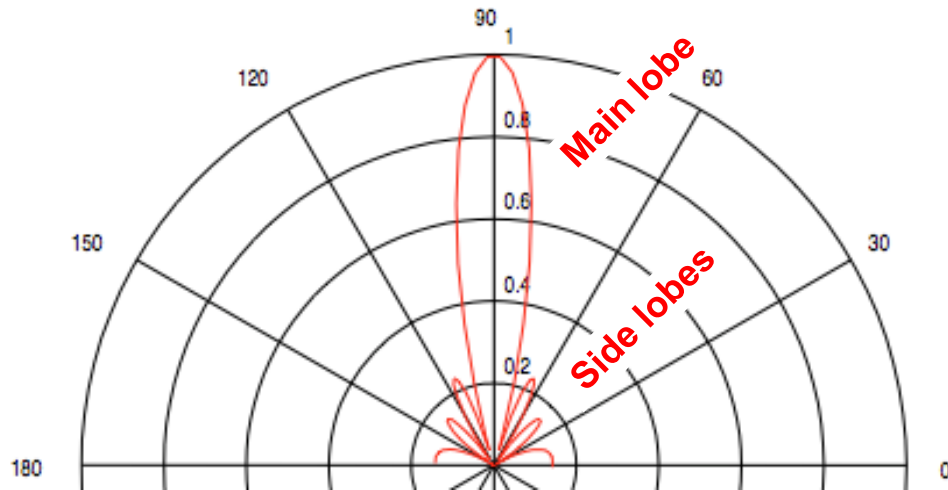
**Generalized 'filter-and-sum' method implemented in frequency domain**

$$s(f) = \sum p_i(f) \cdot w(f) \cdot e^{-2\pi j \tau_i}$$

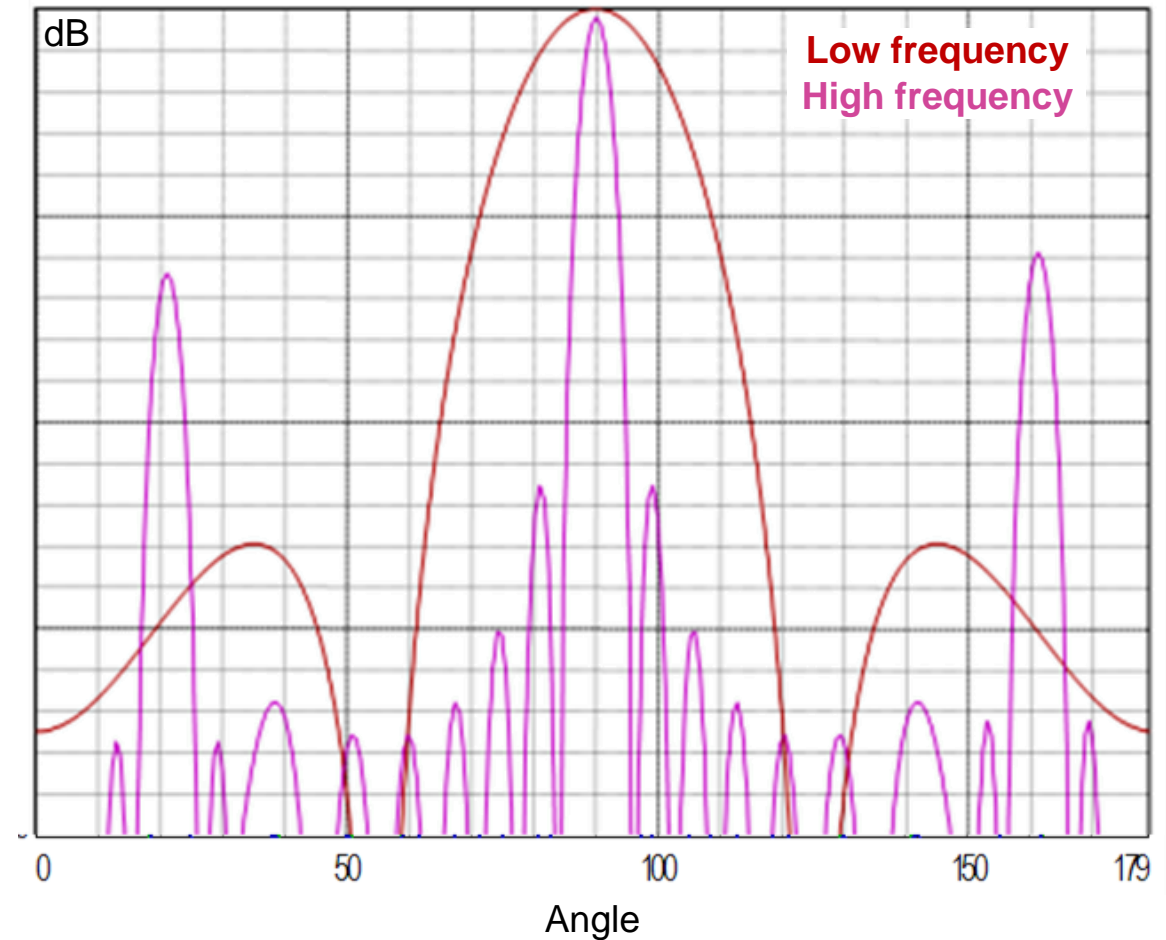
- **Source localization per frequency line possible**
- **Weighting function  $w(f)$  approximates physical source pressure level**

# Beamforming

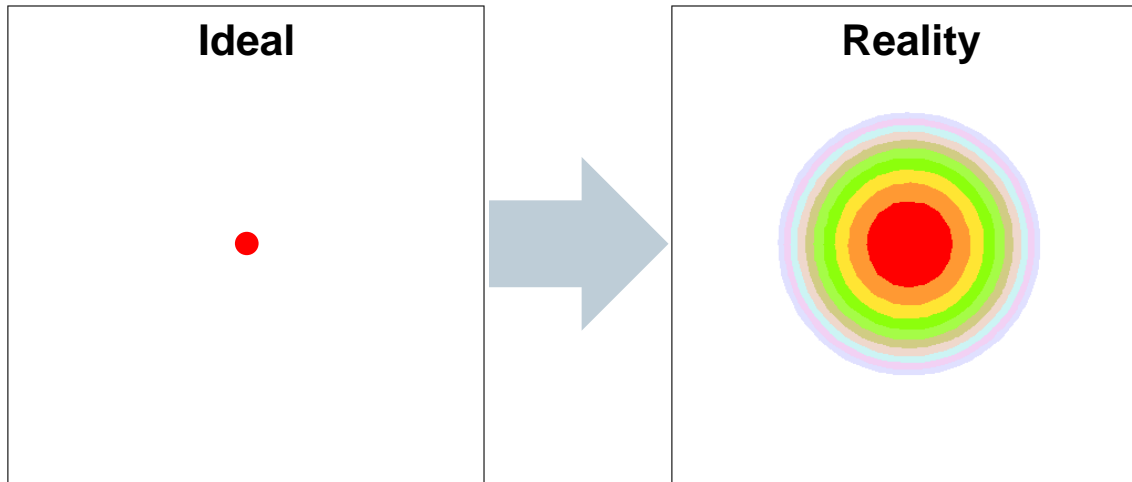
What influences localization quality?



- Delay-and-sum processing has a sensitivity pattern around the main 'beam' direction which causes distortion
- Factors influencing the sensitivity pattern:
  - Localization technique
  - Analysis frequency
  - Array design



# Beamforming Spatial resolution

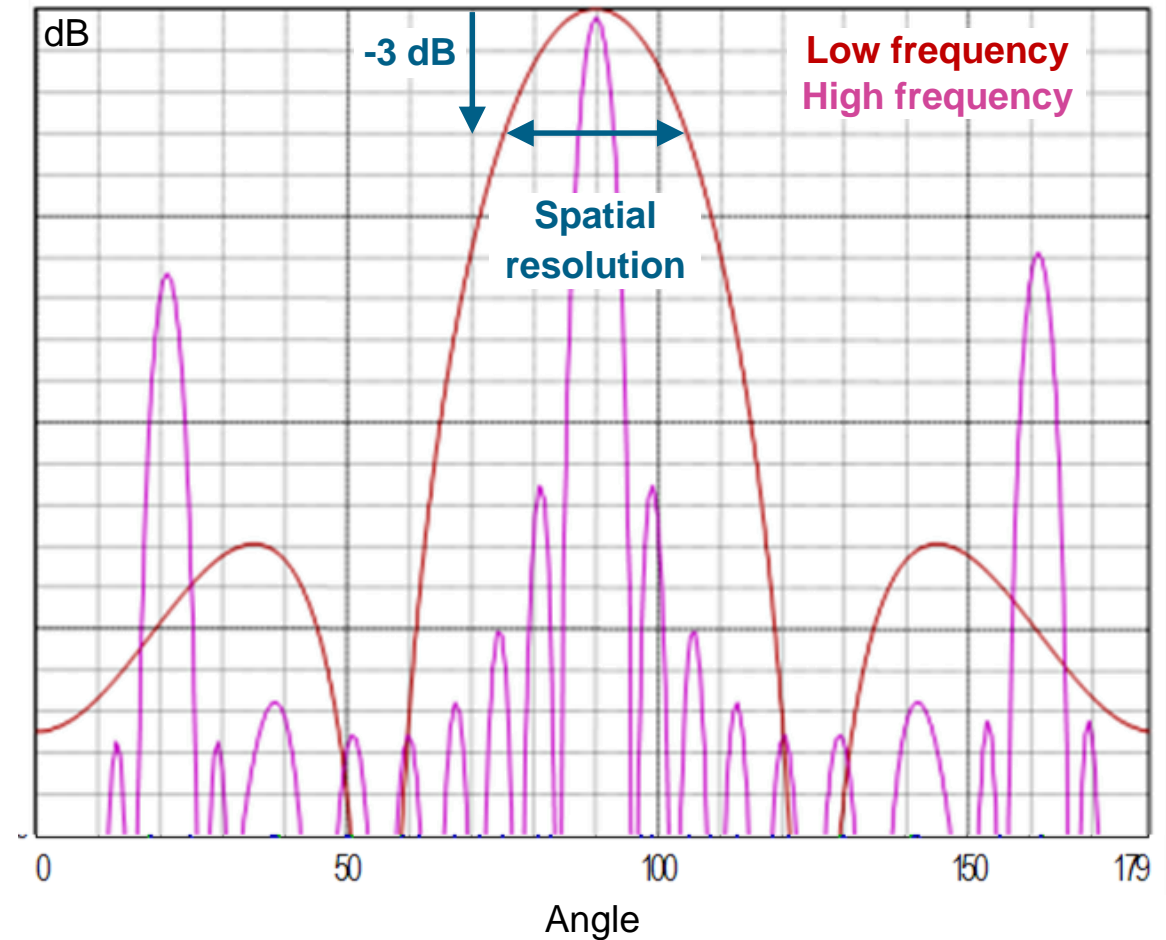


- Beamforming identifies a source **region** with a size depending on the **spatial resolution**:

$$\text{Spatial resolution} \approx \frac{d}{D} \lambda$$

- $d$  = array distance to source
- $D$  = array diameter
- $\lambda$  = analysis wavelength

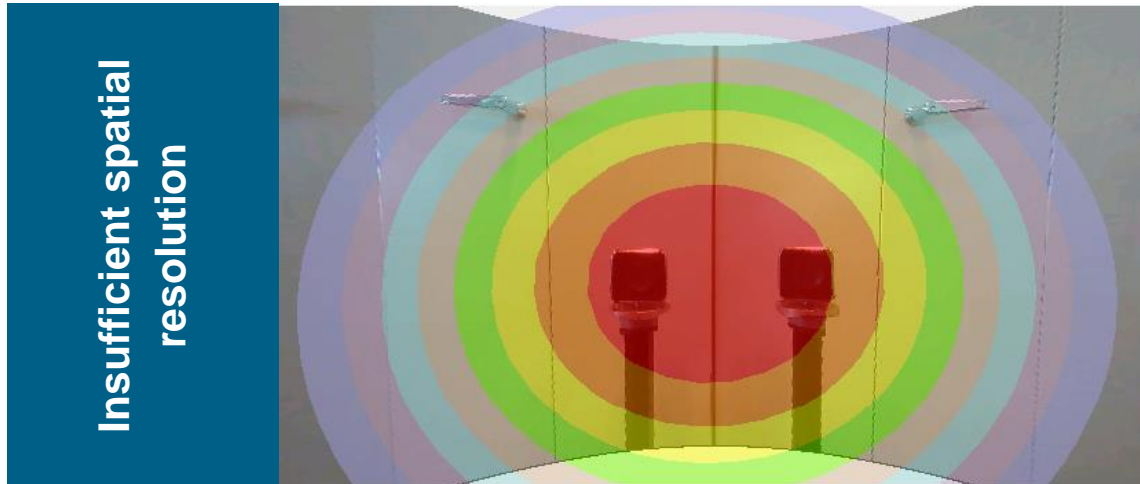
## Spatial resolution determines localization accuracy





# Beamforming

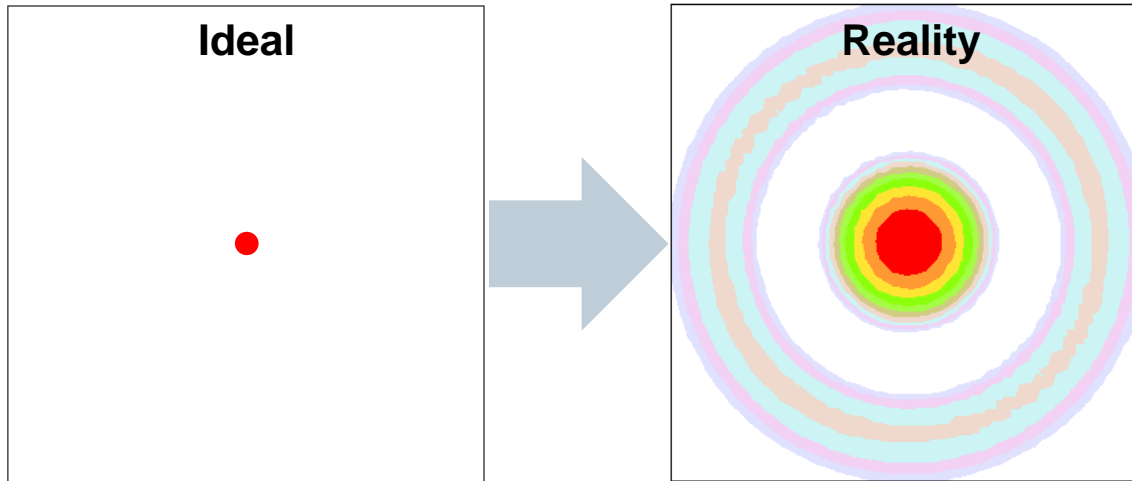
## Spatial resolution – Practical implications



- Sufficient spatial resolution required for:
  - Precise localization of sound source origin
  - Separation of closely spaced sound sources
- Beamforming is very potent for high frequency sound source localization, but suffers in the low frequency range where the wavelengths are large
- Spatial resolution can be improved by:
  - Increasing the array **diameter**,
  - Moving **closer** to the source
  - Analyzing **higher frequency** contributions
- Other localization methods can improve the spatial resolution in certain conditions: Focalization, iNAH, Deconvolution, Bayesian Focusing, etc.

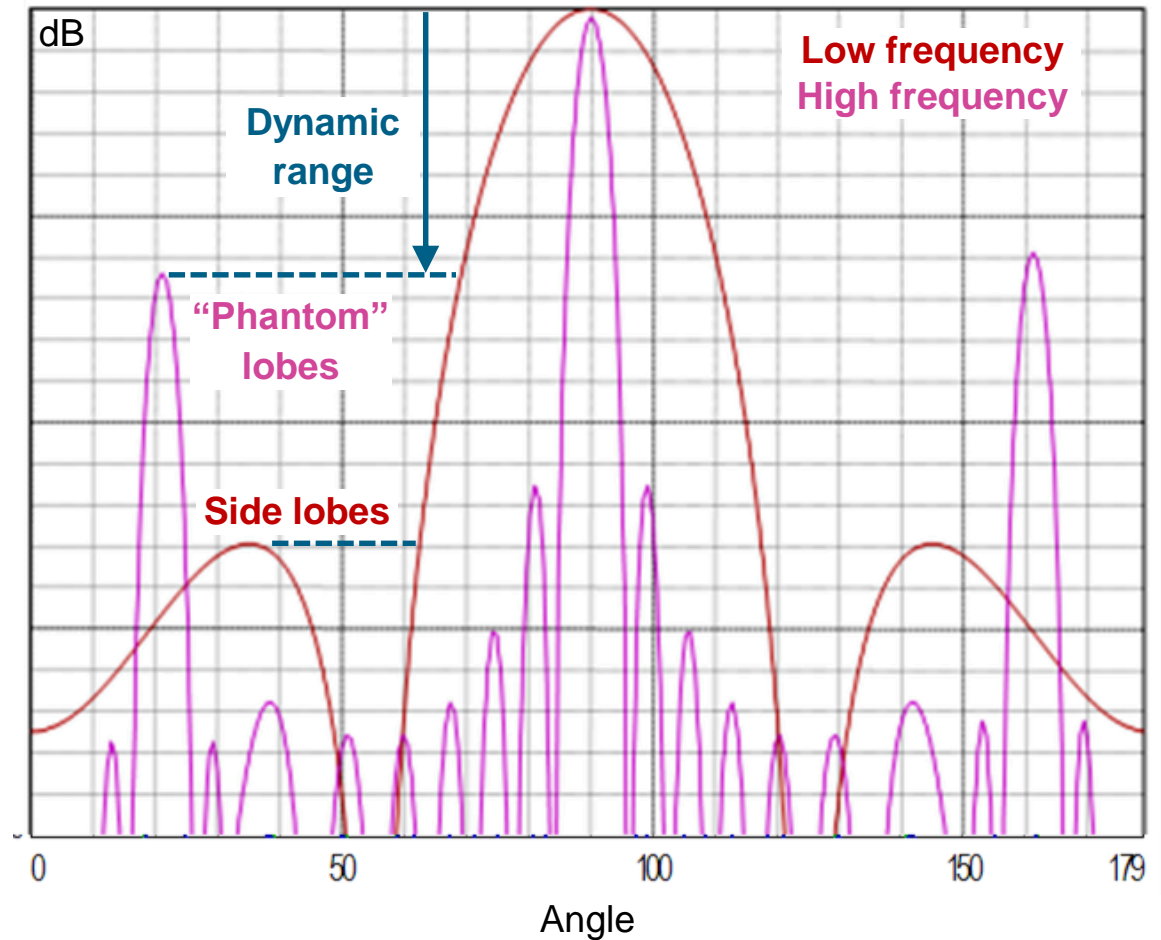
# Beamforming

## Dynamic range



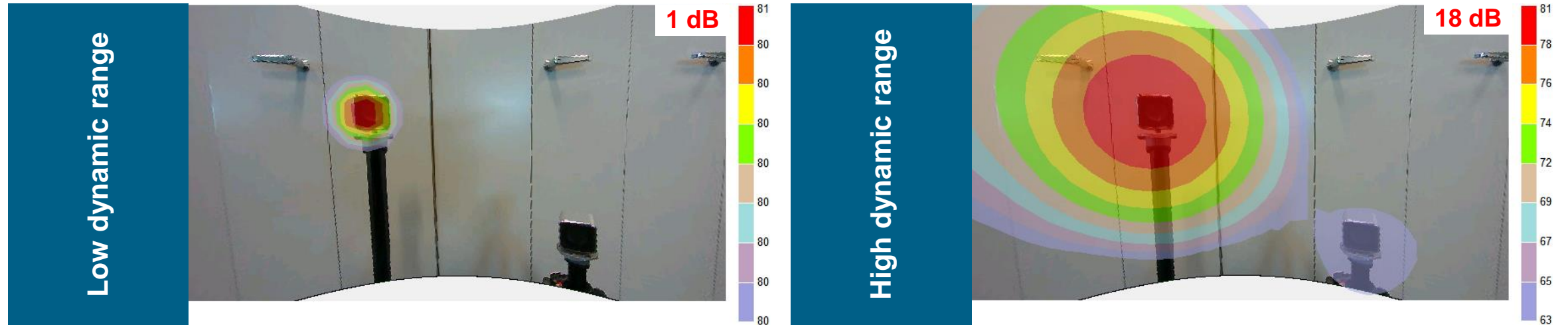
- Side lobes caused by redundancy in phase information between microphone pairs → microphone **layout**
- Phantom lobes caused by spatial undersampling of high frequency wavelengths → microphone **density**
- Dynamic range = zone between main lobe and highest side lobe which is **guaranteed distortion-free**

### Dynamic range determines localization confidence



# Beamforming

## Dynamic range – Practical implications

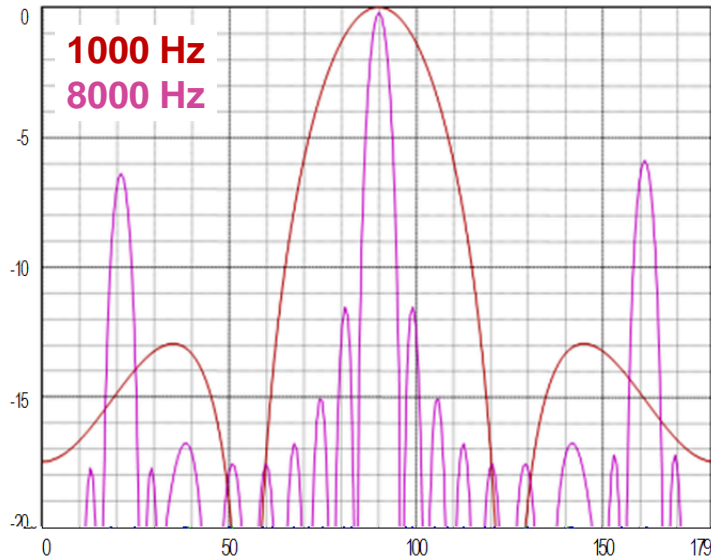


- Dynamic range applies **relative** to dominant source level
- Important aspect for sound source localization for **engineering** purposes:
  - Confidence in sources found within dynamic range
  - Ability to localize secondary (non-dominant) sources
- Dominant factor in determining dynamic range is the **array design**:
  - Microphone layout
  - Array size
- Dynamic range is a **quality indicator** of a given array



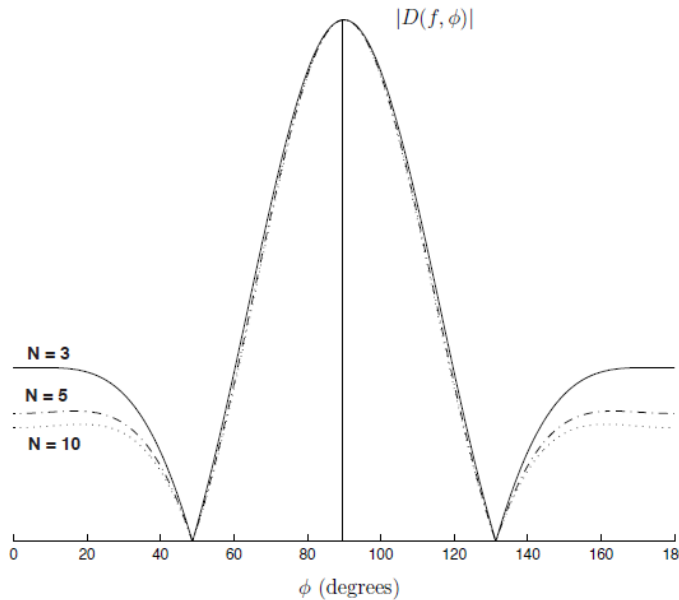
# Beamforming

## Influence of array design parameters



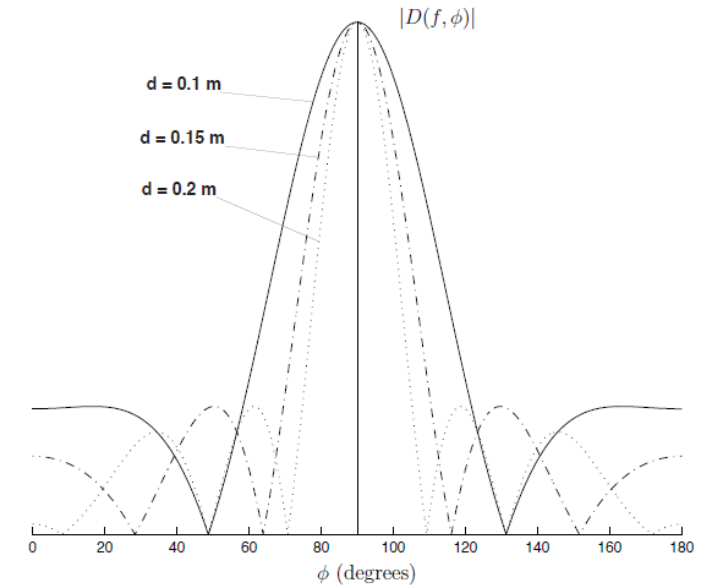
### Increase frequency:

- Main lobe width smaller
- Phantom lobes appear at very high frequencies



### Increase #mics $N$ :

- Side lobe levels reduce
- Array size equal, so spacing  $d$  decreases



### Increase array size:

- Main beam width reduces
- #mics  $N$  equal, so spacing  $d$  increases

# Focalization

## A variant to Beamforming

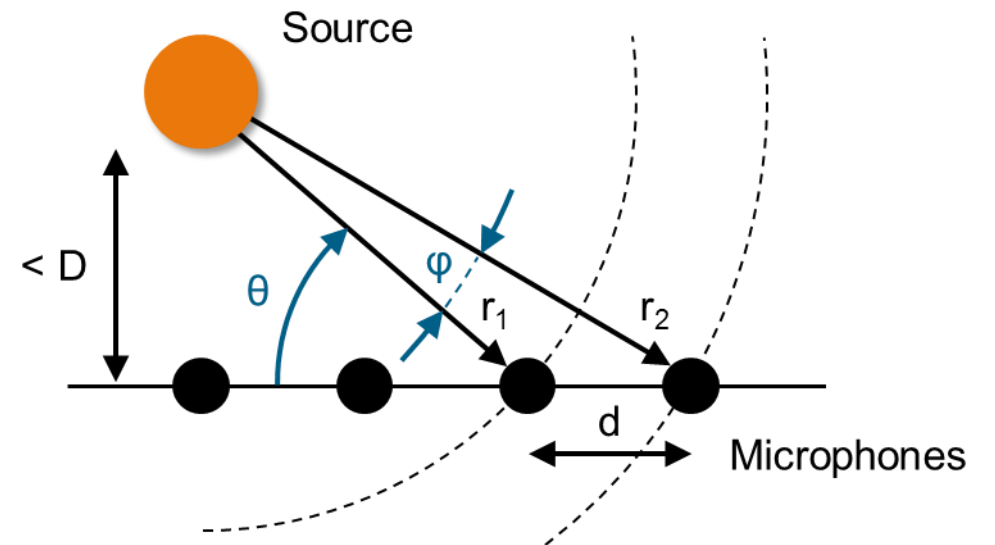
- Planar pressure wave assumption not valid in **near-field**
- When distance to source < array diameter D, pressure waves are better approximated as **spherical** waves

- Time delay between microphones can be expressed as:

$$\tau = \frac{r_2 - r_1}{c} = \frac{d \cdot \cos(\theta) \cdot (\cos(\frac{\varphi}{2}) + \sin(\frac{\varphi}{2}) \cdot \tan(90 - \frac{\varphi}{2}))}{c}$$

- At large source distances ( $r_1 \rightarrow \infty$ ) angle  $\varphi$  becomes small ( $\varphi \rightarrow 0$ ) and Focalization becomes more similar to classical Beamforming:

$$\left. \begin{array}{l} \sin\left(\frac{\varphi}{2}\right) \cdot \tan\left(90 - \frac{\varphi}{2}\right) \rightarrow 0 \\ \cos\left(\frac{\varphi}{2}\right) \rightarrow 1 \end{array} \right\} \tau \rightarrow \frac{d \cdot \cos(\theta)}{c}$$



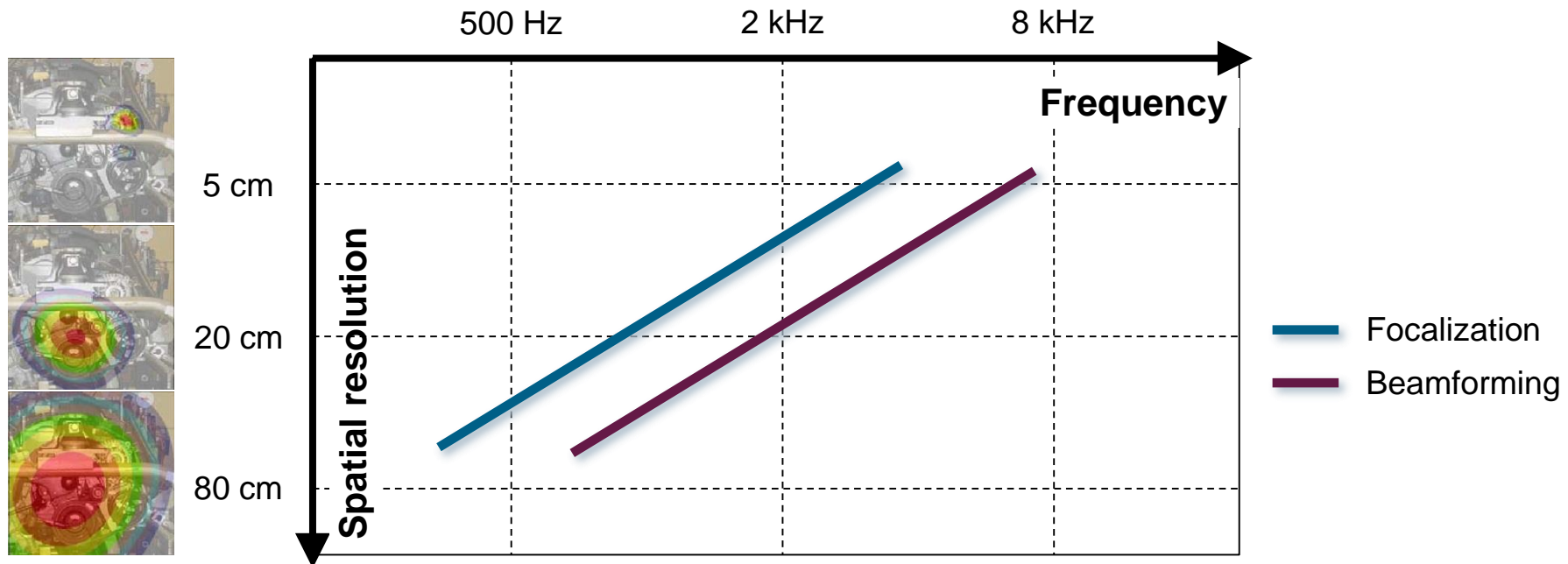
**Practical benefit:**  
Spatial resolution improves by factor ~2

# Beamforming & Focalization Summary

Beamforming spatial resolution  $\approx \lambda$  (ideal)

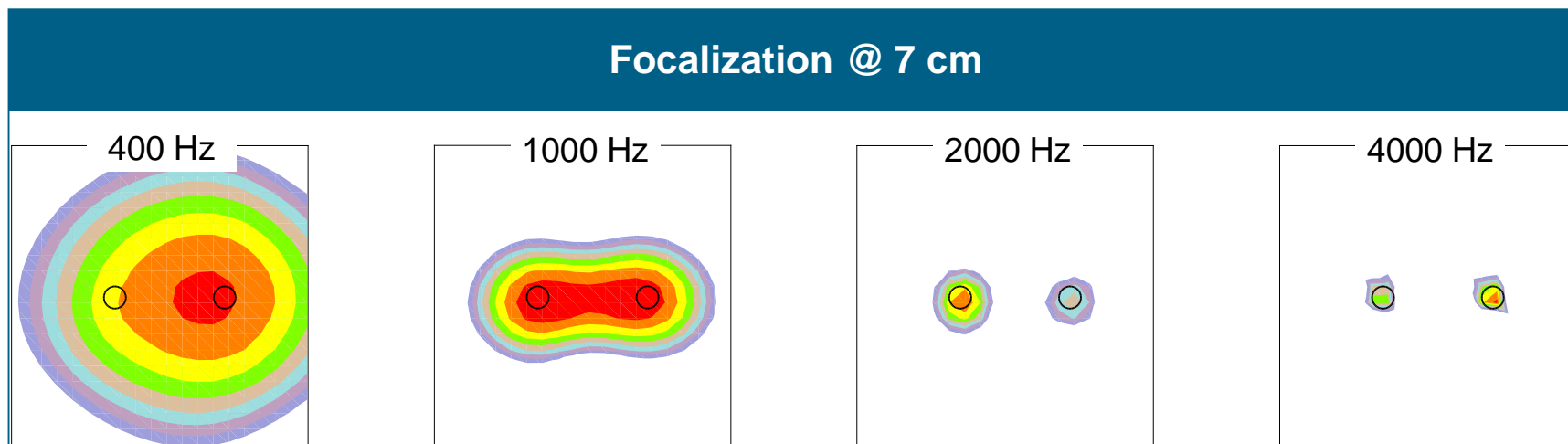
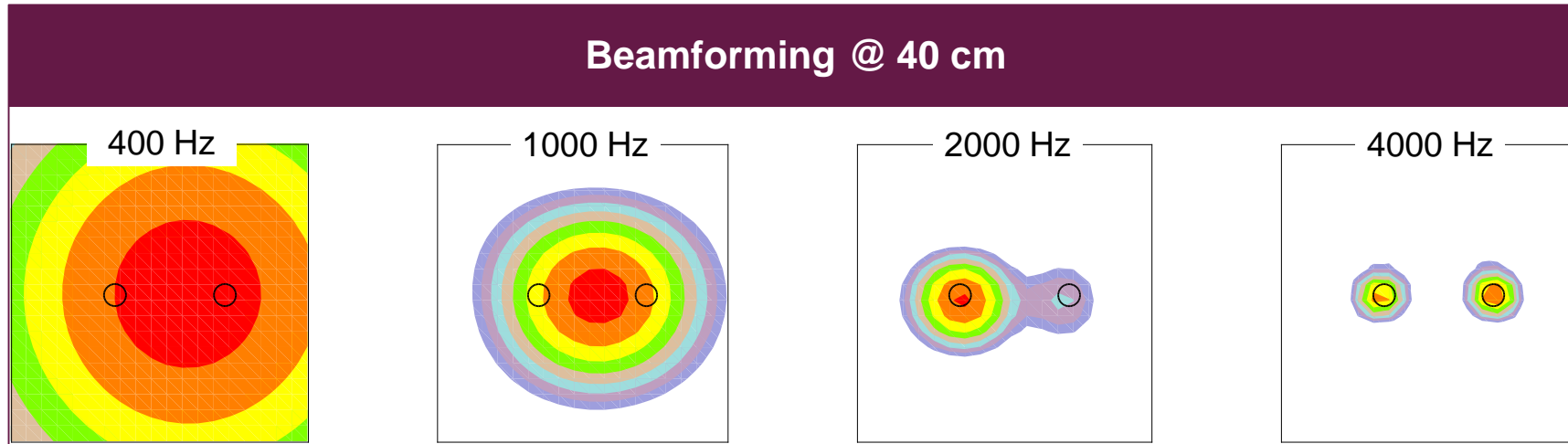
Focalization spatial resolution  $\approx \frac{1}{2} \lambda$  (ideal)

- Many factors have been seen to influence real-world localization performance
- Best results obtained at **high analysis frequencies** and **short measurement distances**





# Beamforming & Focalization Synergy



**Best results obtained  
with combination of  
Beamforming and  
Focalization**

# Beamforming & Focalization Synergy – Practical example

## 1. Beamforming

- Far-field technique
- Low resolution
- Global overview

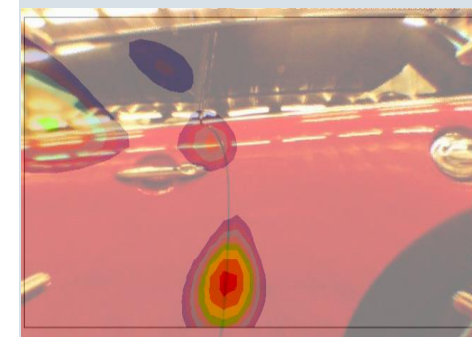


Where does wind noise leak into this vehicle?



## 2. Focalization

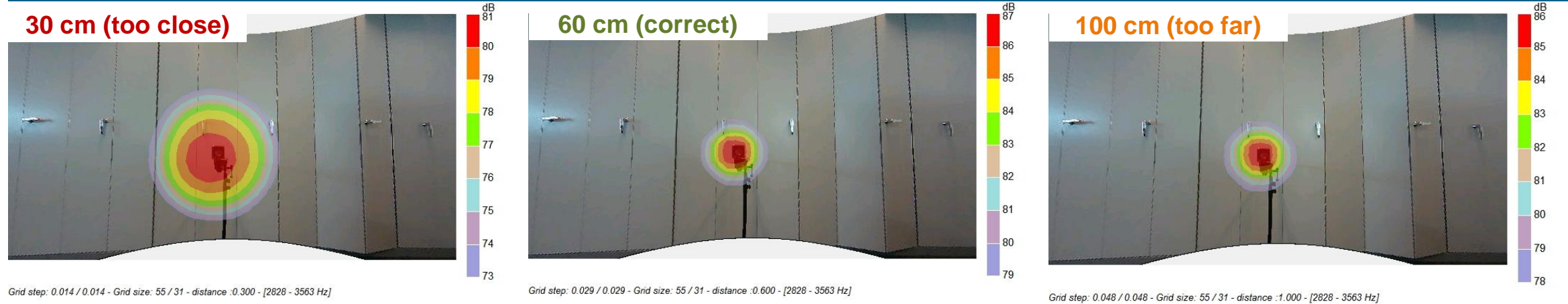
- Near-field technique
- High resolution
- Precise localization



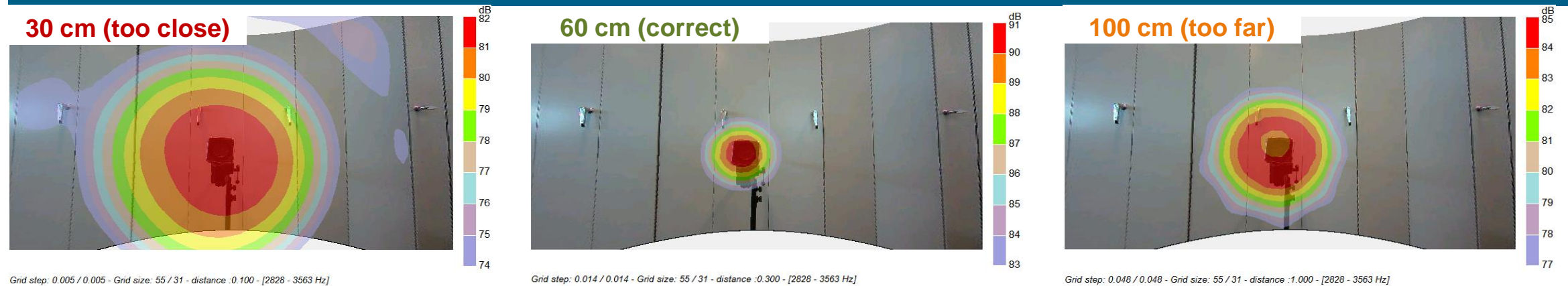
# Beamforming & Focalization

## Measurement distance vs. calculation distance

Far-field: time delay dominated by distance between microphones → minor influence by incorrect distance



Near-field: spherical waves sensitive to propagation distance → major influence by incorrect distance





# Fundamentals of Sound Source Localization

## Agenda



- Introduction to Sound Source Localization
- Beamforming-based localization methods
- **Far-field Deconvolution methods**
- Map averaging methods

# Far-field Deconvolution

## What is it?

### What?

- **Iterative & quantitative** source localization method
- Increases spatial resolution vs. Beamforming-based methods by **factor ~4 to 5**, and enables **objective source ranking** via Sound Power estimations

### Why?

- Beamforming underperforms in cases with **multiple sources** in **close proximity** at **large distances**
- Alternative methods not always applicable in those scenario's (e.g. near-field Holography)

### How?

- Assume discrete **monopole source distribution** with **uncorrelated** spherical sound radiation
- Optimal monopole distribution to match measured Beamforming result found via **iterative optimization**

### When?

- **Far-field**, distance to source > array diameter
- Localization only: **low to mid** frequency range
- Quantification: **low to high** frequency range
- **Uncorrelated** sources only (e.g. aero-acoustics)

# Far-field Deconvolution

## How does it work?

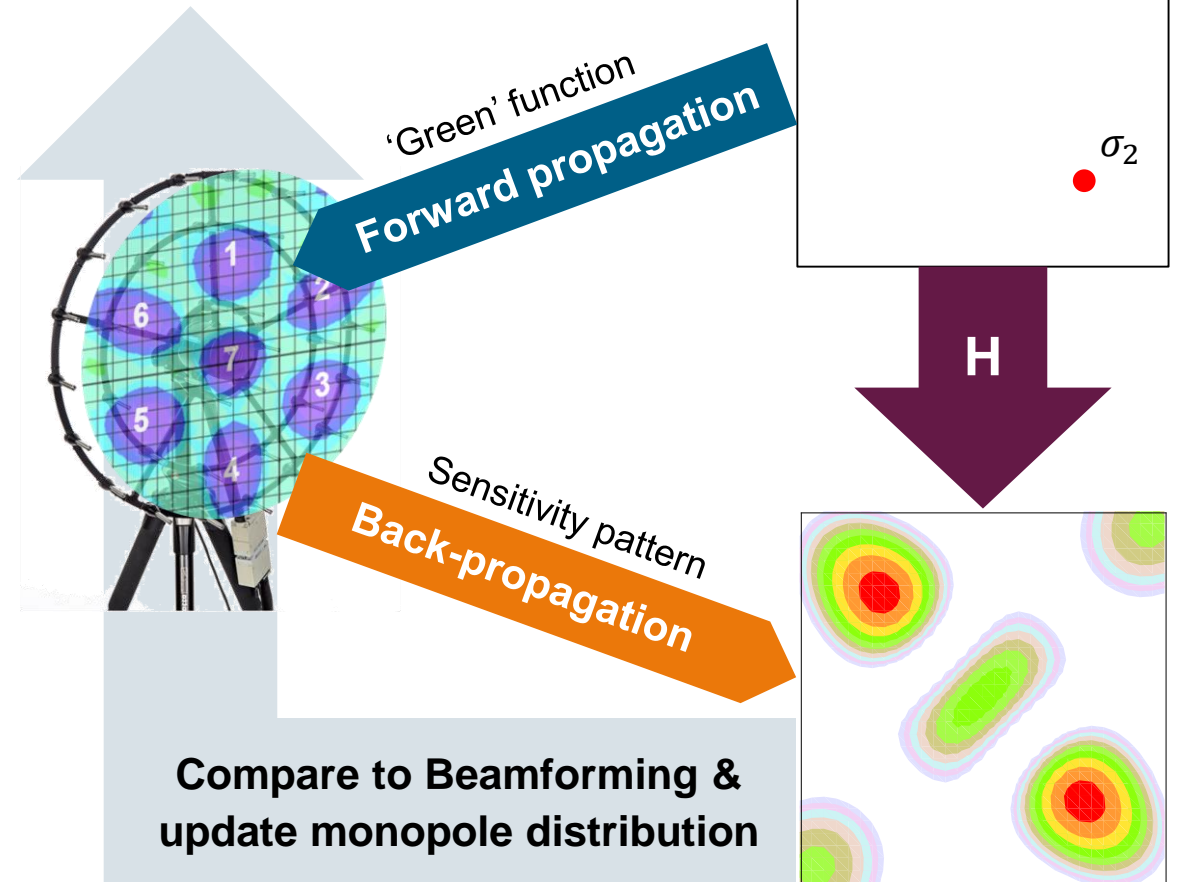
- Find the monopole distribution for which the **predicted** Beamforming result matches the **measured** result
- Mathematically expressed as the minimization of the following cost function (**CIRA** method):

$$\min_{\sigma, \sigma_i \geq 0} C(\sigma) = \sum_i \left( \sum_j (H_{ij} \cdot \sigma_j) - B_i \right)^2$$

- $\sigma$  = distribution of monopole sources  $\sigma_{1..J}$
- $H_{ij}$  = transformation matrix from monopole @ point j to Beamforming result @ point i
- $B_i$  = measured Beamforming result @ point i

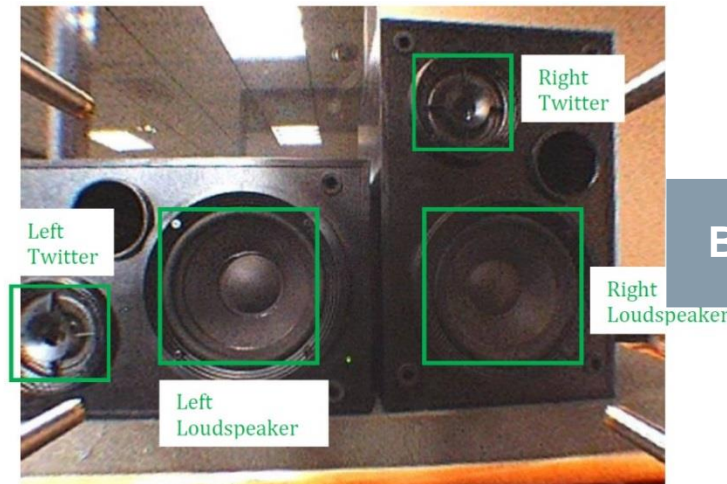
Monopole source model dramatically improves spatial resolution and allows Sound Power estimation!

Monopole source distribution  
@ iteration k

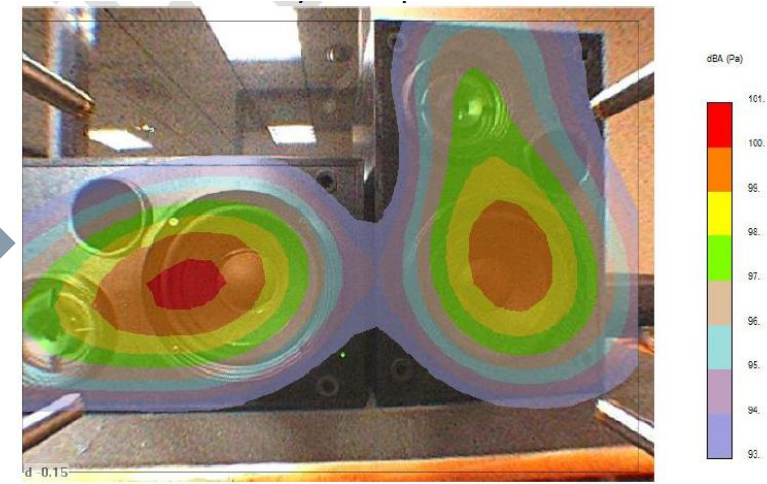


# Far-field Deconvolution

## CIRA – Practical examples

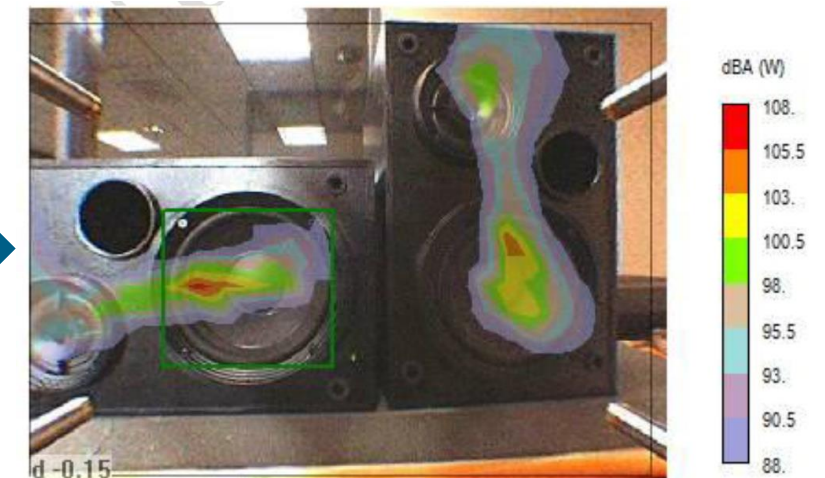


Beamforming (1 – 2 kHz)



- Dramatic improvement of **spatial resolution**
- Dramatic improvement of **dynamic range** (8 dB → 20 dB)
- Quantified results in [dBW]

CIRA (1 – 2 kHz)

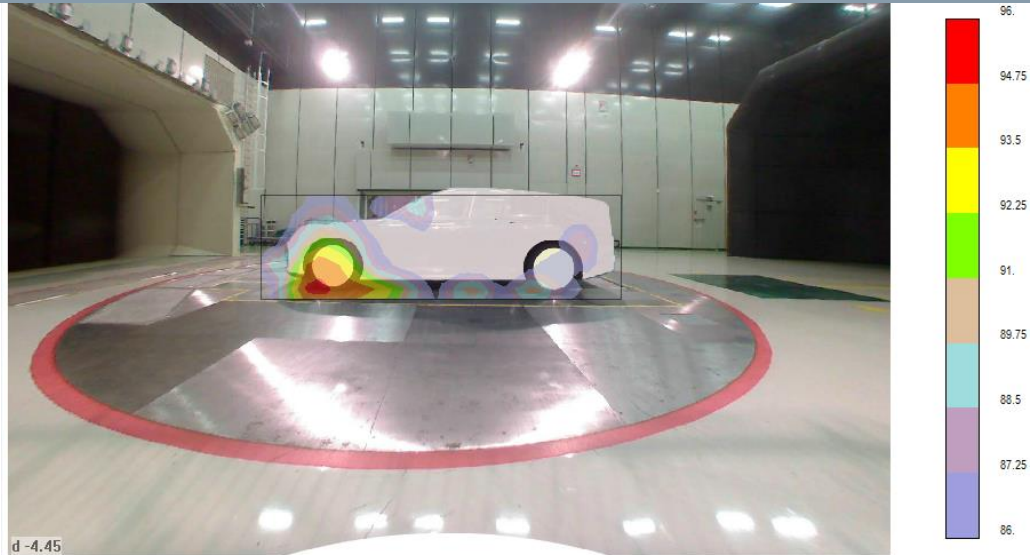




# Far-field Deconvolution

## CIRA – Practical examples

### Beamforming



- Aero-acoustic sources in wind tunnel @ 4.45 meters
- Beamforming provides high-confidence but low-accuracy initial guess on source positions

### CIRA



- Confirms Beamforming result and improves spatial resolution
- Sources quantified → allows objective comparison

# Far-field Deconvolution

## CIRA vs. Clean-SC

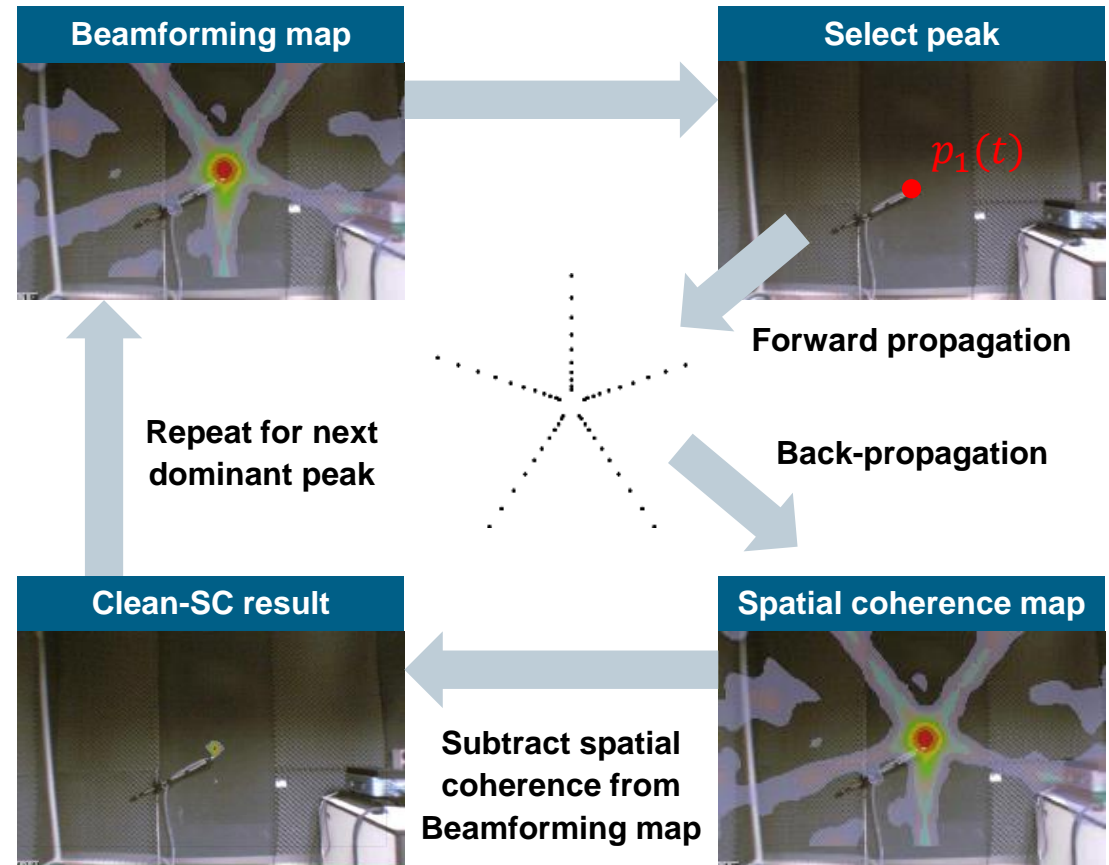
### Iterative Least-squares

- All monopoles are processed in the same loop
- Finds optimal monopole distribution through iterative optimization of cost function (previous slides)
- **CIRA** in Testlab: improved convergence criteria

### Iterative Cleaning

- Each monopole is processed in a separate loop
- Finds optimal monopole position for dominant source in current loop, then removes coherent components
- **Clean-SC** in Testlab: based on spatial coherence

### Clean-SC procedure

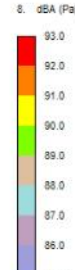
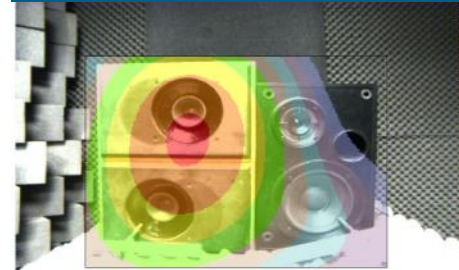


# Far-field Deconvolution

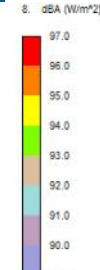
## CIRA vs. Clean-SC – Practical example

900 – 1120 Hz

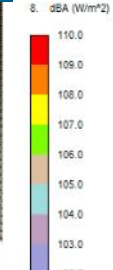
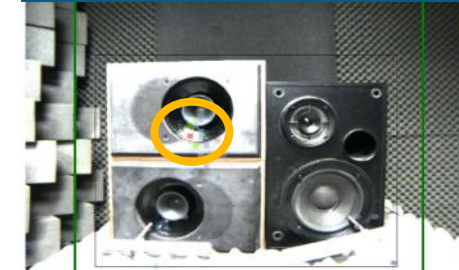
**Beamforming**



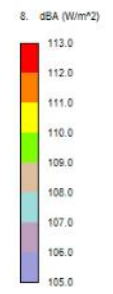
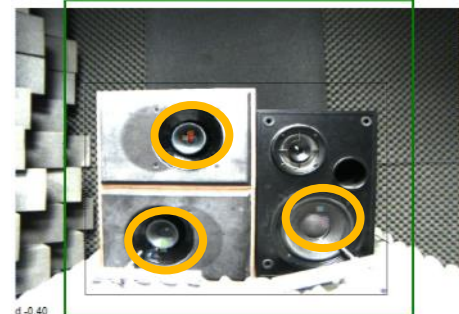
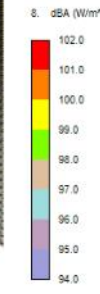
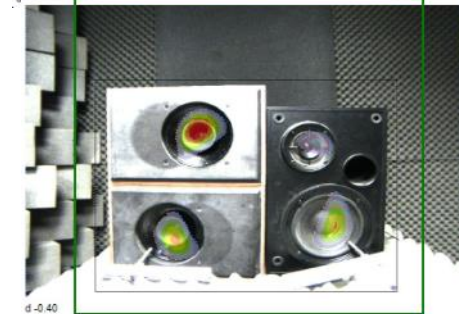
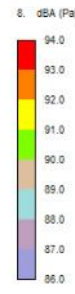
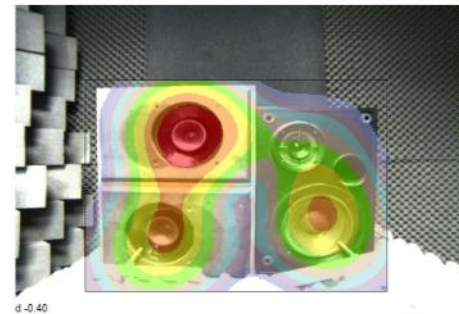
**CIRA**



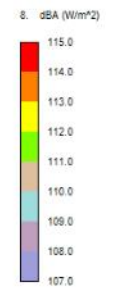
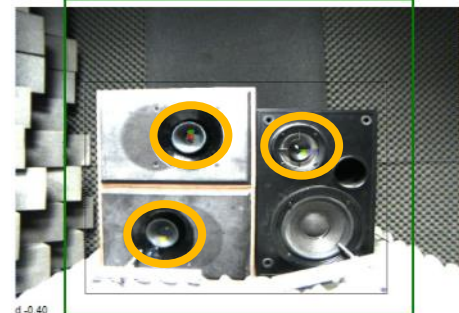
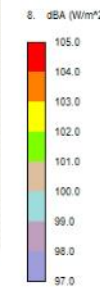
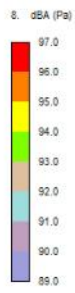
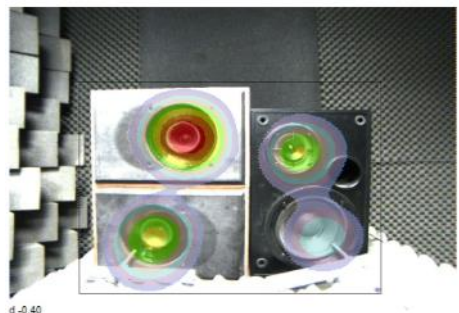
**Clean-SC**



1420 – 1780 Hz



2250 – 2820 Hz



# Far-field Deconvolution

## CIRA vs. Clean-SC – Summary

### CIRA

#### Advantages vs. Beamforming

- Improves dynamic range
- Improves spatial resolution (reconstruction with monopoles)
- Improves source separation capability
- Quantitative results in Sound Power

#### Disadvantages

- Only valid for uncorrelated sources, sensitive to correlation between sources
- Calculation time of multi-variable optimization

### Clean-SC

#### Advantages vs. Beamforming

- Strongly improves dynamic range
- Improves the spatial resolution (reconstruction with monopoles)
- Quantitative results in Sound Power
- Very fast calculation time for Deconvolution method

#### Disadvantages

- Only valid for uncorrelated sources, **very** sensitive to correlation between sources
- Does not improve source separation capability



# Fundamentals of Sound Source Localization

## Agenda

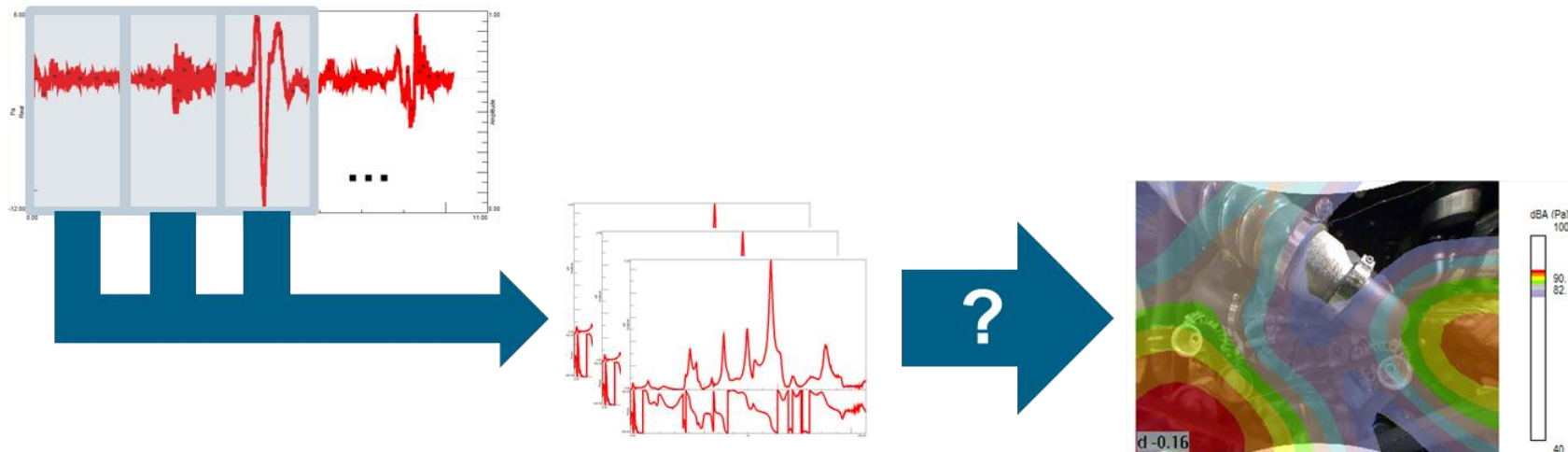
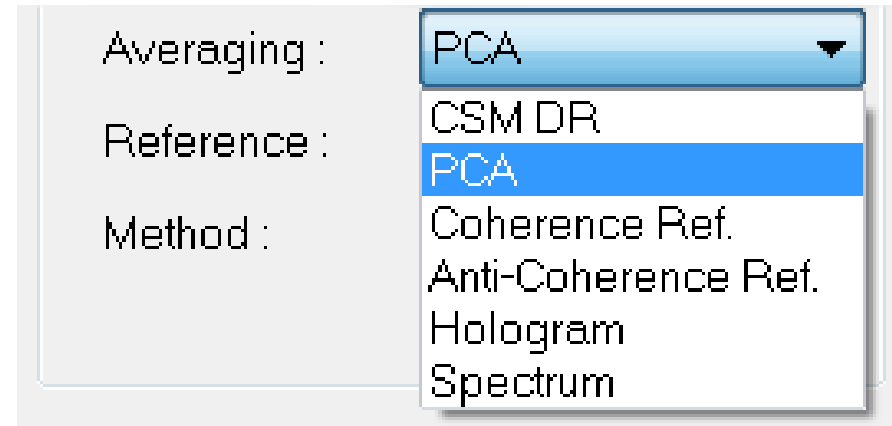


- Introduction to Sound Source Localization
- Beamforming-based localization methods
- Far-field Deconvolution methods
- **Map averaging methods**

# Map averaging methods

## How do we handle time-domain averaging?

- One time block per microphone is sufficient to calculate one hologram result
- Different strategies exist for combining holograms from multiple time blocks, depending on the source type:
  - Stationary vs. transient
  - Correlated vs. uncorrelated



# Map averaging methods

## Spectrum averaging

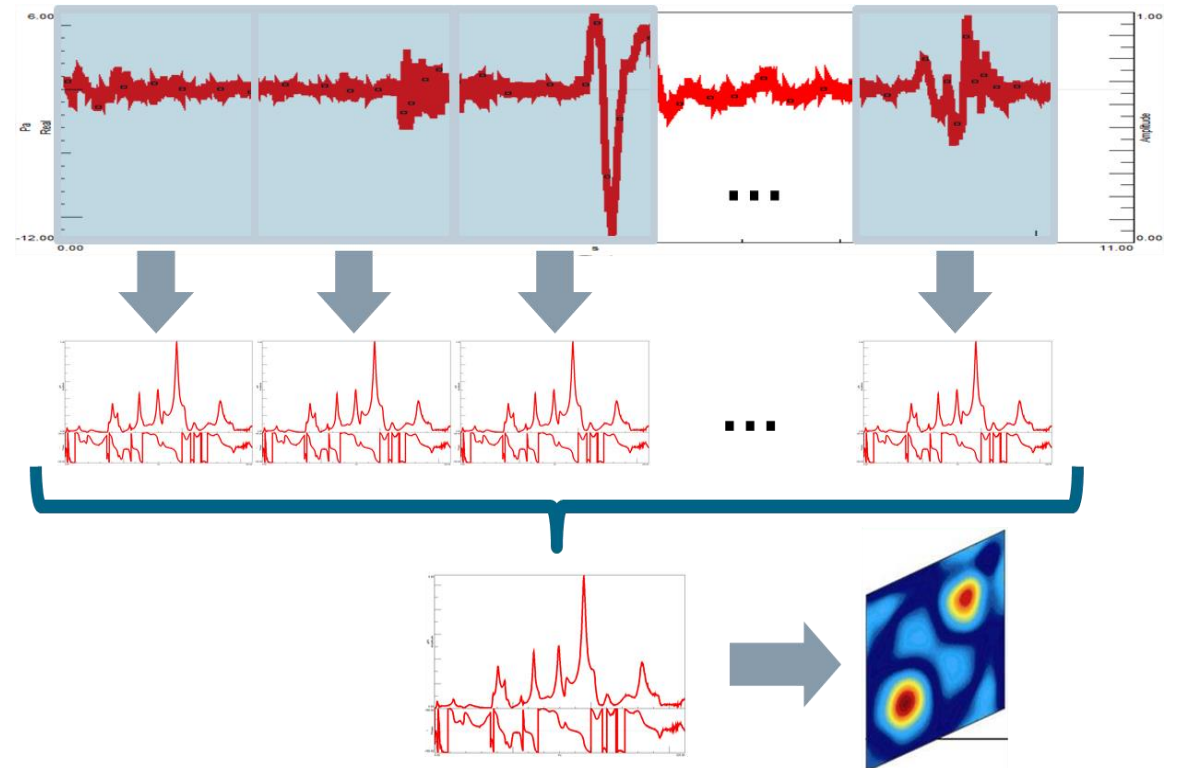
### Process

- 1) Calculate averaged spectrum over all time blocks
- 2) Calculate hologram from averaged spectrum

### Usage

- Extremely fast calculation time → animations!
- Requires phase reference signal
- Requires stationary and correlated sources to avoid information loss during averaging

### Spectrum averaging



# Map averaging methods

## Principal Components Averaging

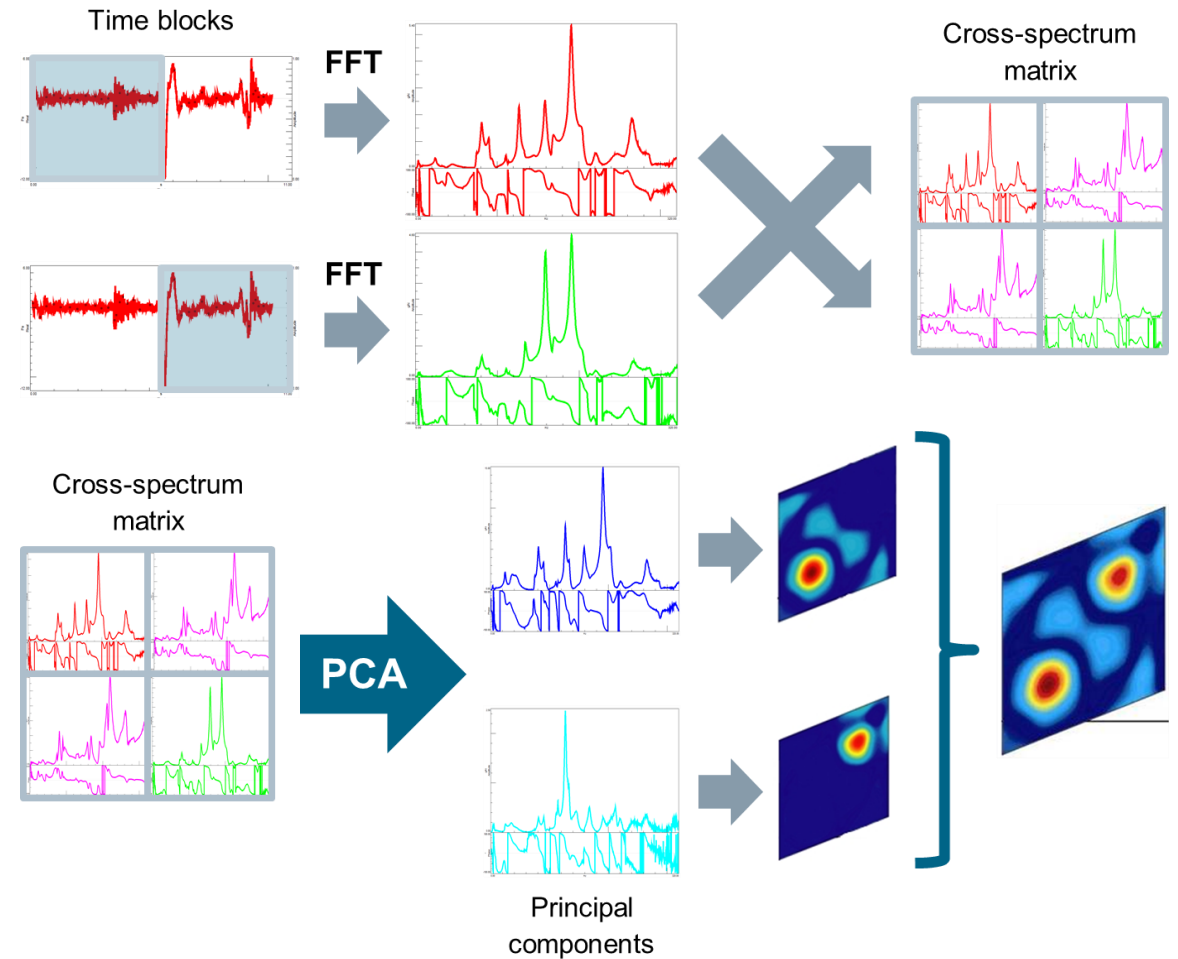
### Process

- 1) Calculate cross-spectrum matrix between time blocks
- 2) Apply PCA to CSM → identifies independent sources
- 3) Calculate spatially averaged hologram over all principal components

### Usage

- Does not eliminate uncorrelated sources and does not require phase reference signal → general purpose
- Only suitable for stationary sources

### Principal Components Averaging





# Map averaging methods

## Hologram averaging

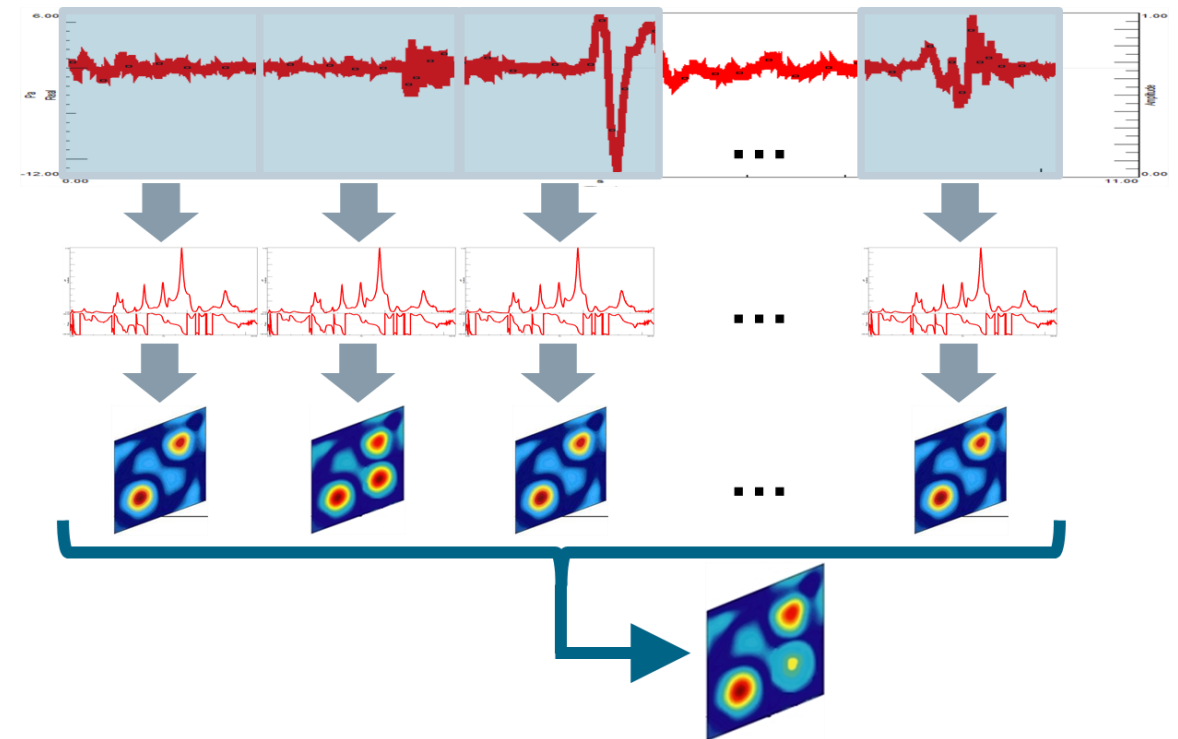
### Process

- 1) Calculate one spectrum per time block
- 2) Calculate one hologram per spectrum
- 3) Calculate spatially averaged hologram over all time blocks

### Usage

- Averaging process does not completely eliminate transient components
- Does not require phase reference signal
- High computational effort

### Hologram averaging



# Map averaging methods

## Cross-Spectrum Matrix Diagonal Removed averaging

### Process

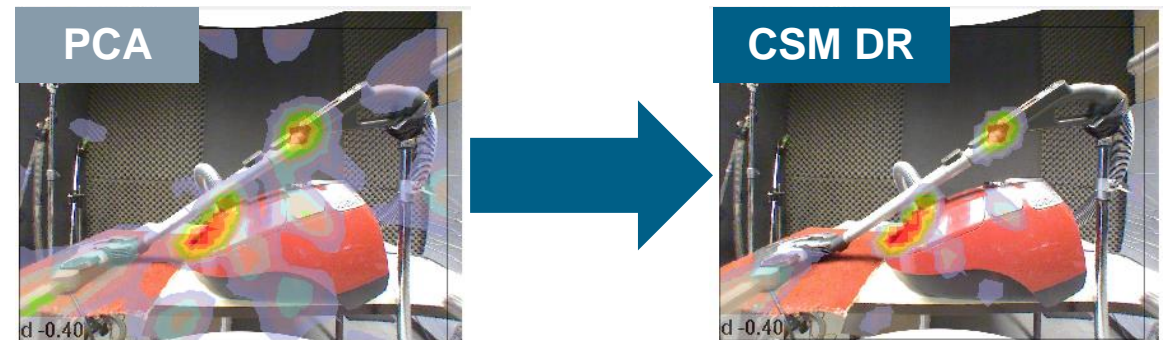
- 1) Remove diagonal components from CSM in the classical Beamforming expression
- 2) Apply Hologram averaging to the result

### Usage

- Removal of microphone **self-noise** (e.g. wind noise, sensor noise, etc.)
- Extends Hologram averaging, significantly improves dynamic range (especially **aero-acoustic** sources)

















### CSM DR averaging

- Cross-Spectrum Matrix formulation of classical Beamforming with N microphones at point k:  
$$B_k(f) = w_k'(f) CSM(f) w_k(f), CSM(f) = [p_i(f) p_j^*(f)]_{N \times N}$$
- Observation: microphone autopowers on diagonal are very sensitive to **self-noise** → dynamic range reduction
- Removal of diagonal components improves quality of Beamforming result by reducing overall noise:



# Map averaging methods

## Summary

	Spectrum	PCA	Hologram	CSM
Correlated sources				
Uncorrelated sources				
Non-stationary sources				
Calculation time				

[www.siemens.plm/simcenter](http://www.siemens.plm/simcenter)



# Fast new processing kernel in HDCAM



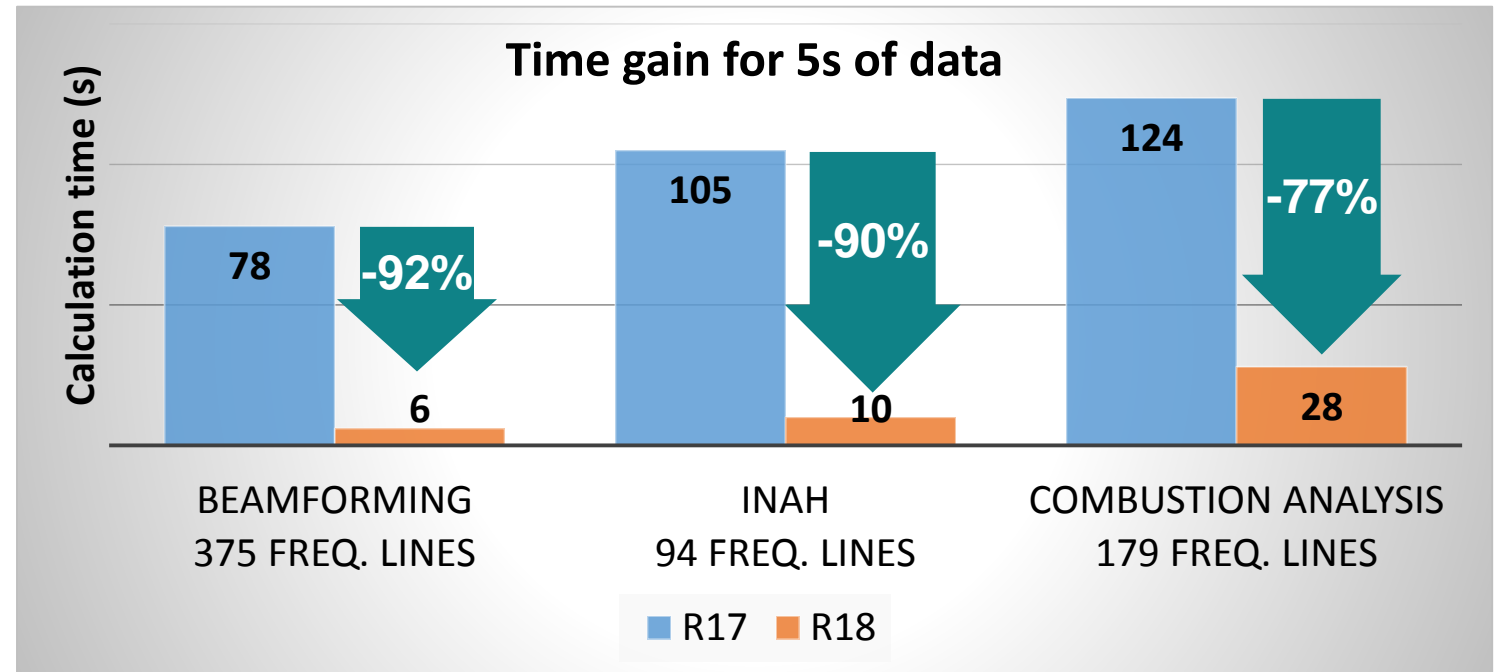
## Up to 10x faster results for all methods

Supports all methods: beamforming, focalization, iNAH, deconvolution

Benchmark shows improvement by factor 7-10 for all methods

Fast processing using state of the art GPU and multi-core computing.

Supports NVidia cards with CUDA GPUs: (GeForce 9xx, Quadro Mxxx).



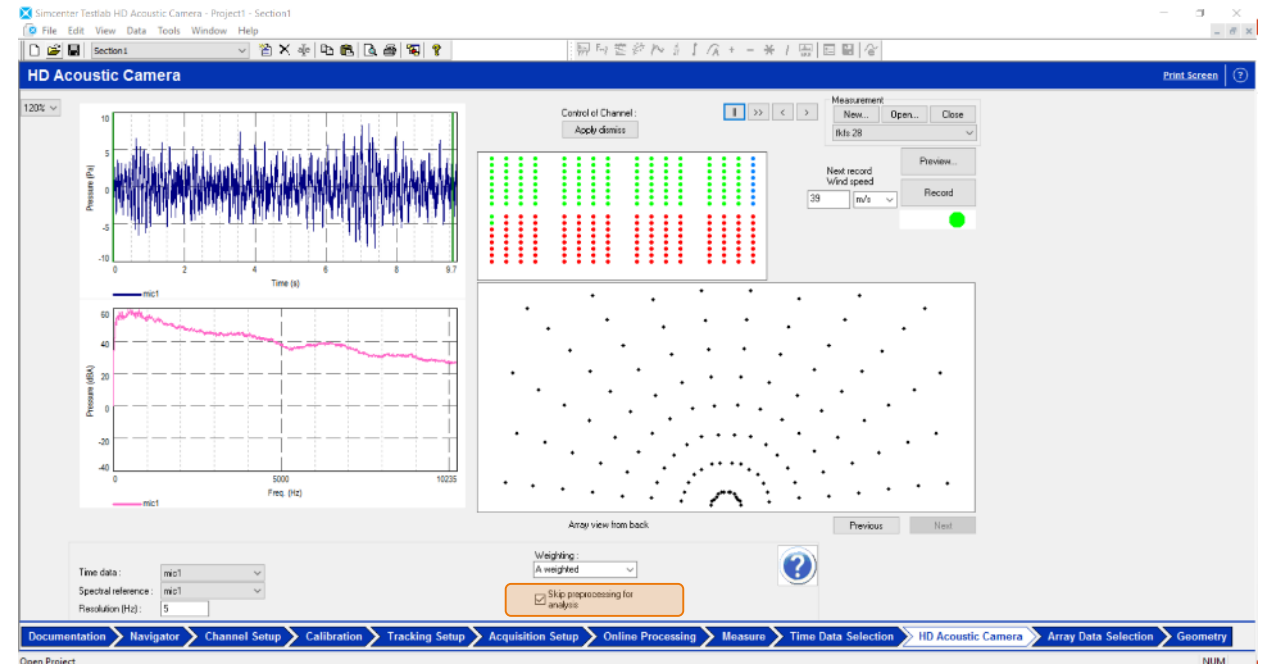
# Shortened time to next measurement in HDCAM



**HDCAM enables faster repetitive measurements without having to wait for preprocessing for analysis**

**Improves time-to-next-measurement to 5-10s.**

**Repetitive sequential measurements do not require preprocessing for analysis after each recording**



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The screenshot displays the Simcenter Testlab HD Acoustic Camera software interface. The main window is titled 'HD Acoustic Camera' and contains several panels. On the left, there are two plots: a time-domain plot of Pressure (Pa) vs. Time (s) and a frequency-domain plot of Pressure (Pa) vs. Freq. (Hz). The right side of the interface features a 'Control of Channel' section with a 'Next' button and a 'Skip preprocessing for analysis' checkbox. A red box highlights the 'Calculation' section, which includes a 'Next' button and a 'Skip preprocessing for analysis' checkbox. A red arrow points from the 'Skip preprocessing for analysis' checkbox to the 'Next' button, indicating that skipping preprocessing allows for a faster transition to the next measurement. The bottom of the interface shows a navigation bar with various tabs like 'Documentation', 'Navigator', 'Channel Setup', 'Calibration', 'Tracking Setup', 'Acquisition Setup', 'Online Processing', 'Measure', 'Time Data Selection', 'HD Acoustic Camera', 'Array Data Selection', and 'Geometry'.