THE DEVELOPMENT OF ROOM ACOUSTICAL CALCULATION AND MEASUREMENT METHODS FROM SABINE TILL TODAY

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Contents of the presentation

- Calculation:
  - the Sabine model and refinements
  - models to calculate impulse responses
  - state-of-the-art (don't forget perception!)
- Measurement:
  - measuring the 'Sabine quantitites'
  - measuring impulse responses
  - state-of-the-art: array measurements

Room acoustical calculation

- Required: a model of the sound field in the room – i.e., a (physically acceptable) simplification of the complex reality
- And then: the correct input data have to be filled in!

Sabine's model (1900): diffuse field (1)

- Diffuse field: in terms of expectation (rms) values the sound pressure is equal at all places ('homogeneous') and in all directions ('isotropic')
- Attractive by its simplicity
- Physically hardly acceptable:

  - concentrations of absorption (dressed audience!)

  - l.f.-resonances

  - ……

Sabine's model: diffuse field (2)

- Reverberation time formula:

$$T_{60} = V/6A$$

- SPL formula:

$$L_p = L_w - 10\log A/4$$

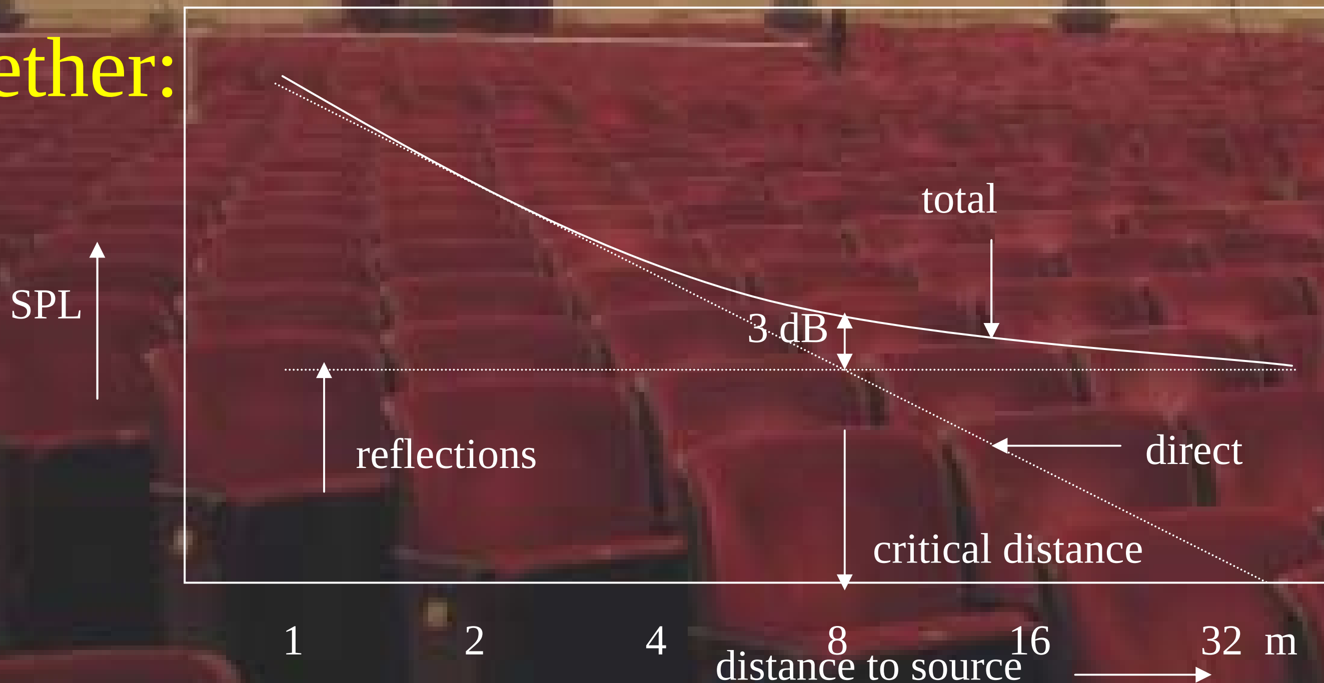
- Results unreliable in practice:
  - model doubtful
  - A difficult to determine
- Therefore: permanent attempts for improvement

Refinement: critical distance

- Direct sound is considered separately: ‘one way’ propagation away from source (= non-isotropic)
- Reflected sound (‘reverberation’) is considered to be diffuse
- At the critical distance, direct and reflected sound are equally strong

The critical distance in figure

- Direct sound pressure level decreases with 6 dB per distance doubling
- Reflected sound pressure level, if diffuse, is place-independent
- Together:



Kosten's model for concert halls (ca. 1960)

- Escape from requirement to know *A*: in concert halls, all absorption is assumed to be caused by, or related to audience and musicians, i.e., proportional with 'seated surface'
- Kosten verified assumption with data from Beranek's book (1962)
- m.f (500-1000 Hz) result for halls with full audience:

α_{eq}

$$T60 = \frac{1}{6}\frac{V}{1.07 S_s}$$

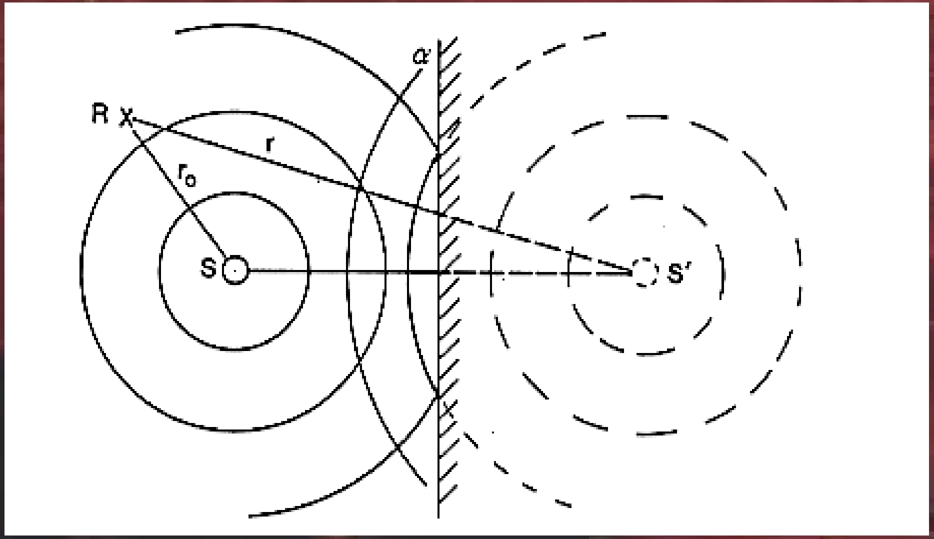
- Remember Sabine: $T60 = \frac{1}{6}\frac{V}{A} = \frac{1}{6}\frac{V}{\bar{\alpha} S}$

From global field approach to source-receiver approach

- In the 1960s, the tendency grows to consider the sound field not only in a global sense, but also related to specific source and receiver positions
- The transfer function between source and receiver is fully specified by the corresponding impulse response
- This approach asked for new models

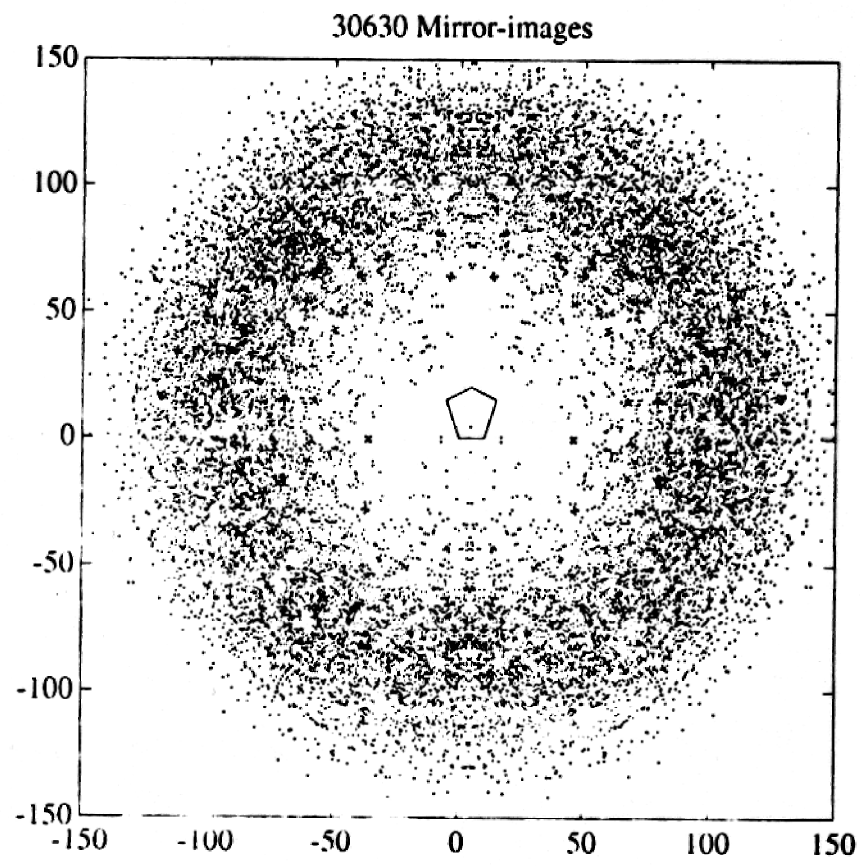
Mirror image source model (1)

- Approach to calculate impulse responses
- Reflected wave can be interpreted as the direct sound wave of a source *behind* the reflector; cf. mirror image
- In figure:



Mirror image source model (2)

- mirror image source pattern of a pentagonal 2D hall:
- gives insight in spatial distribution of reflections!



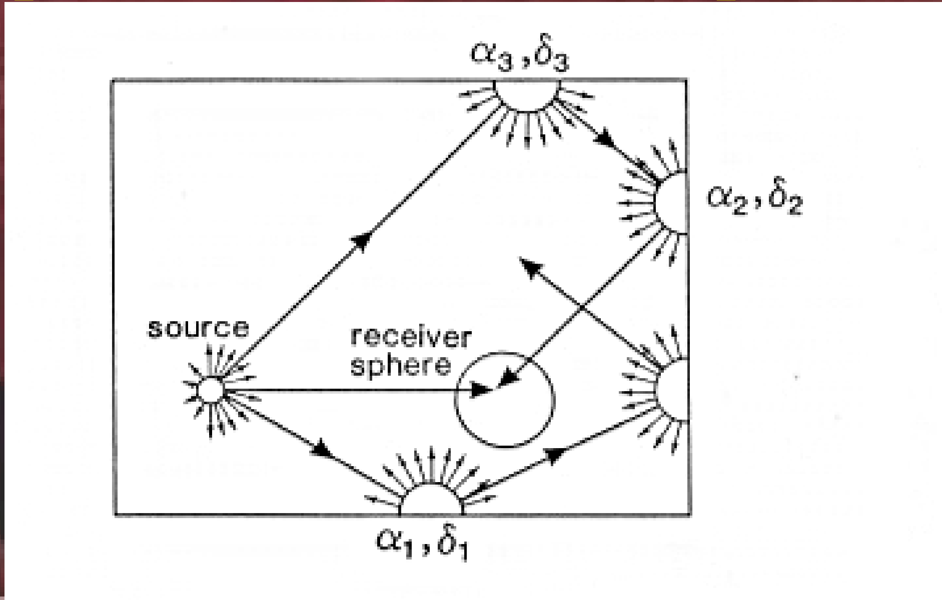
Mirror image source model (3)

- M.i.s.m. can be used to derive Eyring's reverberation time formula (already derived in 1929 in a different way) which does not require Sabine's diffusivity assumption
- Formula: $T_{60} = \dfrac{V}{6\ln(1-\bar{\alpha})^{-1}}$
- For $\bar{\alpha} \ll 1$ (diffuse!) : $\ln\left(1-\bar{\alpha}\right)^{-1} \rightarrow \bar{\alpha}$, or:

  Eyring → Sabine!

Ray tracing model (1)

- Also intended to calculate impulse responses
- wave fronts of source are ‘spatially sampled’ into rays, which are traced on their way through the room to build up the impulse response at the listener position
- in figure:



Ray tracing model (2)

- Validity of model is disputable: sound is waves, no rays – it can be seen as a h.f.-approximation → l.f. results useless?
- Receiver should have (arbitrary) non-zero dimensions, since chance that ray (= line) hits receiver point is zero…
- Only a statistical version of the impulse response can be obtained
- (Better) variant: cone tracing

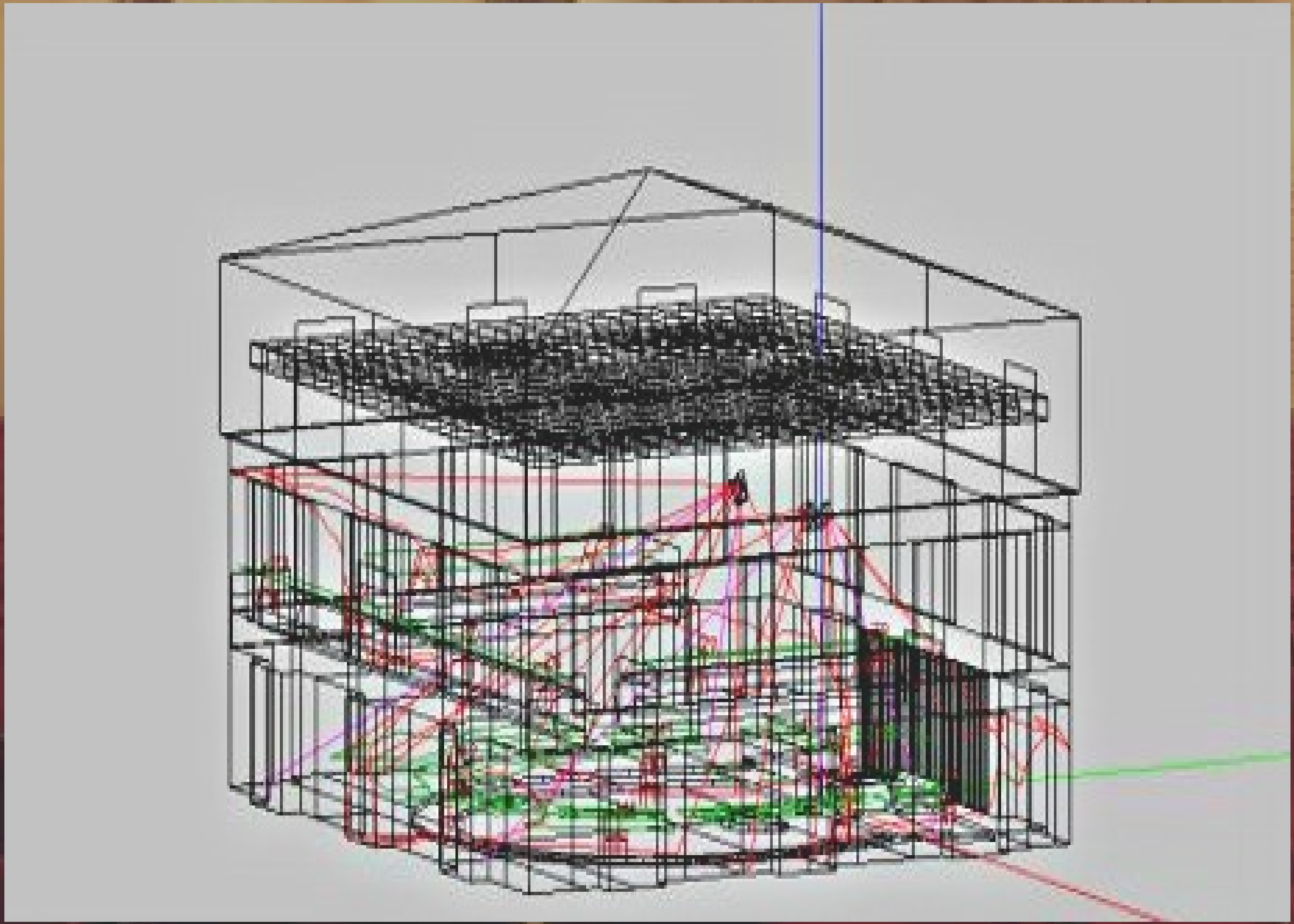
Recent developments

- Implementation of models for scattering reflection – appeared necessary after 'round robin' quality test
- Application of FEM/BEM modeling concepts
- "Full-wave modeling" (Lahivaara et al., 2008)
- …

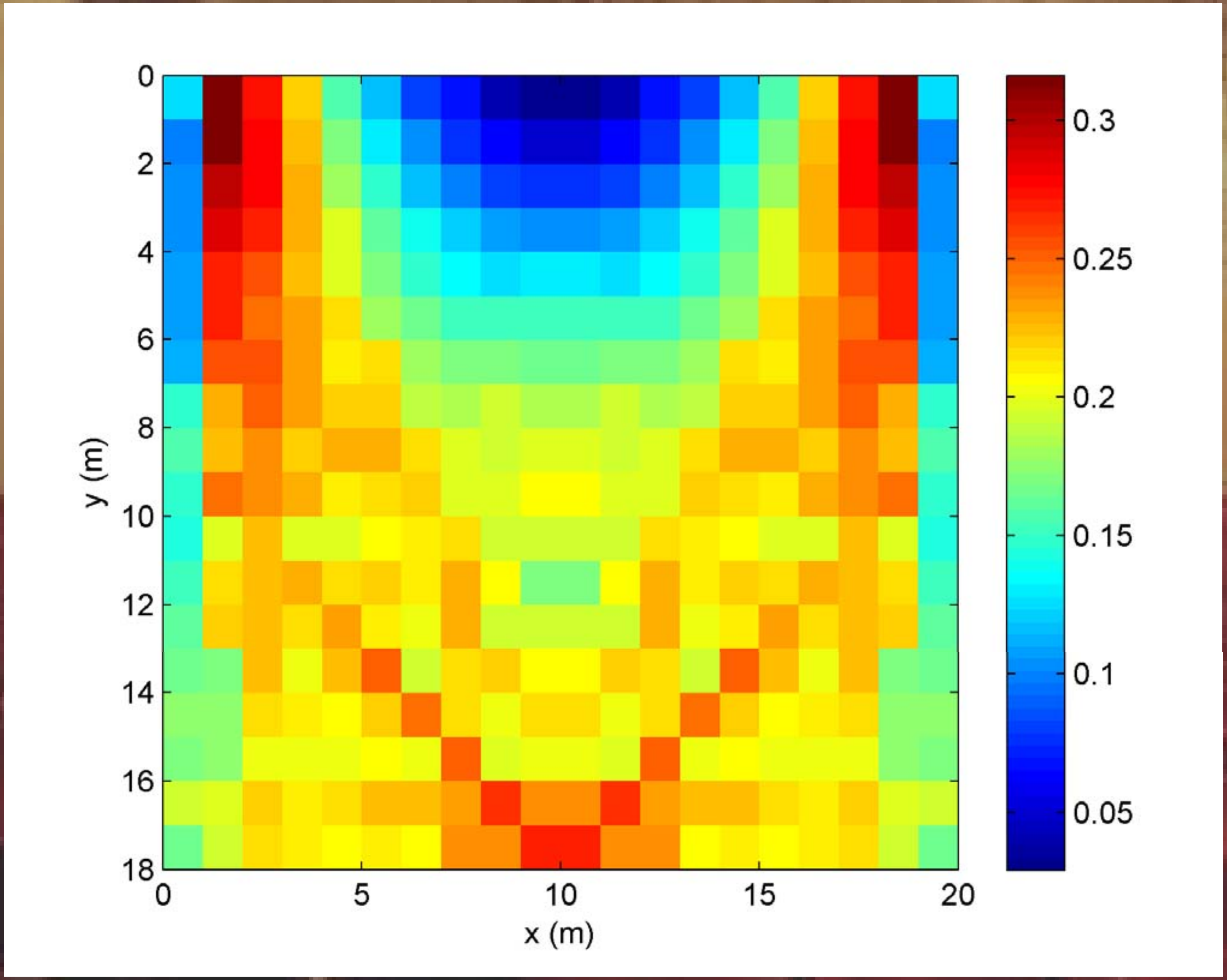
State-of-the-art

- Availability of several commercial and self-made (universities!) simulation software packages where the mentioned models are synergetically combined
- Can yield reliable results… when applied in the right way and with critical consideration!

Example of simulation output (1)



Example of simulation output (2)



LEF distribution

Perceptual quality

- When applied for auralization – making the simulated sound field audible, e.g., during the design process of a hall, the perceptual quality ('natural room sound') is of high importance
- Some examples:

measured sim 1 sim 2

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Sabine's RT measurement setup

'microphone' was Sabine's ear!

from:

Sabine, Collected Papers

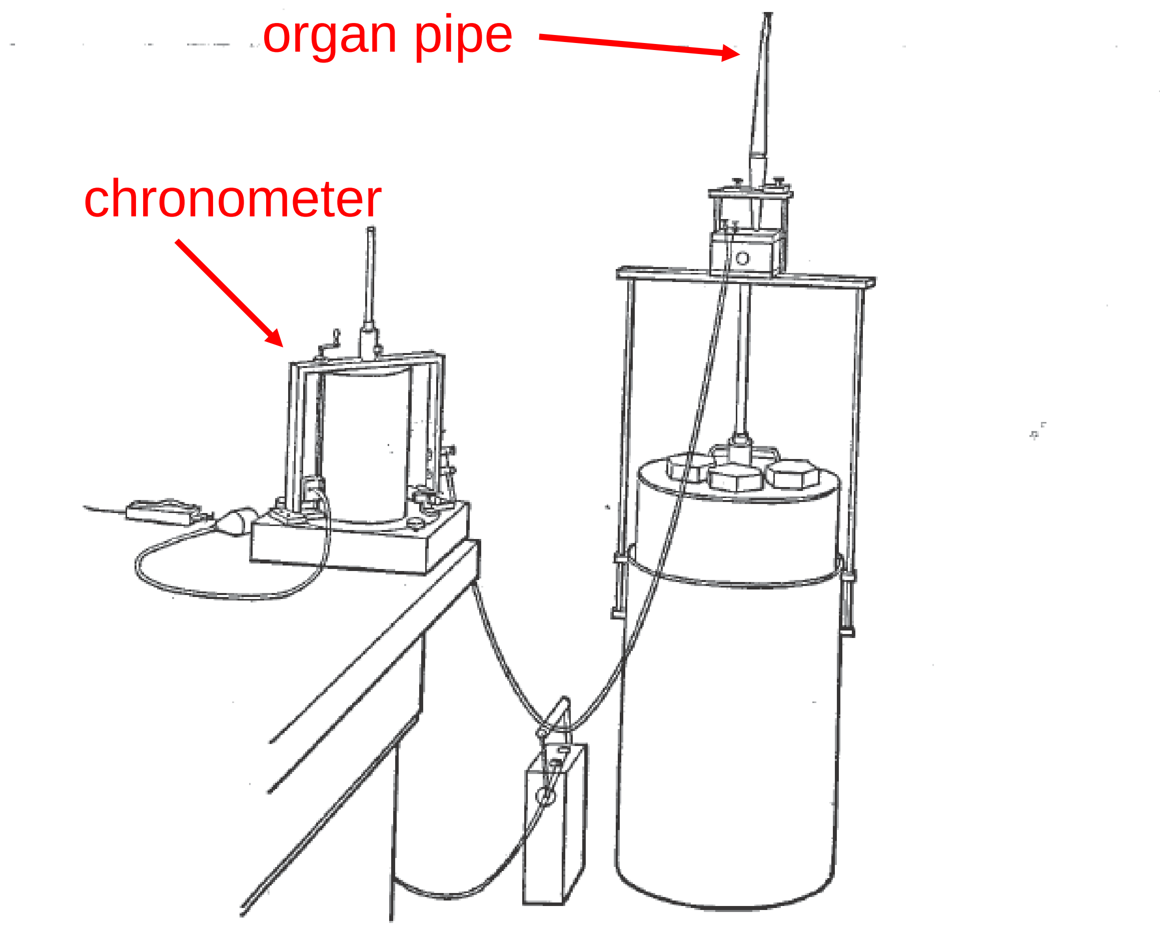


FIG. 1. Chronograph, battery, and air reservoir, the latter surmounted by the electro-pneumatic valve and organ pipe.

Until the 1960s:

- SPLs and reverberation time were the measured quantites, usually in (1/3rd) octave bands
- Noise generator (+ loudspeaker) and level recorder – fed by (filtered) microphone signal – formed the standard equipment

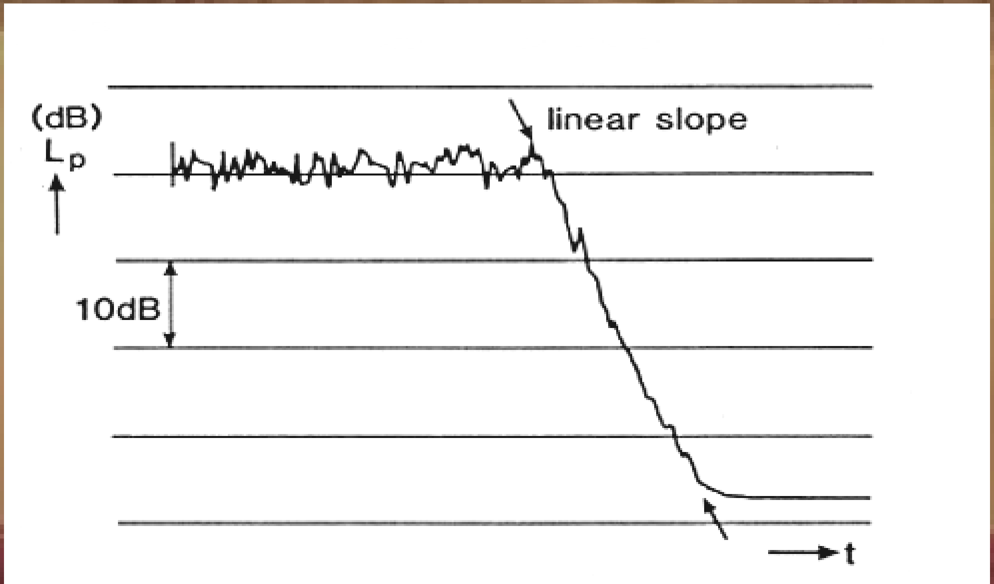
Reverberation curves

‘diffuse’ field,

$$p_{rms}^2(t) = p_{rms}^2(0)e^{-\beta t}$$

linear slope,

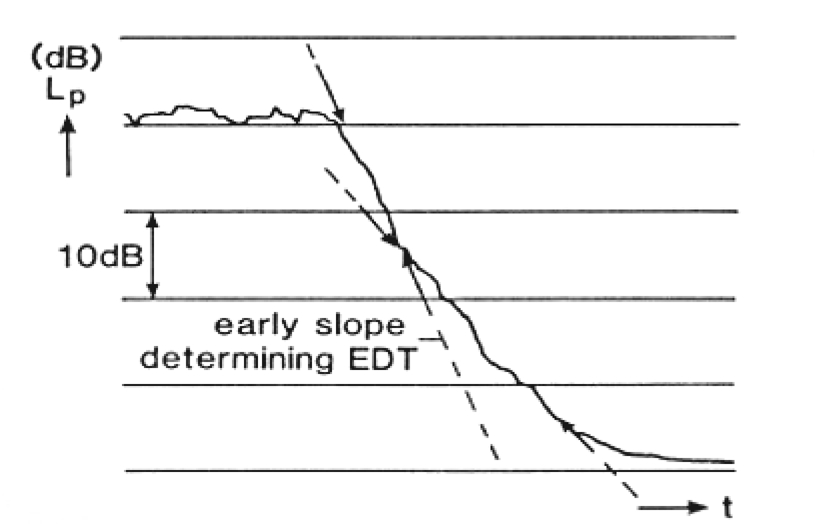
T60 = 2 T30 etc



coupled spaces,

double decay,

T60 undefined



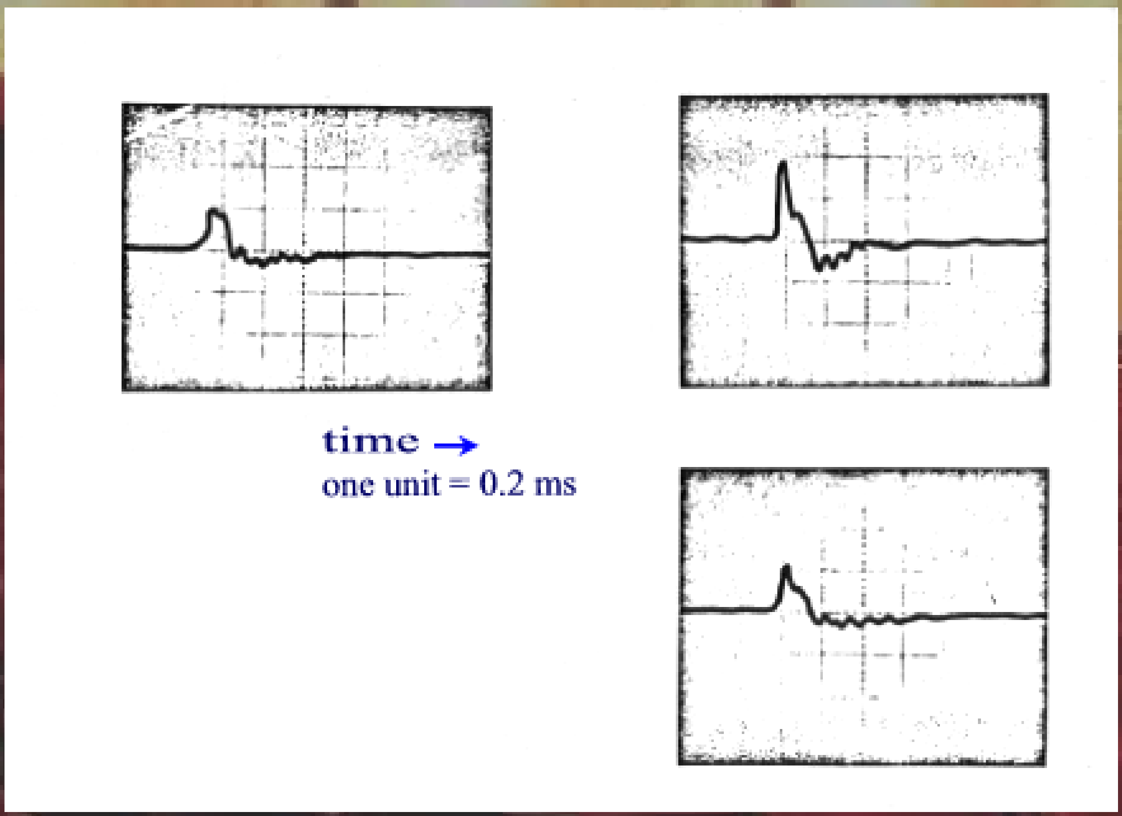
From the 1960s:

- Impulse responses had to be measured, asking for a new approach

Impulsive sources

- Requirements on the signal:
  - short (in time)
  - (→) broadband (in frequency)
  - high energy (→ high pressure at large distance)
  - omnidirectional
  - repeatable
- Candidates as (alarm-) pistols, popping balloons, pulse generator driven loudspeakers,….) all have serious shortcomings

Example: Diemer's MSc pistol

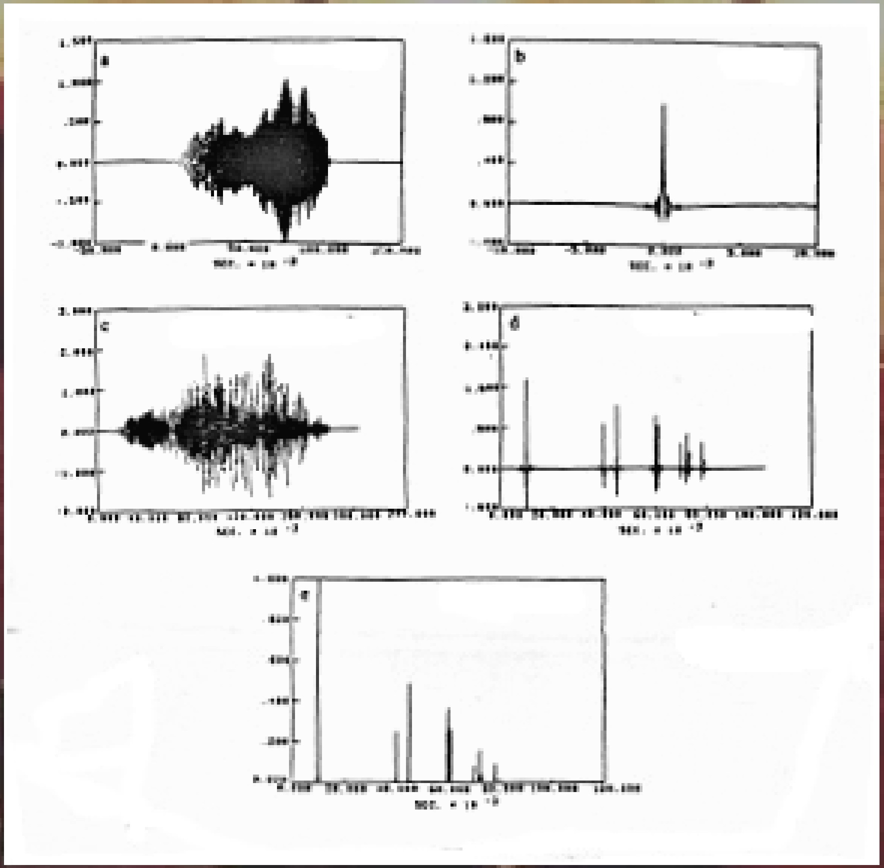


- pretty short and wide-band
- pretty rich of energy

but:

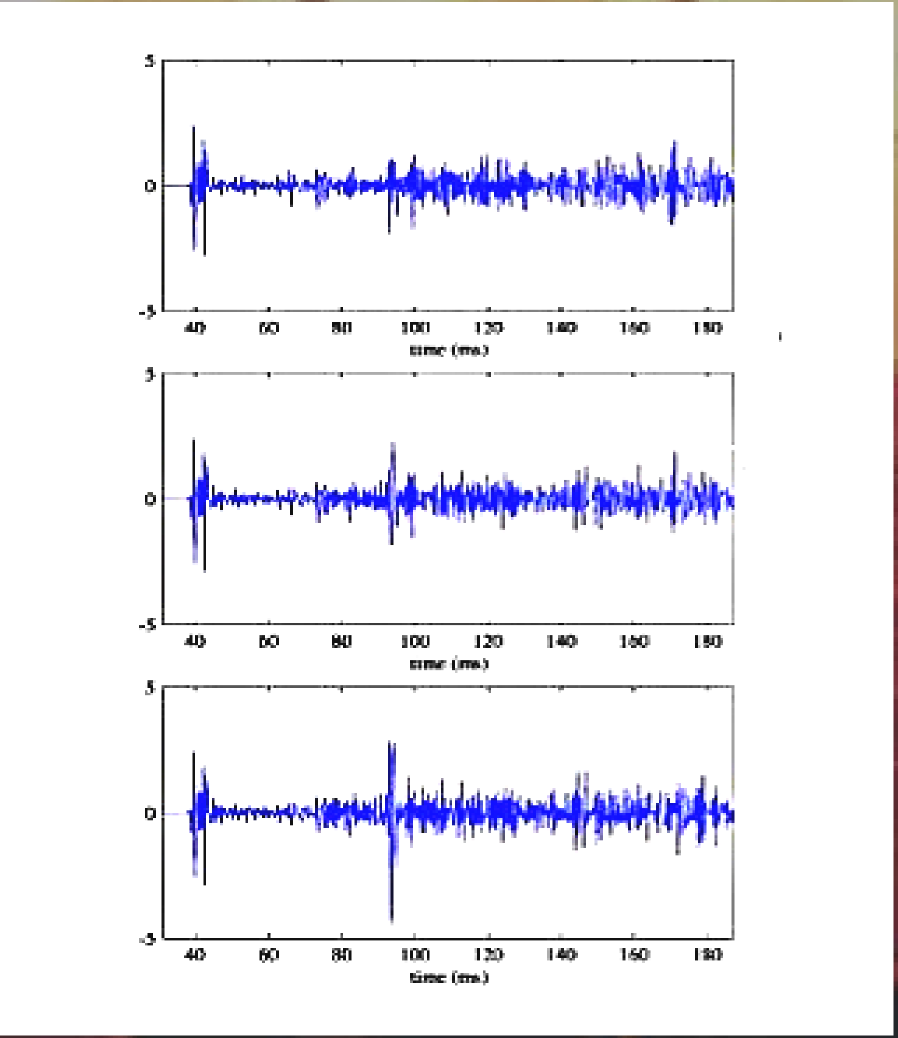
- not omni-directional
- not repeatable

Solution: inverse filtering

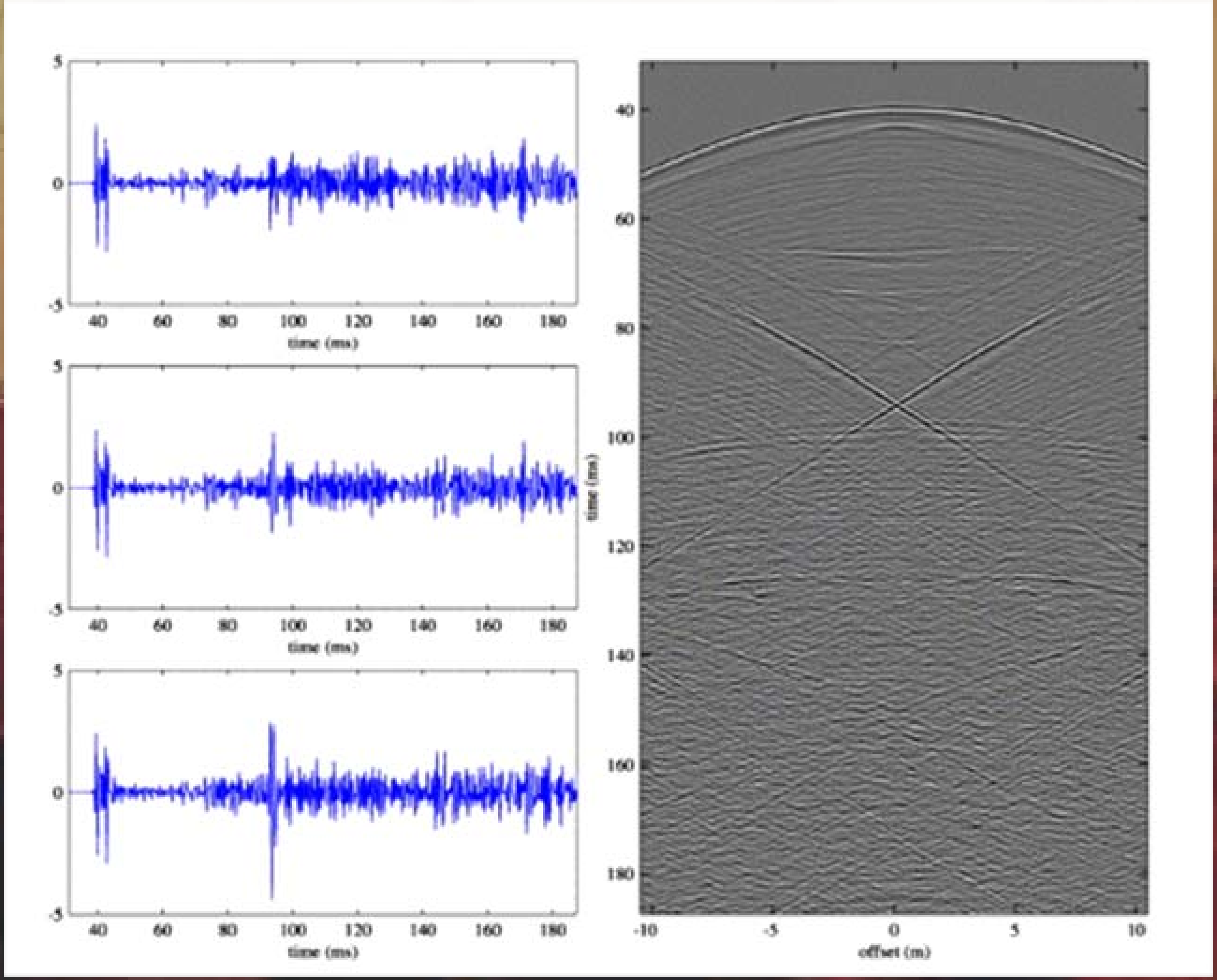


- a. sweep (or chirp) – can be any other appropriate signal
- b. deconvolved sweep
- c. sweep response
- d. deconvolved sweep response
- e. squared version of d

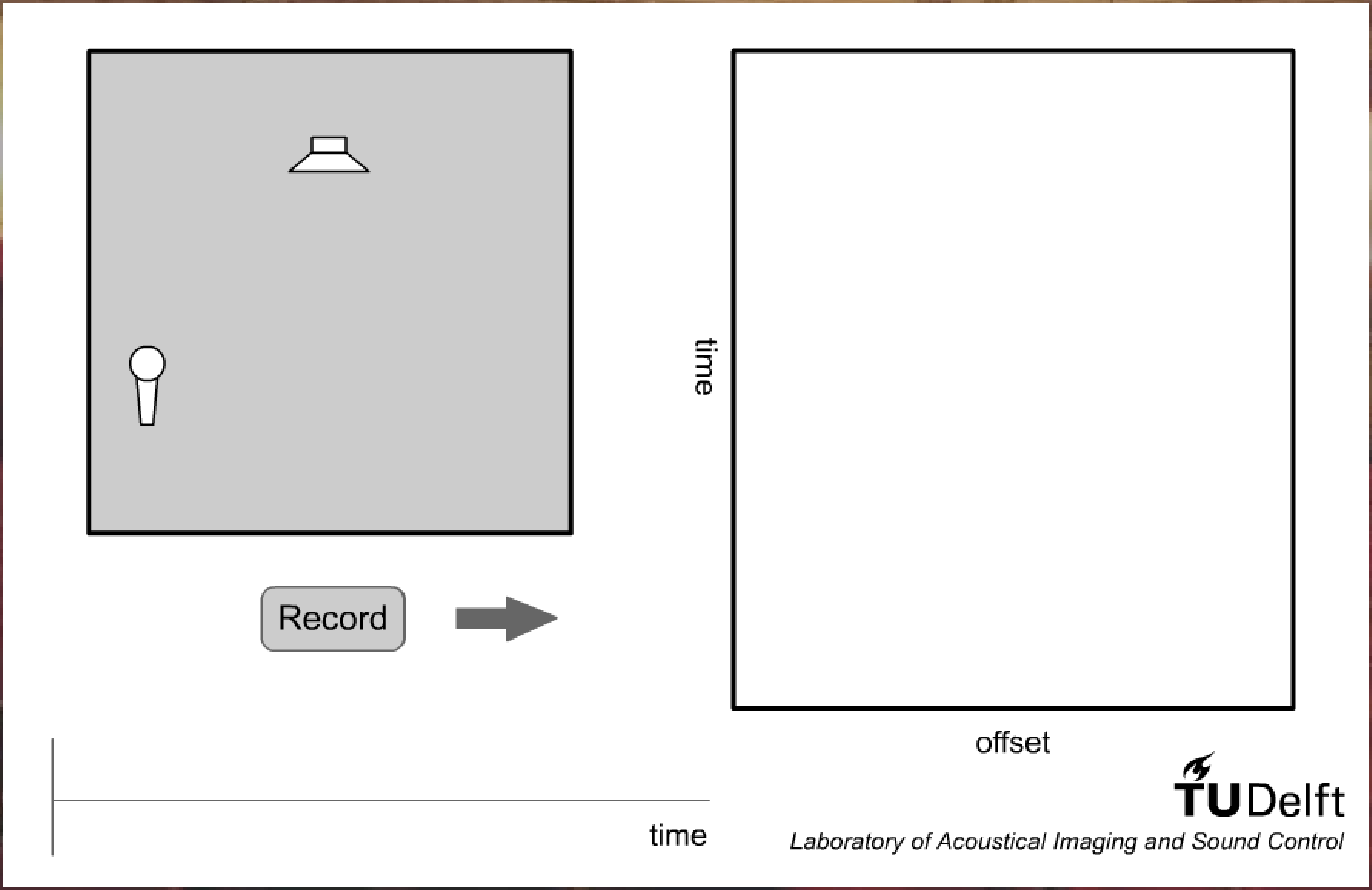
Individual impulse responses: difficult to interpret



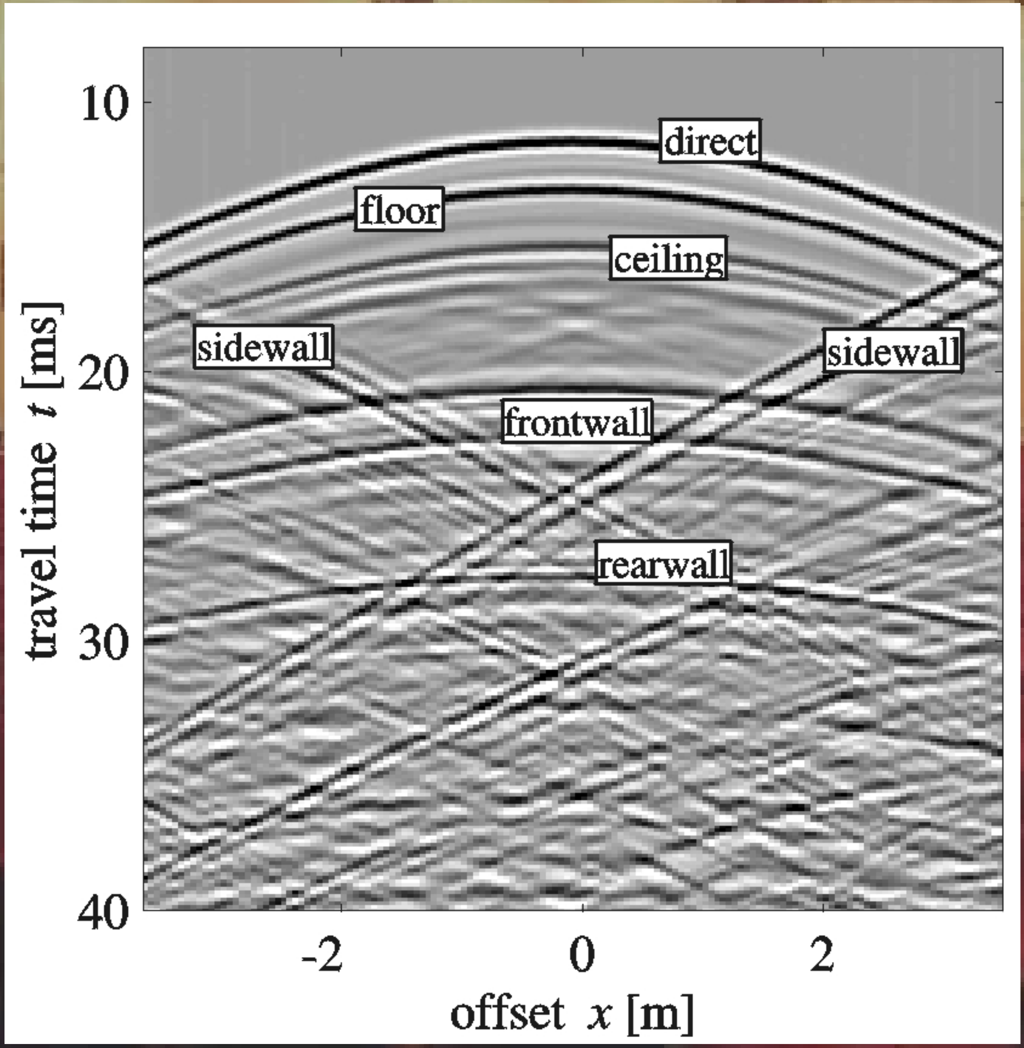
Eyeopener: multi-trace i.r. measured along mic array



Array Recording



First attempt (1996) in a small TUDelft lecture hall

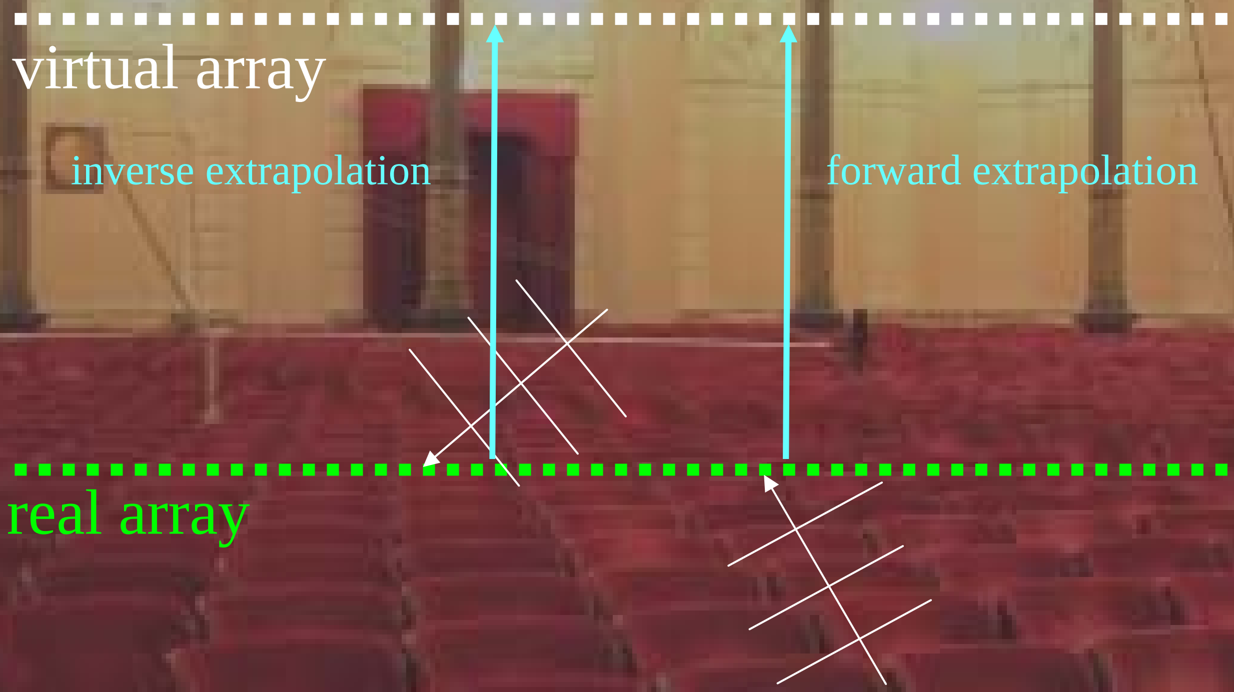


Concertgebouw measurements

Plane wave decomposition and extrapolation

- Each sound field can be decomposed into plane waves by spatial Fourier Transform or Radon Transform (beyond scope of this presentation)
- Plane waves can easily be extrapolated (only phase shift)
- This way, with some approximations, *measurements at other array positions can be simulated* → one array measurement gives data for complete hall!

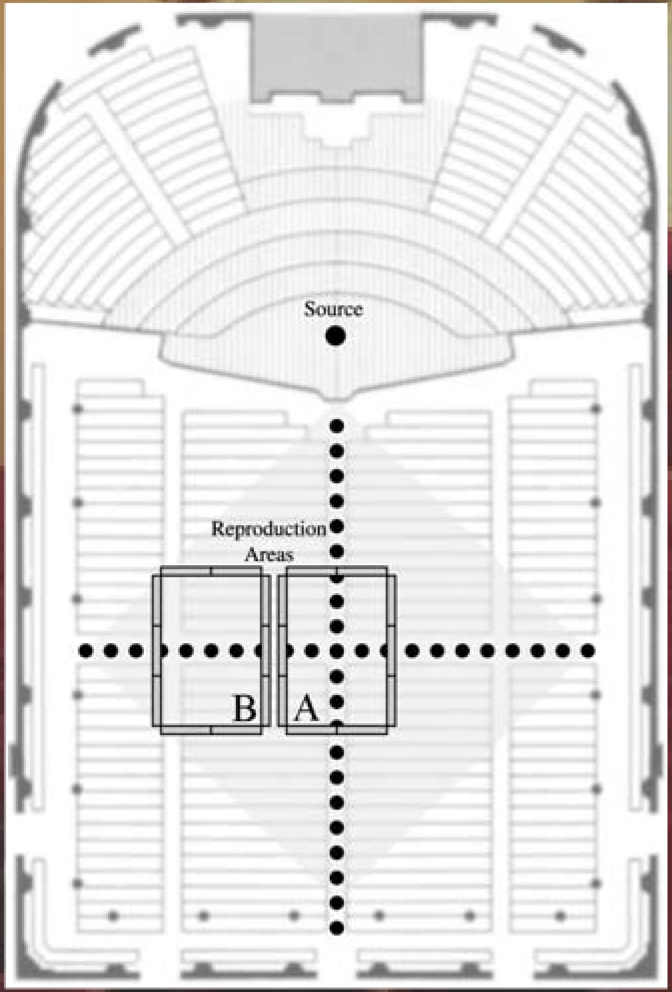
Extrapolation of plane wave components



Extrapolation accuracy

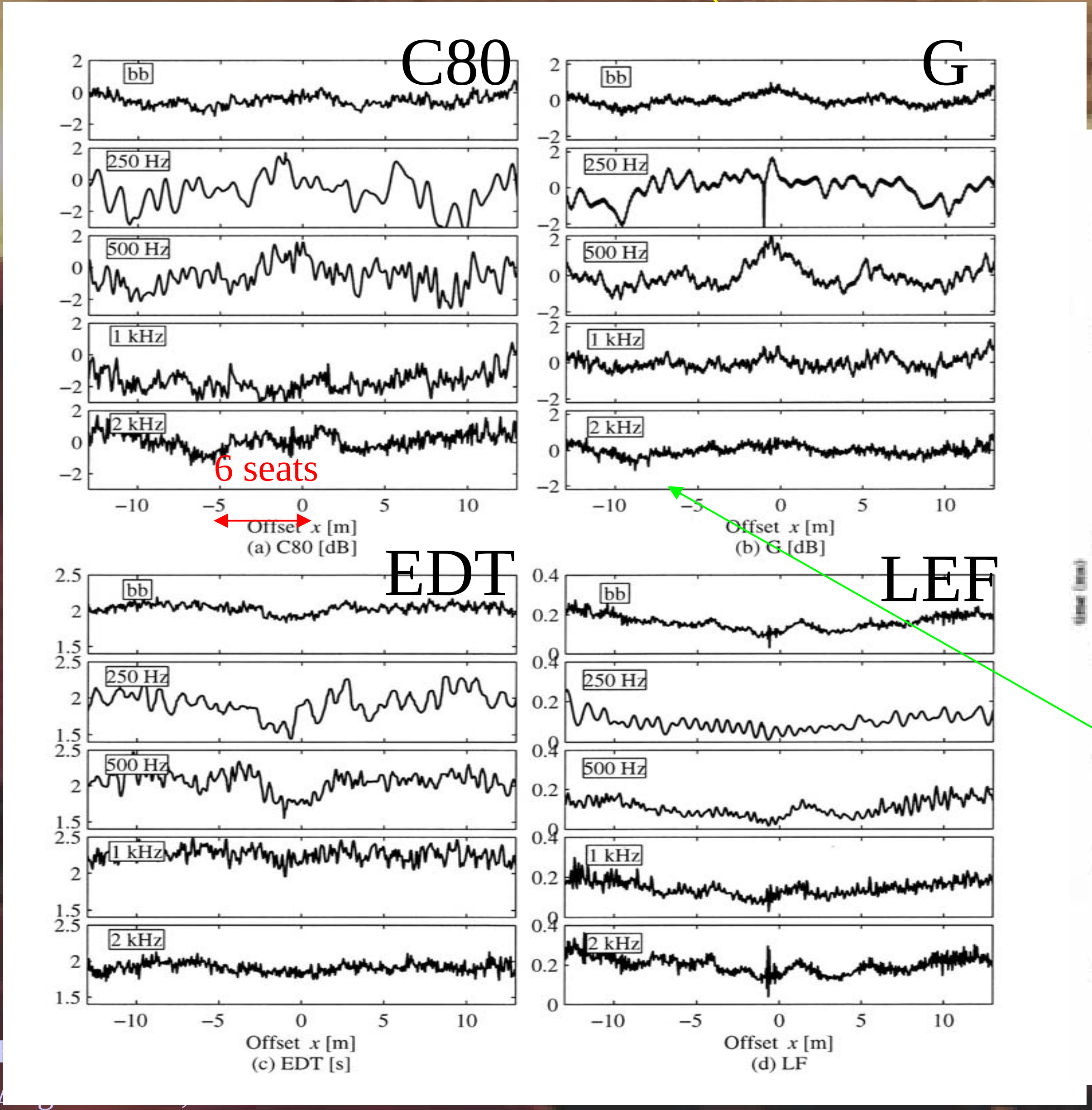
- Problem: extrapolation is *less accurate* for plane wave components with *grazing* incidence on the array
- Solution: measure with *cross-shaped* array: components with grazing incidence on one cross leg have close-to-normal incidence to the other one – and vice versa.

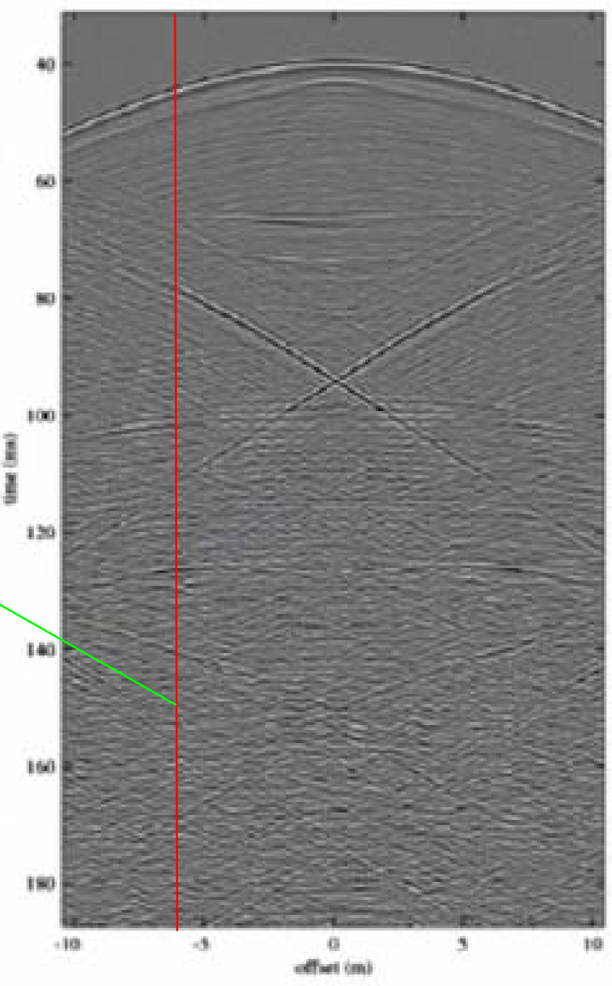
Cross-shaped array measurements Concertgebouw Amsterdam



Extrapolation to different rectangular arrays for reproduction by (old) TU Delft WFS system

Calculation of parameters from array measurements (Concert hall “De Doelen”, Rotterdam)





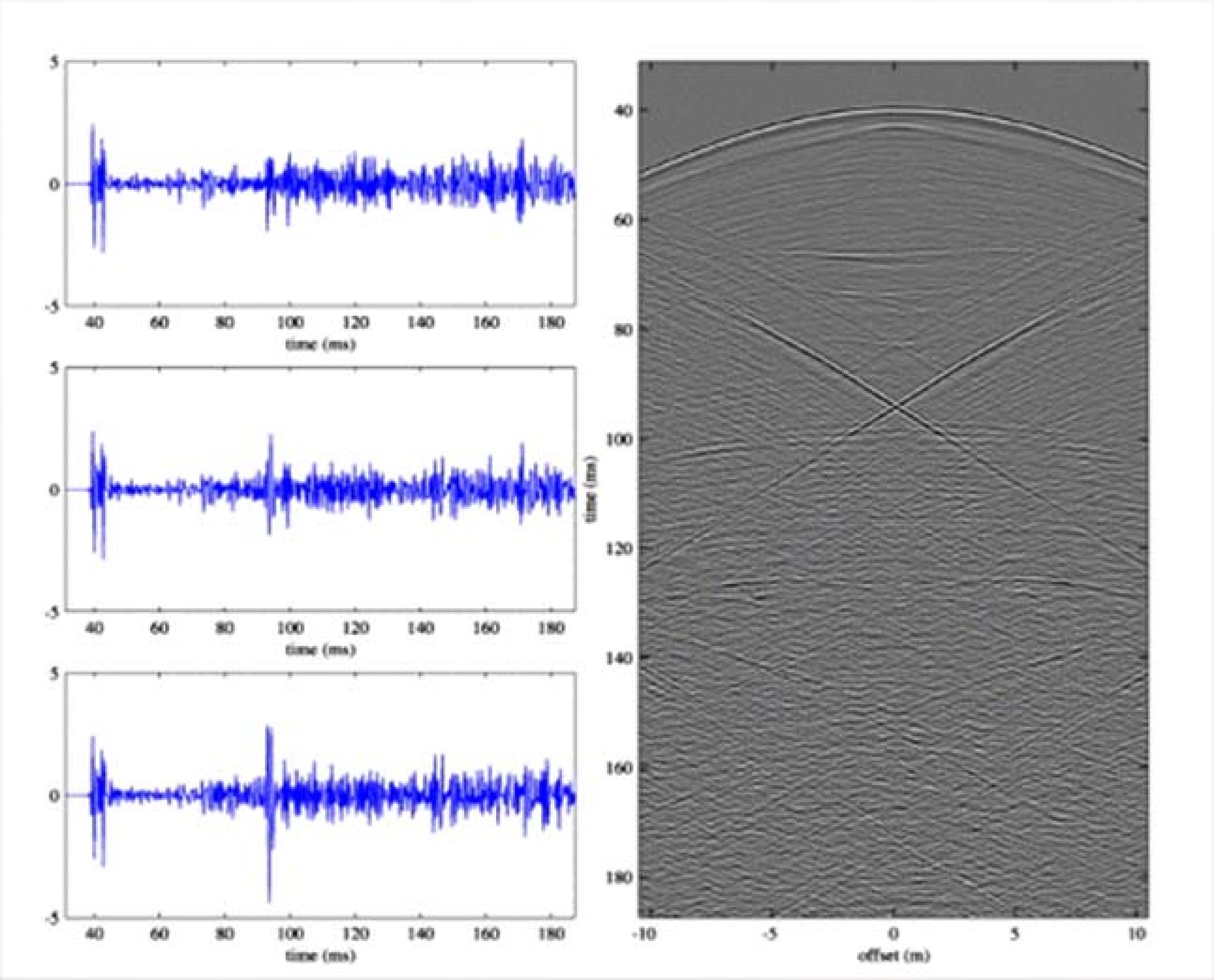
We see:

- Array measurements show that all room acoustic parameters heavily fluctuate with small spatial variation (up to factor 2 in front of the same seat!)
- This does not correspond with human acoustical perception
- So: predicting perception from a local measurement is ‘gambling’

How come?

- Measured i.r. show interference on small spatial scale, to which human listeners – with two ears! – are not sensitive

Wave interference



How come?

- Measured i.r. show interference on small spatial scale, to which human listeners – with two ears! – are not sensitive
- Parameters appear to be very sensititive to noise: when i.r. are measured with high signal-to-noise ratio, fluctuations decrease for some parameters

Remedies (research in progress at TU Delft)

- Measure impulse responses with high S?N-ratio (> 60 dB!)
- Process measured i.r. such that parameters correspond with human perception:

  - remove interference

  - apply filter to i.r. which takes perception mechanism (masking etc.) into account

  - average parameter values over space

  - …..

Literature:

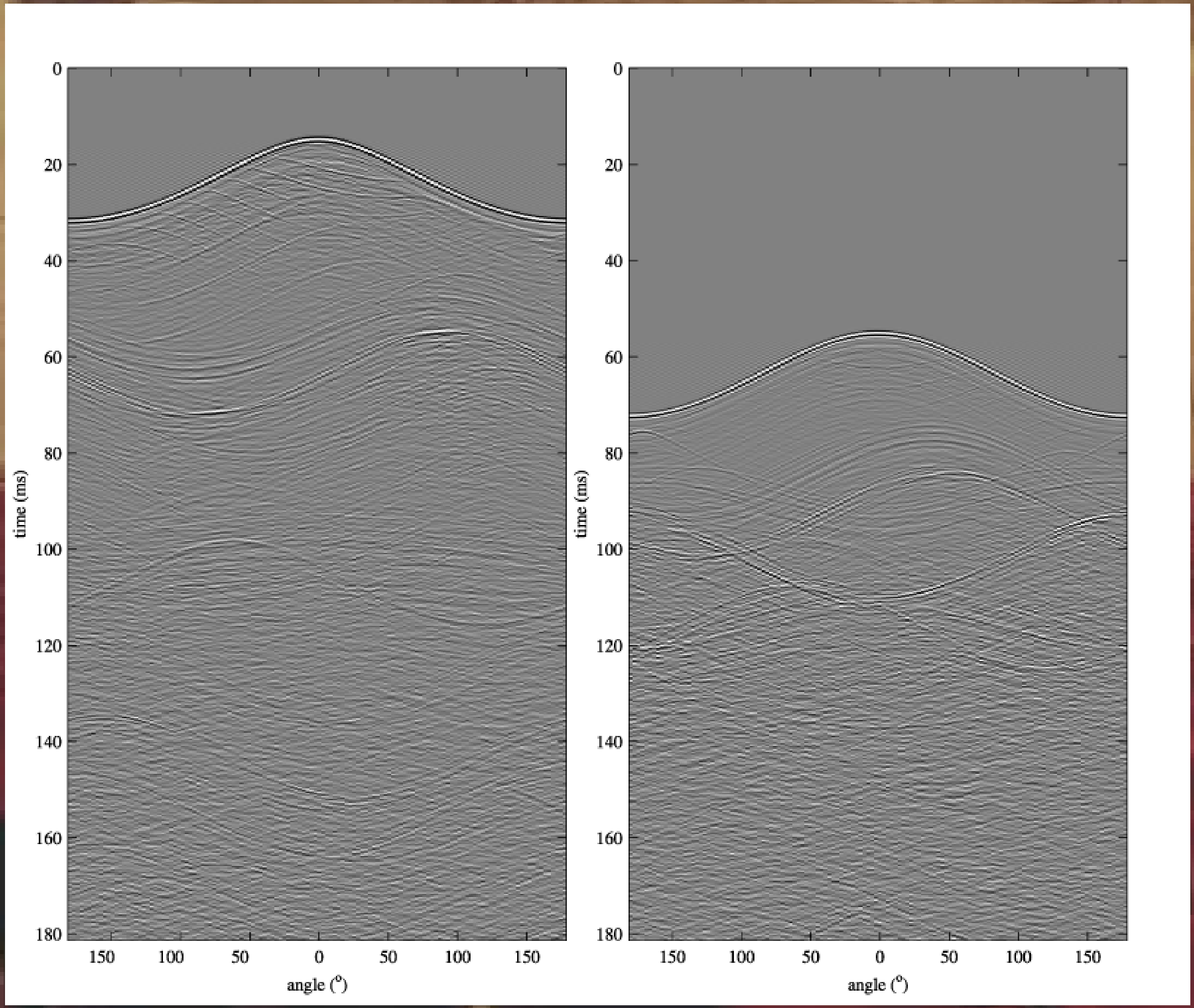
Diemer de Vries, Edo M. Hulsebos, and Jan Baan, “Spatial fluctuations in measures for spaciousness”, J. Acoust. Soc. Am. Vol. 110, pp. 947-954, Aug. 2001.

From linear to circular array

- Problem: linear array measurement is rather time consuming (several hours for 1 source position)
- Solution (Dr. Edo Hulsebos): measurement along a circular array of positions (1 microphone on a rod rotating on a turntable) is much faster (12 min. for 1 source position) and provides (almost) same features

Concertgebouw measurements

Circular array measurement results

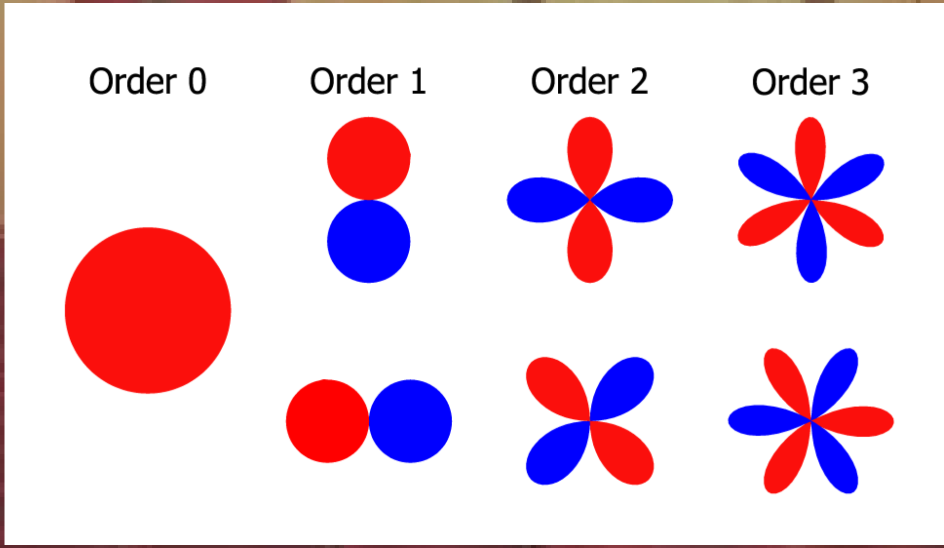


Extrapolation of circular array data

- For extrapolation purposes, the measured data can be decomposed into plane waves via decomposition into *cylindrical harmonics*

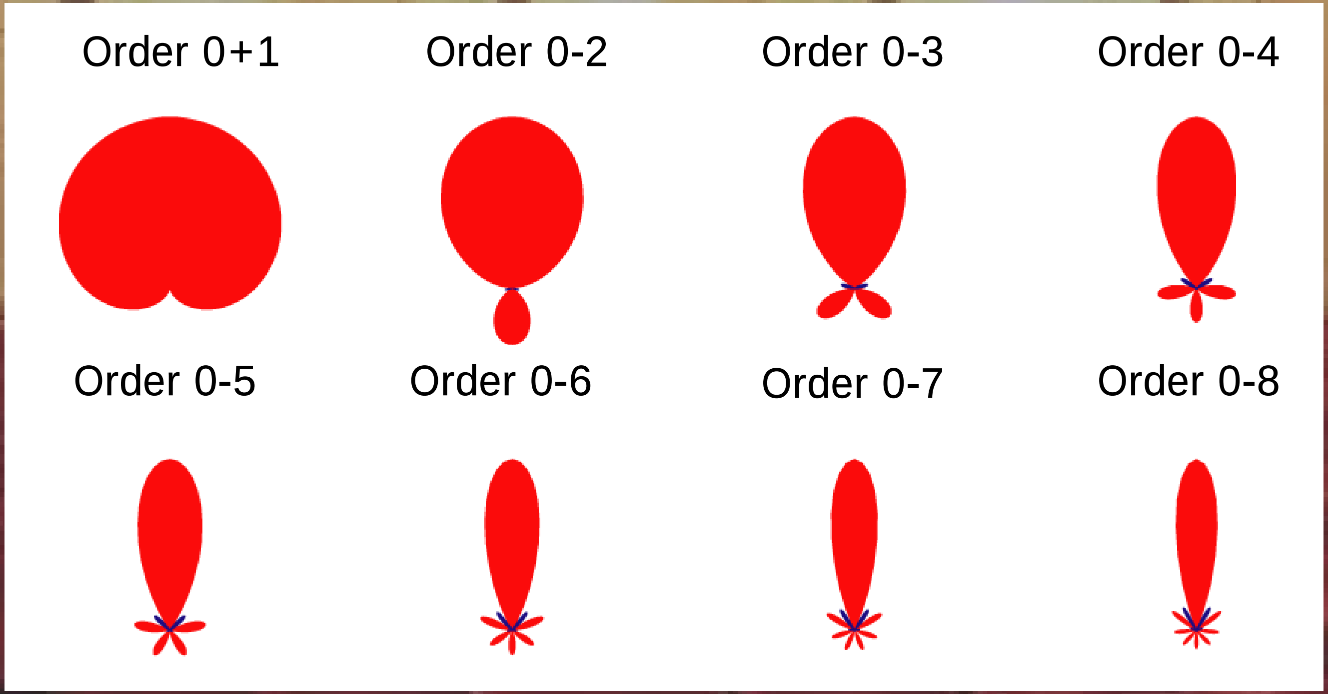
Cylindrical harmonics

- They represent a certain directivity pattern: 0th order is monopole, 1st order is dipole, etc.
- In figure:



- They can be combined to simulate a microphone with any directivity!

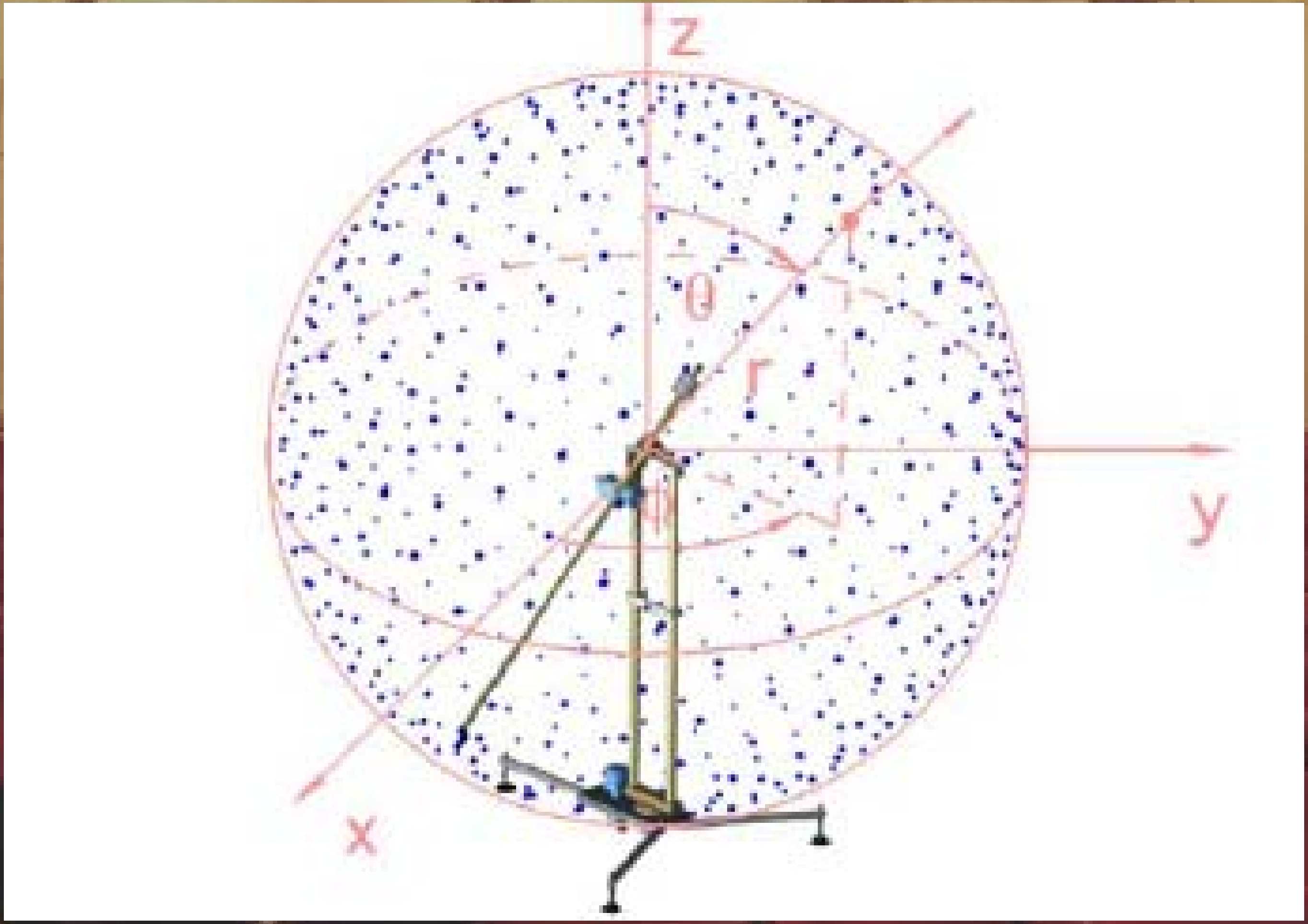
Cylindrical harmonics and plane wave decomposition

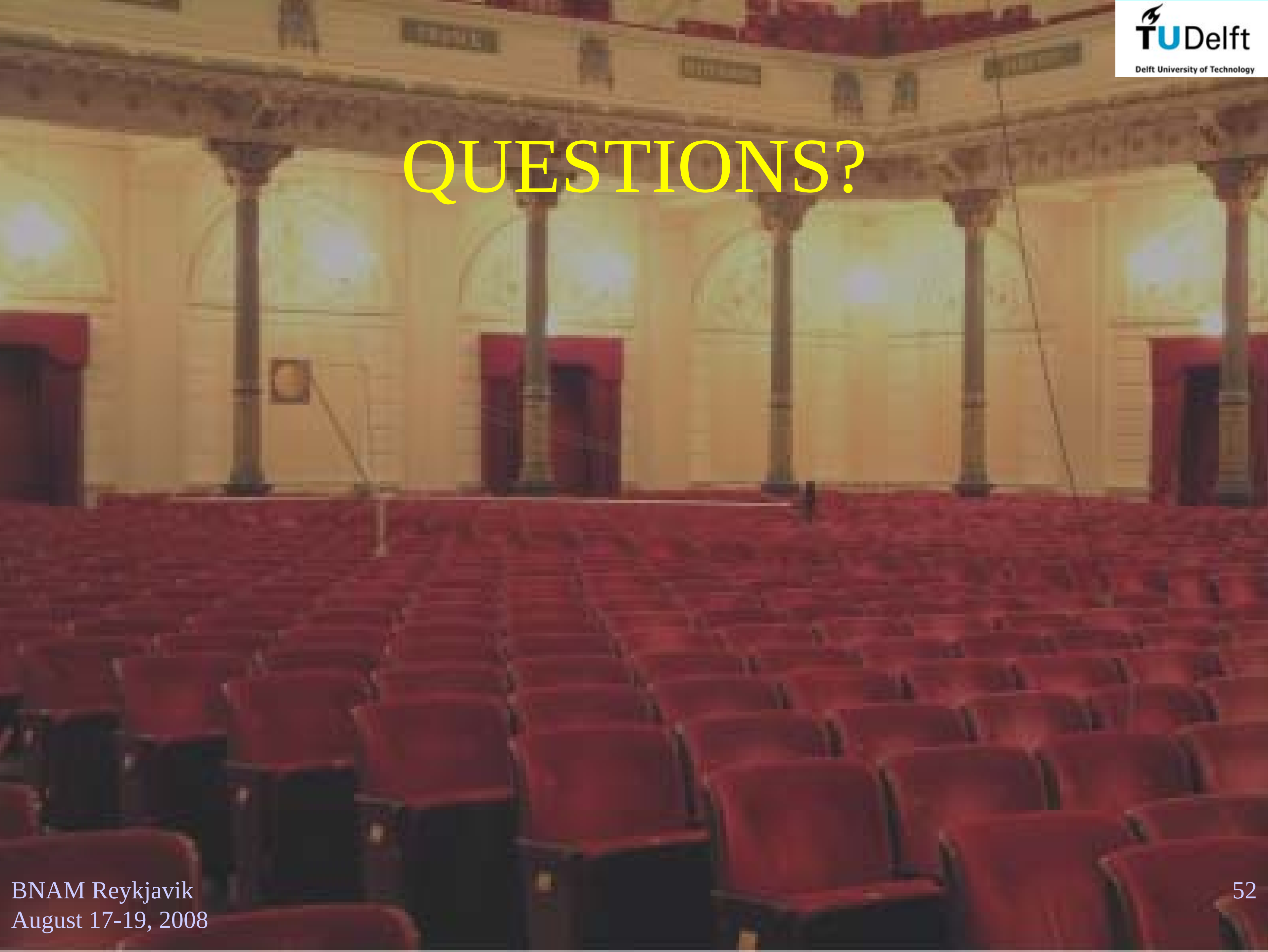


Logical next step…

- … a spherical array!
- 3D processing using spherical harmonics
- Research in progress at TU Delft in cooperation with TU Ilmenau

In figure:





QUESTIONS?