



Spatial acquisition, digital archiving, and interactive auralization of church acoustics

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Toon van Waterschoot

STADIUS Center for Dynamical Systems, Signal
Processing, and Data Analytics

ESAT – Department of Electrical Engineering
KU Leuven, Belgium



Context & Motivation



Ghent Altarpiece (J. & H. van Eyck, 1432)

Flemish primitives

Context & Motivation

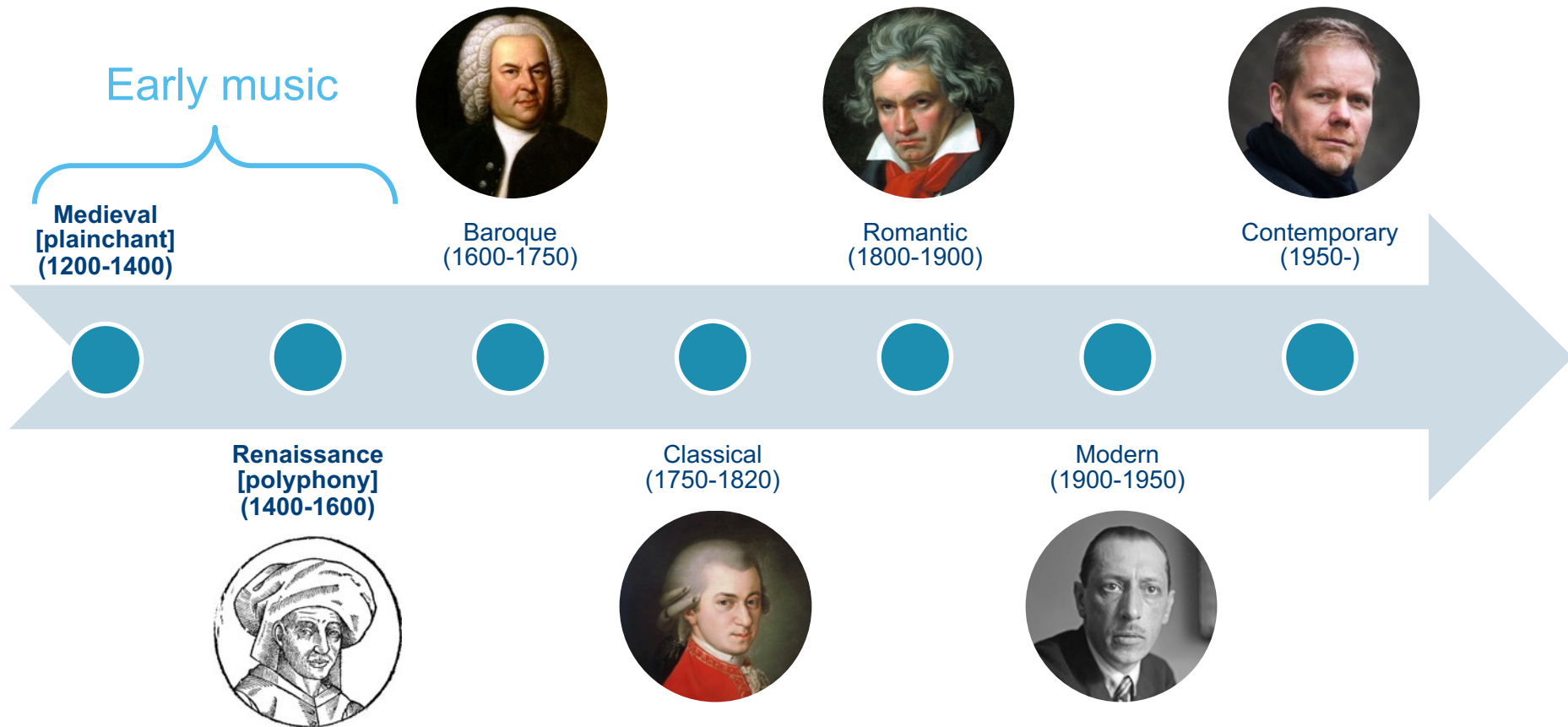


Nymphes des bois (J. des Prez, 1497, performed by Capella Pratensis)

Franco-Flemish polyphony

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Context & Motivation



A concise timeline of Western music

Context & Motivation

Early Music Research

Source



Performer



Environment



Context & Motivation

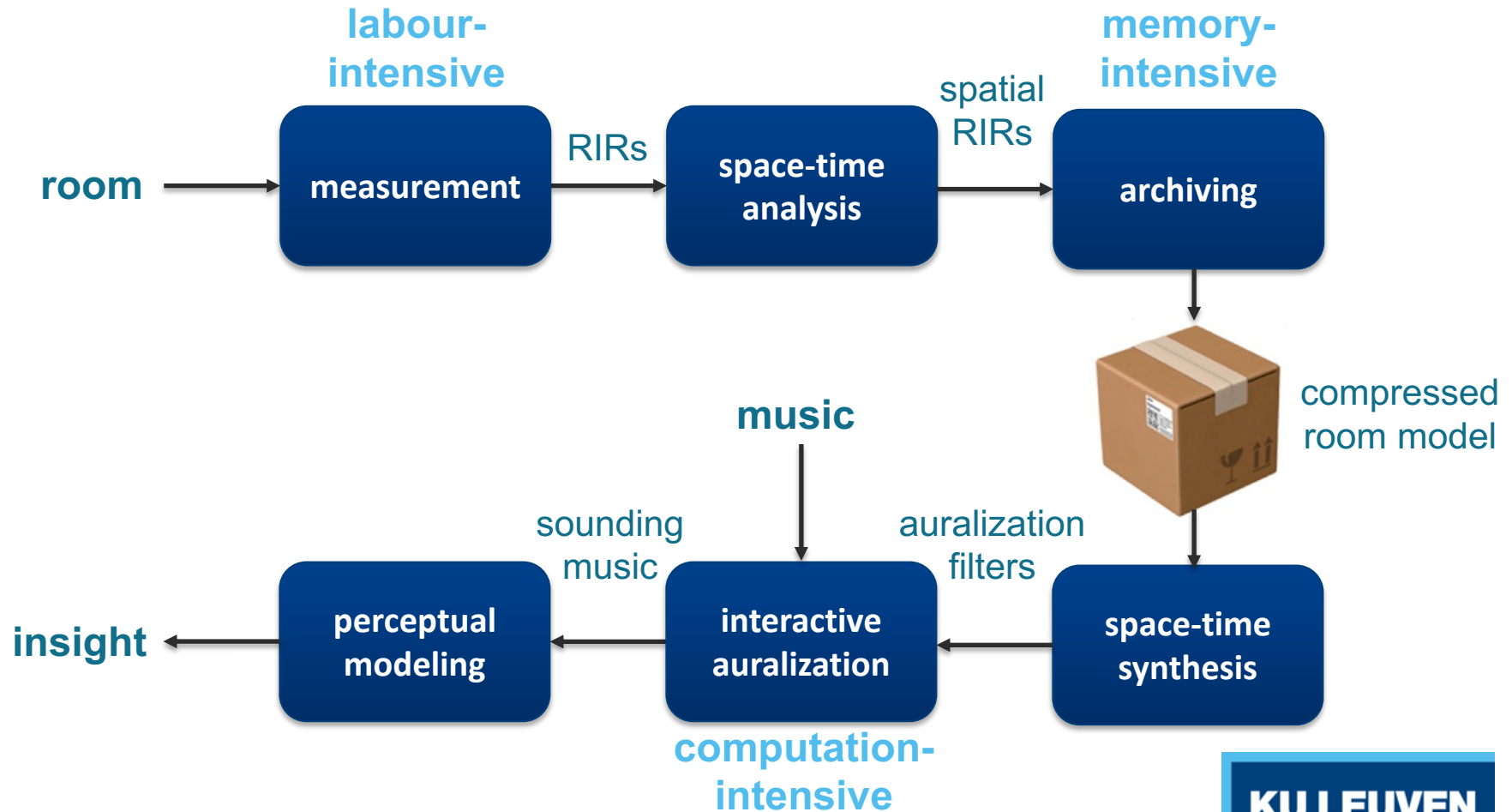


- **Research questions**

- How to **capture** the acoustics of a space to auralize/reproduce it at a different time and place?
- How to **archive** the acoustics of a historically valuable space for later reuse?
- How to achieve **interactive auralization** in a flexible and ecological manner?
- How to increase **understanding** of the relevance and quality of the acoustic characteristics of a space for musical performance?

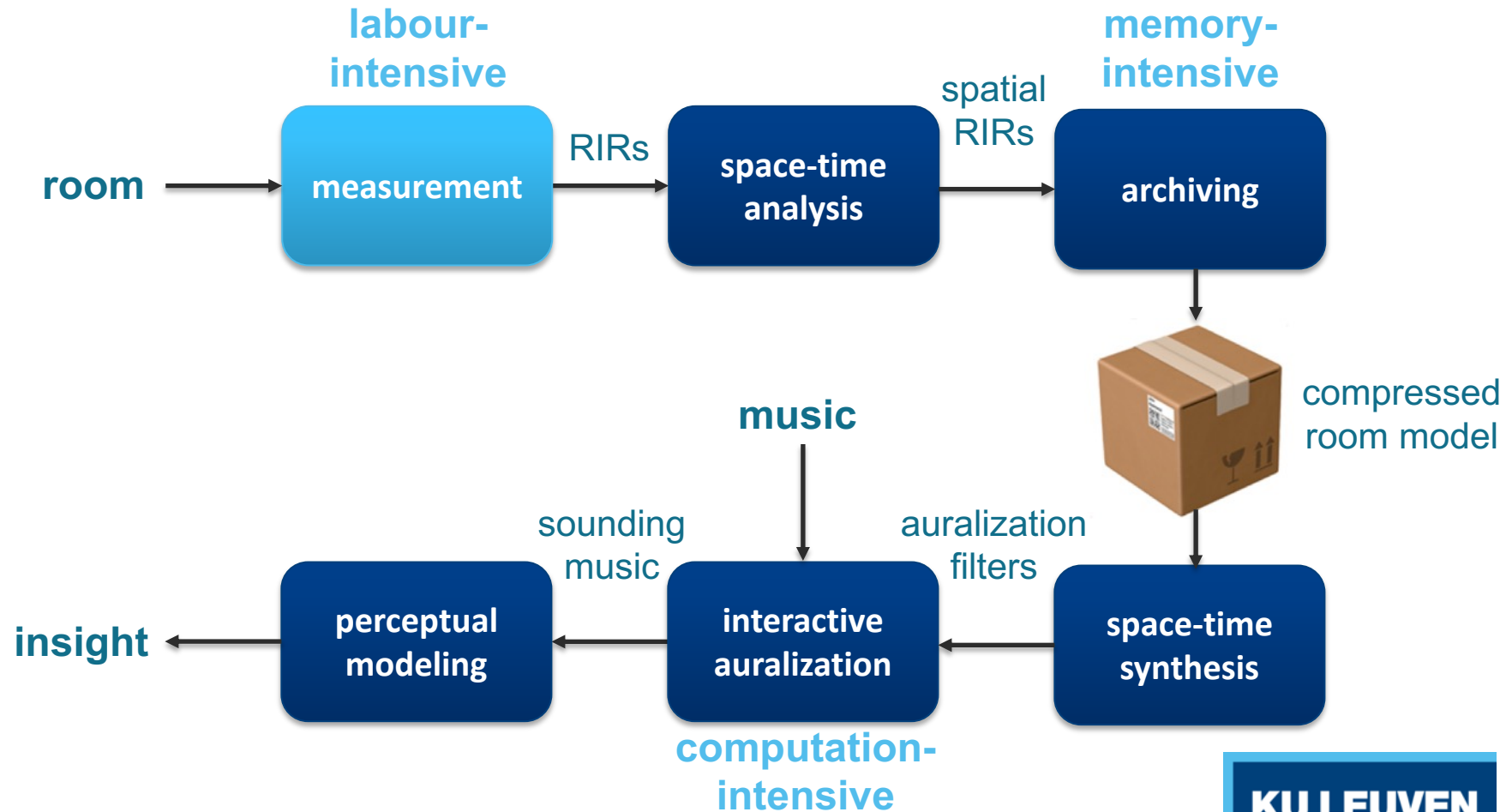
Outline

- System overview & challenges



Outline

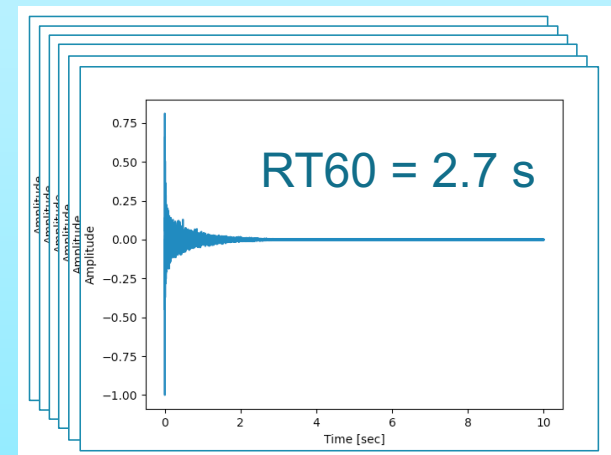
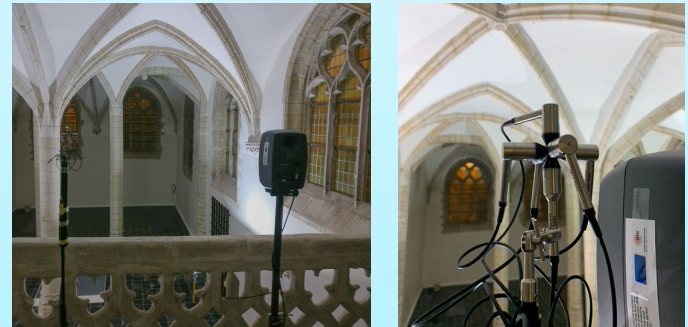
- **System overview & challenges**



Measurement

- **RIR measurement**
 - = well-established procedure
 - **Challenges:**
 - (Nightly) access to venue
 - Ownership of measurements
 - Labour-intensive
 - RIR = point-to-point model
- *how to choose source/mic positions?*

Case study: Nassau Chapel, Royal Library Brussels



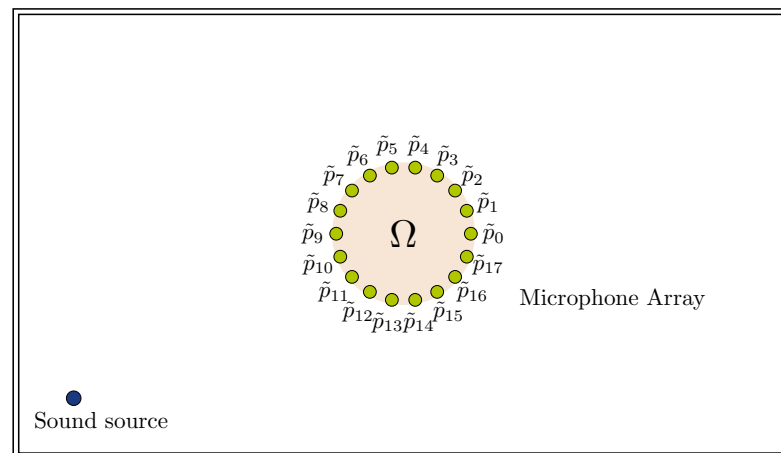
Measurement

- **Key idea: RIR measurement at “few” positions + RIR interpolation**
 - Reduce labor-intensity
 - Expand potential use to other positions than those measured
 - Reduce memory required for archiving
 - Allow for auralization with moving sources/listeners
- **RIR interpolation problem:**

given RIR measurements

$$\tilde{\mathbf{P}} = [\tilde{\mathbf{p}}_0 \dots \tilde{\mathbf{p}}_{N_m-1}]$$

find RIR for same source
at any mic position $\mathbf{r}$ in Ω

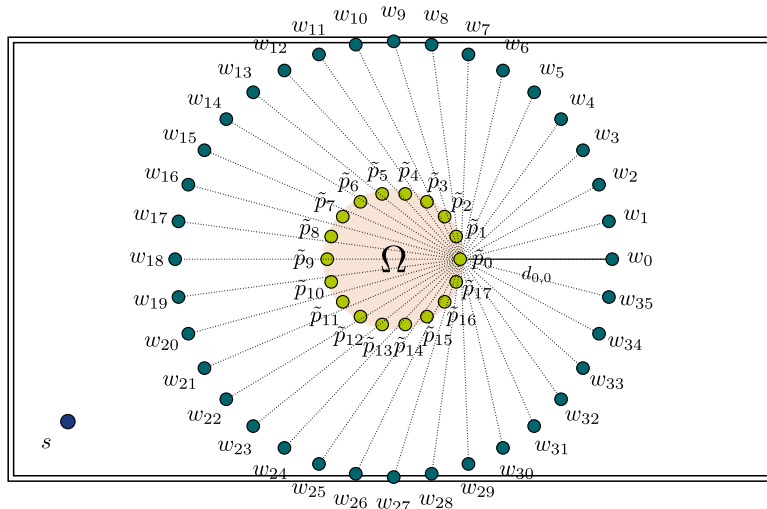


Measurement

- **Time-domain equivalent source method (TESM)**
 - Model RIR $h(t, \mathbf{r})$ for mic position $\mathbf{r}$ in Ω as sum of propagating equivalent source signals $w_l(t) \neq \delta(t)$

$$h(t, \mathbf{r}) = \sum_{l=0}^{N_w-1} \frac{1}{4\pi d_l(\mathbf{r})} \delta(t - d_l(\mathbf{r})/c) * w_l(t) = \mathcal{D}(\mathbf{r})[\mathbf{W}]$$

independent of $\mathbf{r}$!



$$\mathbf{W} = [\mathbf{w}_0 \dots \mathbf{w}_{N_w-1}]$$

$\mathcal{D}(\mathbf{r})[\cdot]$ = linear operator

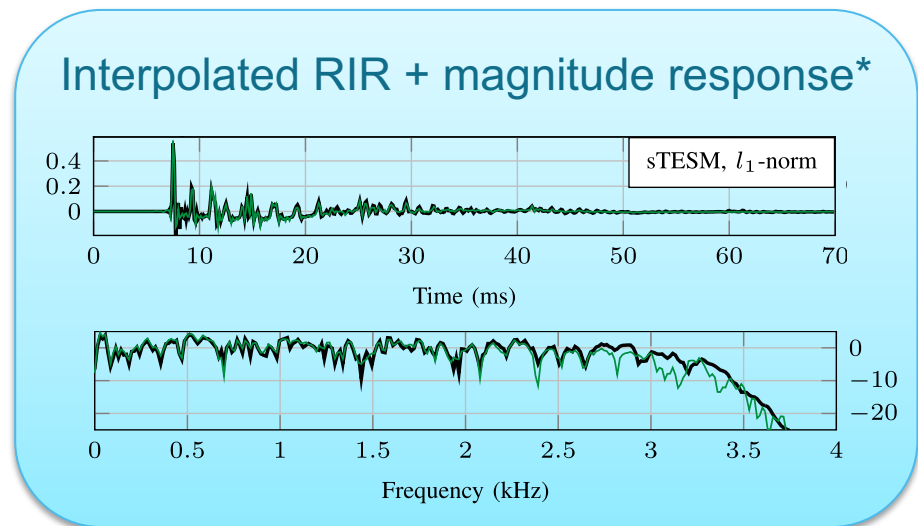
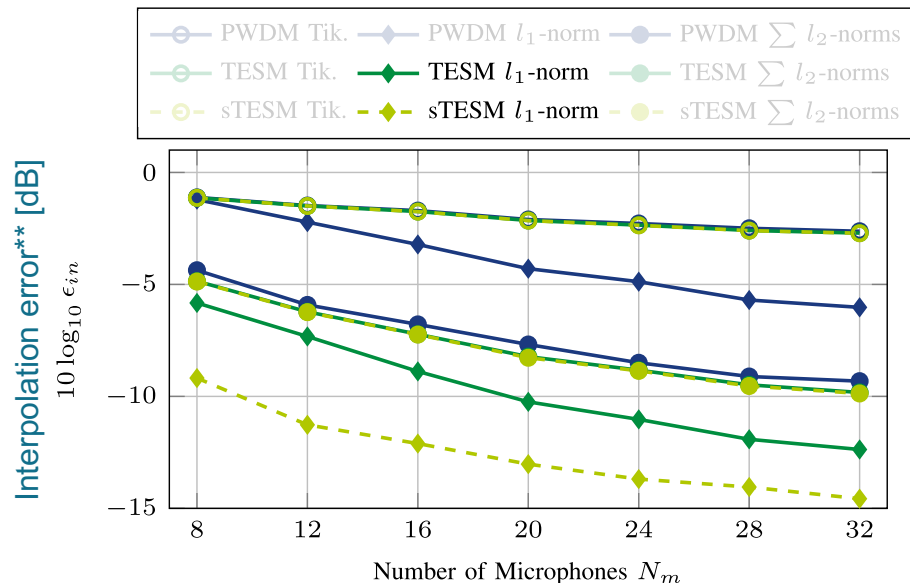
(delays + attenuations)

Measurement

- **Time-domain equivalent source method (TESM)**

- Finding TESM signals $w_l(t)$ by matching pressure signals $\tilde{p}_m(t)$ (= measured RIR) at microphones = **underdetermined problem**
- Regularization by **imposing spatio-temporal sparsity**

$$\min_{\mathbf{W}} \|D(\mathbf{W}) - \tilde{\mathbf{P}}\|_F^2 + \lambda \|\text{vec}(\mathbf{W})\|_1$$



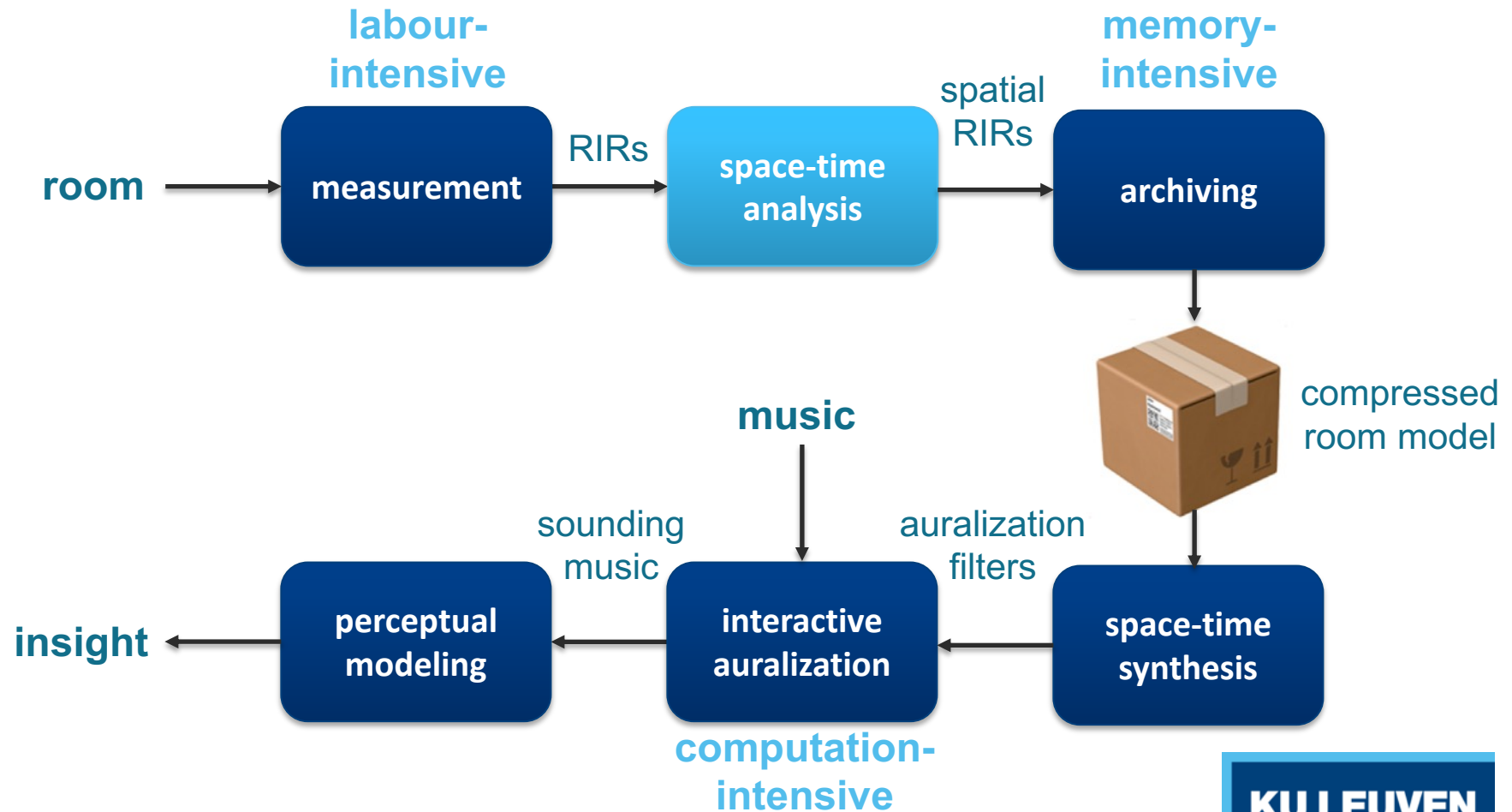
*worst case result

**average result

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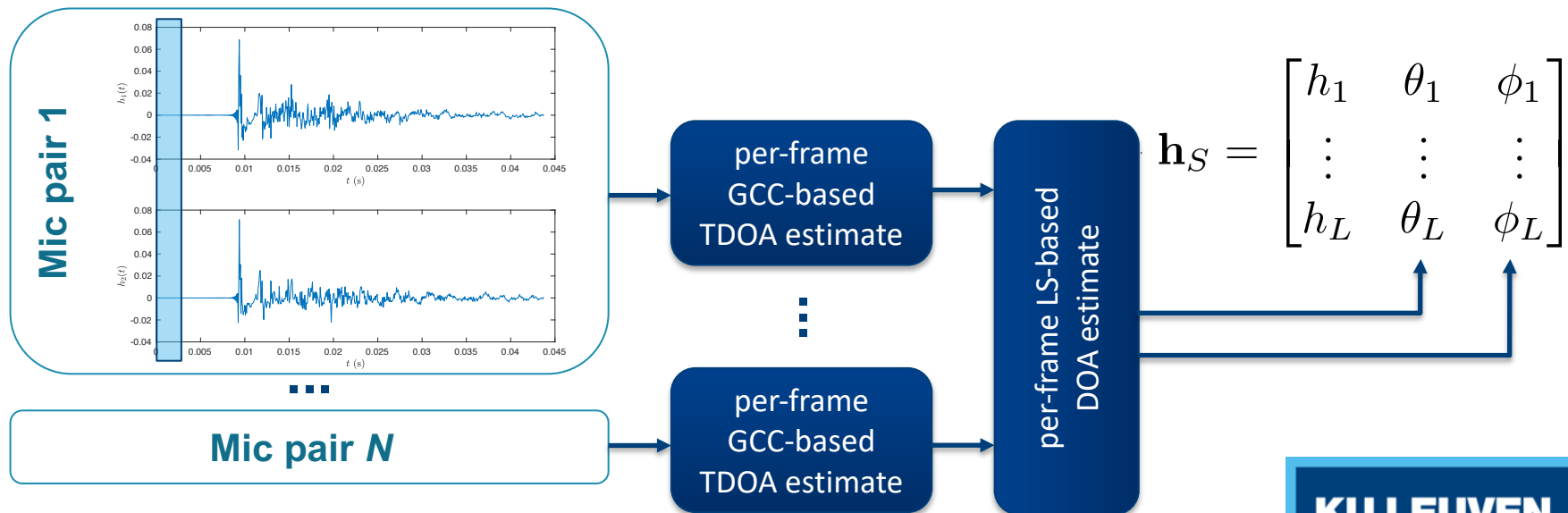
Outline

- **System overview & challenges**



Space-time analysis

- **Key model: Spatial room impulse response (SRIR)**
 - Include **direction of arrival information** (azimuth and elevation angles θ, ϕ) to each RIR “peak” representing room reflection
- **Key method: Spatial decomposition method (SDM)**
 - TDOA estimation of corresponding RIR “peaks” in each mic pair
 - Least-squares DOA estimation from TDOAs of all mic pairs in array



Space-time analysis

- **SDM challenges**

- Find phase-matched microphones 
- Find RIR “peaks”
- Match “peaks” in different RIRs $\rightarrow$ *generalized cross-correlation (GCC)*
- Estimate TDOAs with subsample accuracy $\rightarrow$ *TDOA interpolation*



- **Subsample TDOA estimation**

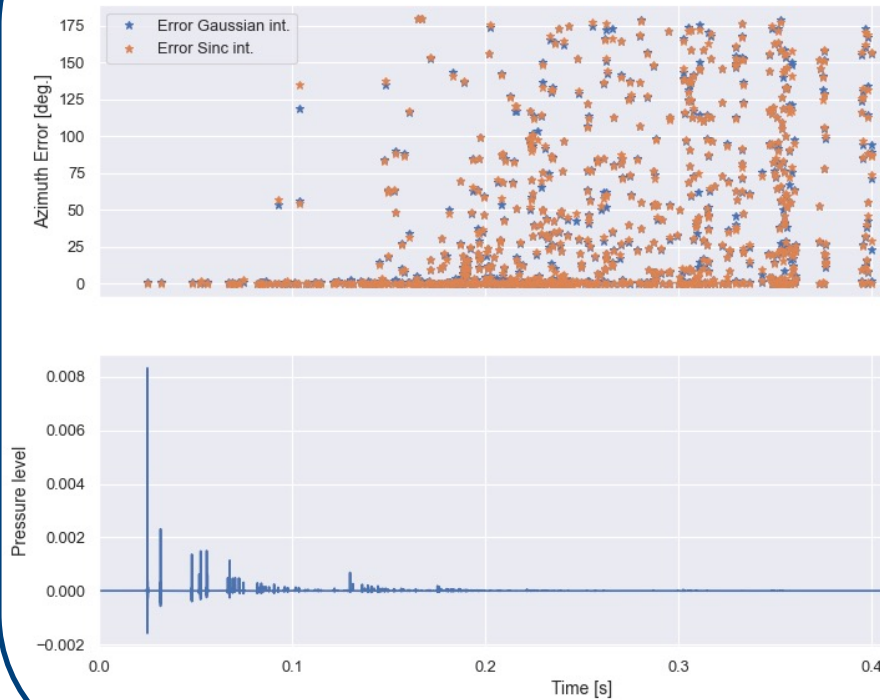
- Due to bandlimited nature of measurement system, RIR “peaks” are sinc functions rather than impulse functions
- GCC $R_{m,n}(t)$ of two time-shifted sinc functions = sinc function
- Optimal subsample TDOA estimation method is hence based on sinc interpolation rather than parabolic/Gaussian interpolation

$$\tau_{\text{sinc}} = \arg \min_{\tau} \int_{-\infty}^{\infty} \left| \text{sinc}(\pi f_s(t - \tau)) - \frac{R_{m,n}(t)}{\max(R_{m,n}(t))} \right|^2 dt$$

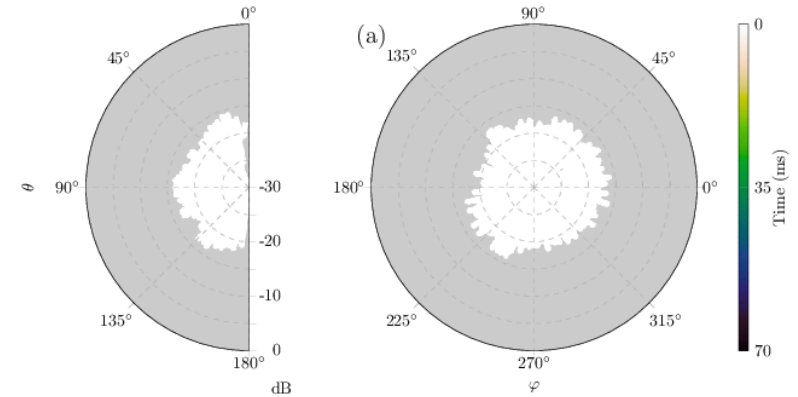
Space-time analysis

- Resulting space-time model

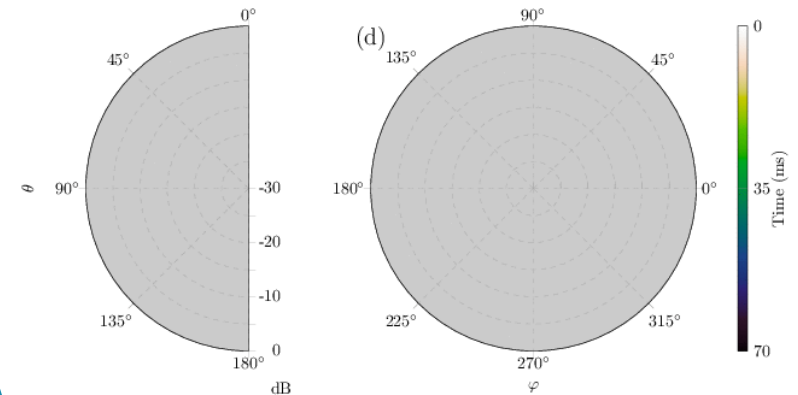
TDOA estimation errors



SRIR visualization of interpolated RIRs



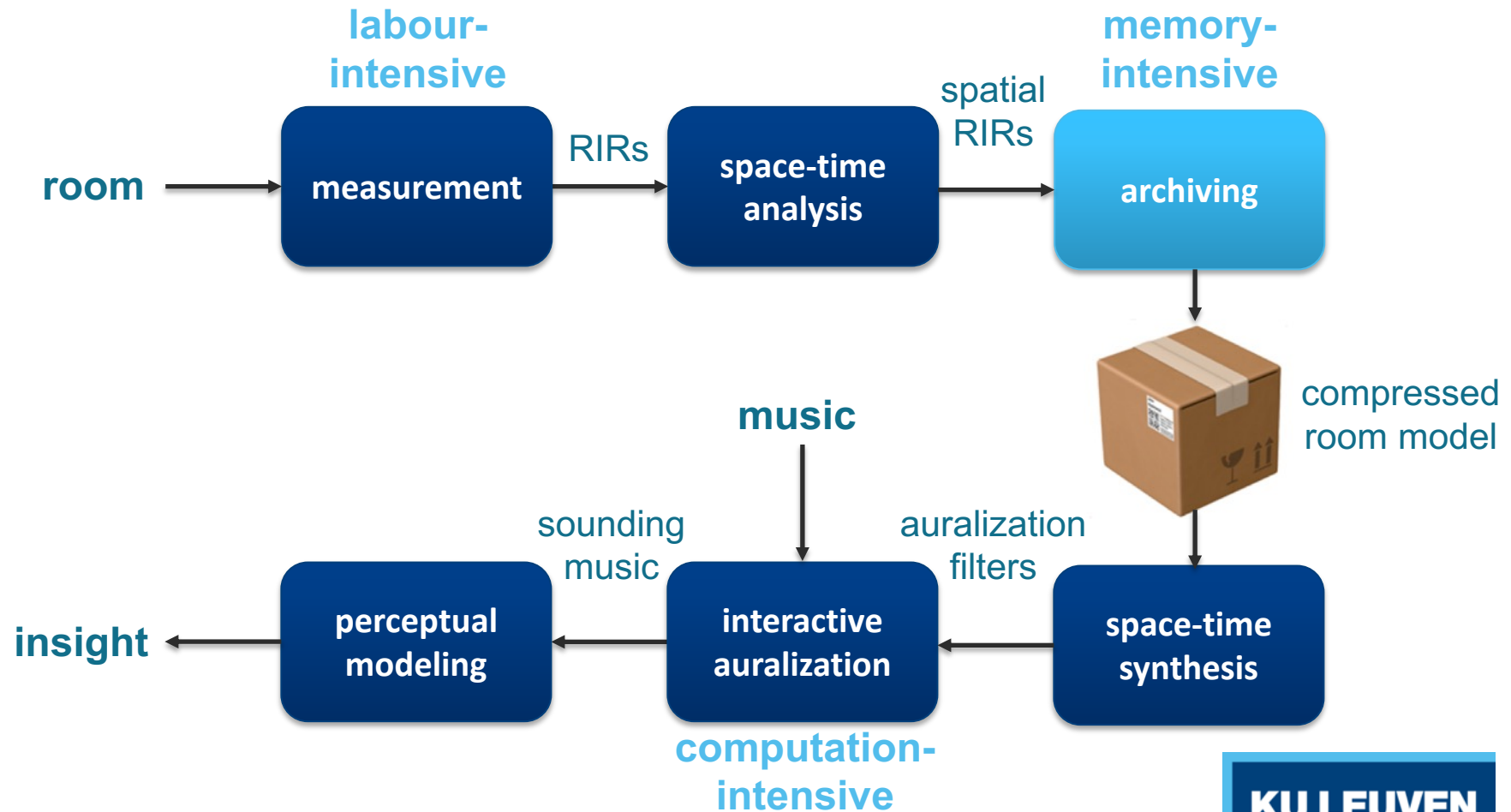
without sparse regularization



with sparse regularization

Outline

- **System overview & challenges**



Archiving

- **Why is archiving of church acoustics a challenge?**
 - Impulse response contains $L \sim 10^4$ samples $\sim 10^5$ bits
 - 1 spatial impulse response = 6 impulse responses $\sim 10^6$ bits
 - Plenacoustic sampling theory: accurate sound field reconstruction requires spatial resolution of ~ 10 cm
 - $\sim (100)^3$ source positions x $(100)^3$ observer positions
 - $\sim 10^{12}$ spatial impulse responses $\sim 10^{18}$ bits $\sim$ **100 petabyte**



Archiving

- How to “compress” a room impulse response?

- Truncation
- Hard thresholding
- Sparse approximation
- Low-rank approximation

Physical justification

non-homogenous Helmholtz equation

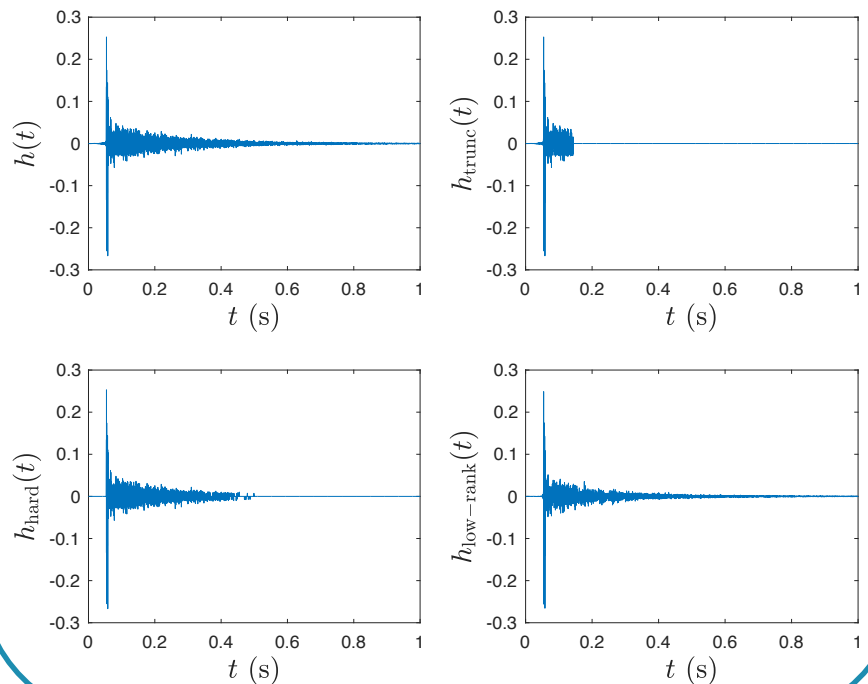
$$(\nabla^2 + k^2)H(\mathbf{r}, \mathbf{r}_s, \omega) = -j\omega\rho_0 Q \delta(\mathbf{r} - \mathbf{r}_s)$$



RIR $\approx$ sum of damped sinusoids

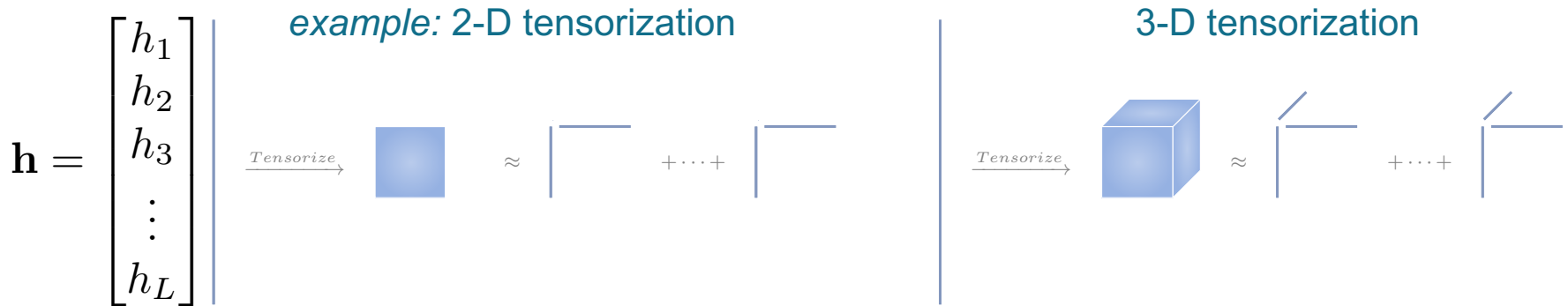
$$h(t) \approx \sum_{m=1}^M \mu_m e^{-\beta_m t} \cos(\omega_m t + \phi_m), \quad t \geq 0$$

RIR with three compression methods
(compression rate = 85 %)



Archiving

- **Key idea: Sum of M (damped) sinusoids admits matricization/tensorization of rank $2M$**
 - Impulse response vector $\mathbf{h}$ (= 1-D array) of length L can be **reshaped** into N -D array of dimensions $\sqrt[N]{L} \times \sqrt[N]{L} \times \dots \times \sqrt[N]{L}$
 - N -D array can be **approximated** as sum of $2M$ rank-1 terms (SVD or canonical polyadic decomposition)

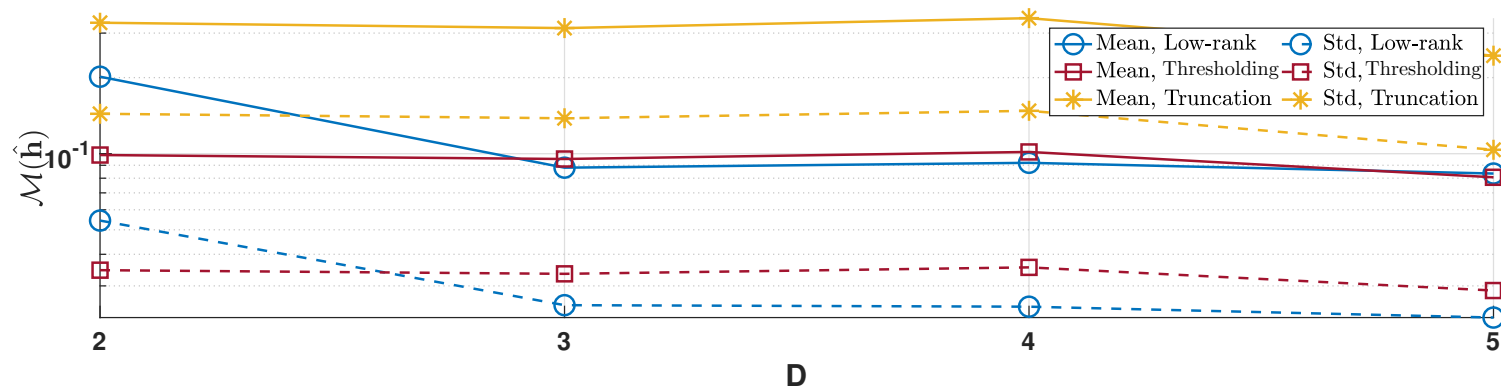


$$\mathbf{h} = \text{vec} \left(\sum_{m=1}^{2M} \mathbf{u}_m^{(1)} \circ \mathbf{u}_m^{(2)} \circ \dots \circ \mathbf{u}_m^{(N)} \right)$$

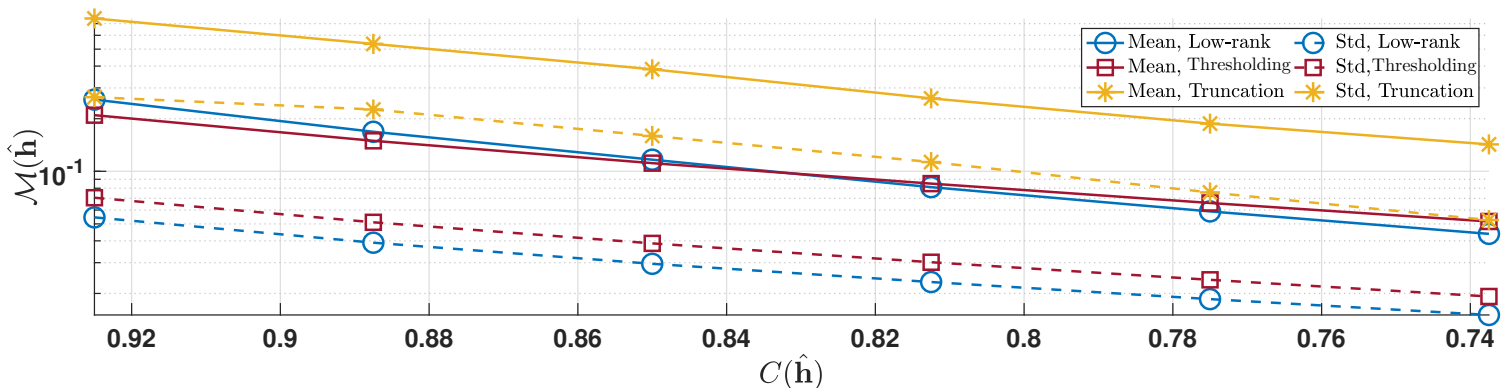
Archiving

• Misadjustment results after RIR compression

- Methods: truncation, hard thresholding, low-rank approximation
- Misadjustment vs. tensorization dimension D & compression rate C



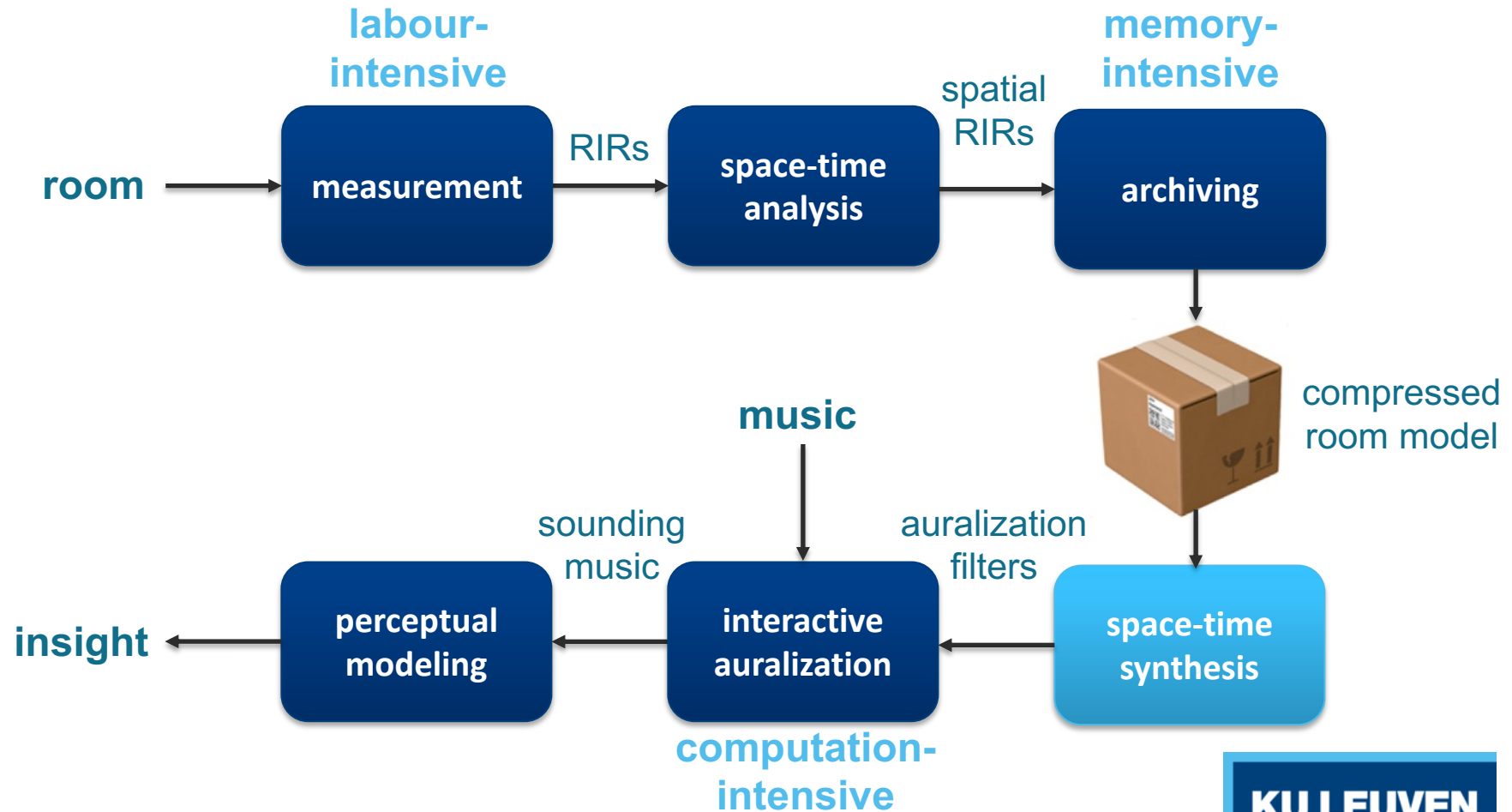
$C = 80 \%$



$D = 3$

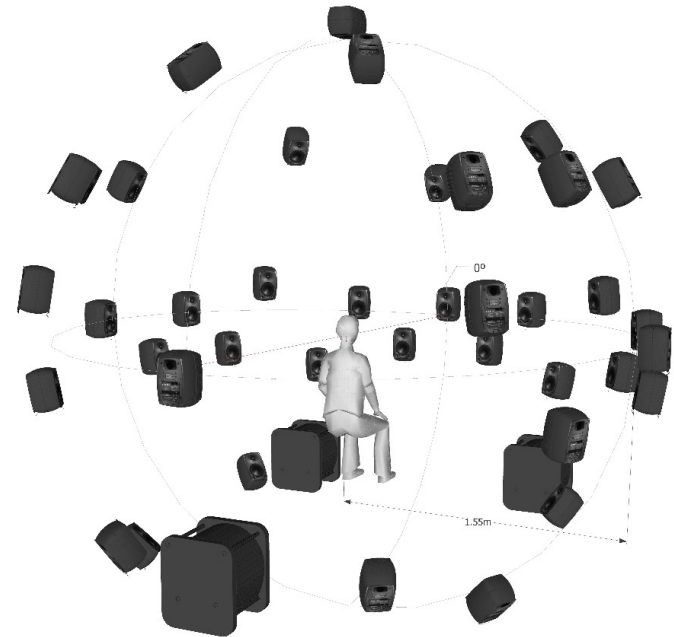
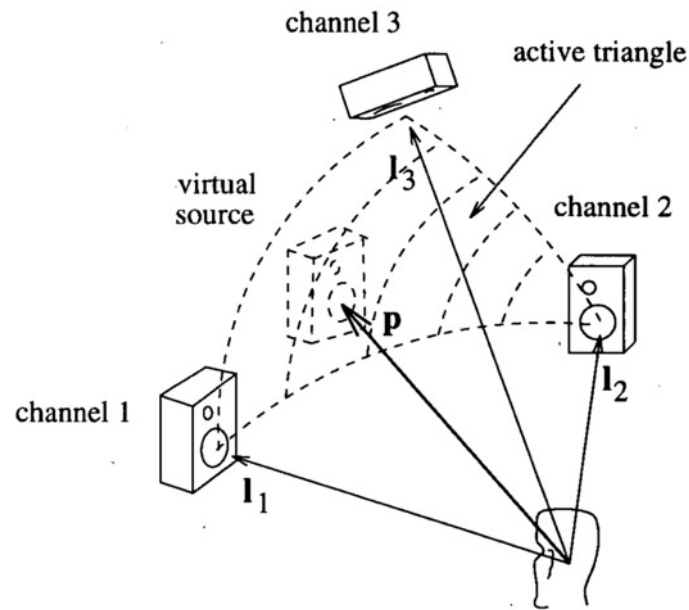
Outline

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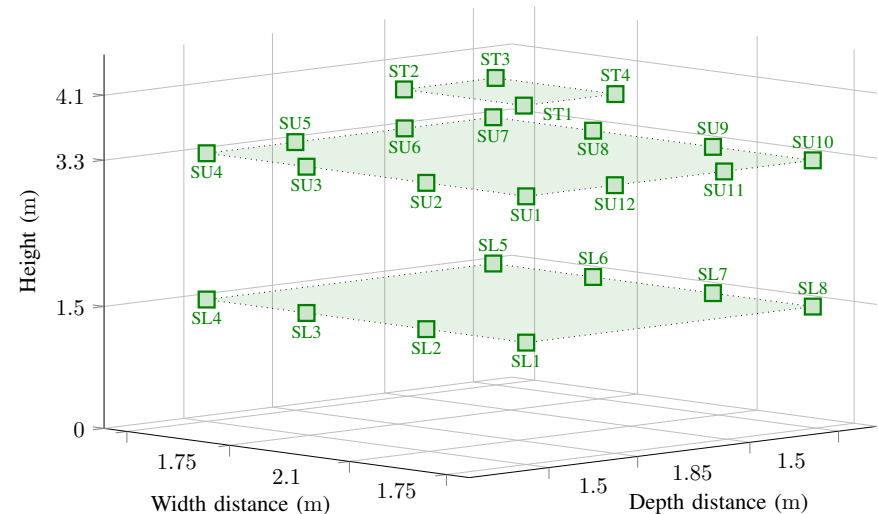
Space-time synthesis

- **Key idea: Mapping of room reflections to loudspeakers**
 - Conversion of SRIR to set of per-loudspeaker impulse responses
 - Vector base amplitude panning (VBAP)
 - Nearest-loudspeaker mapping



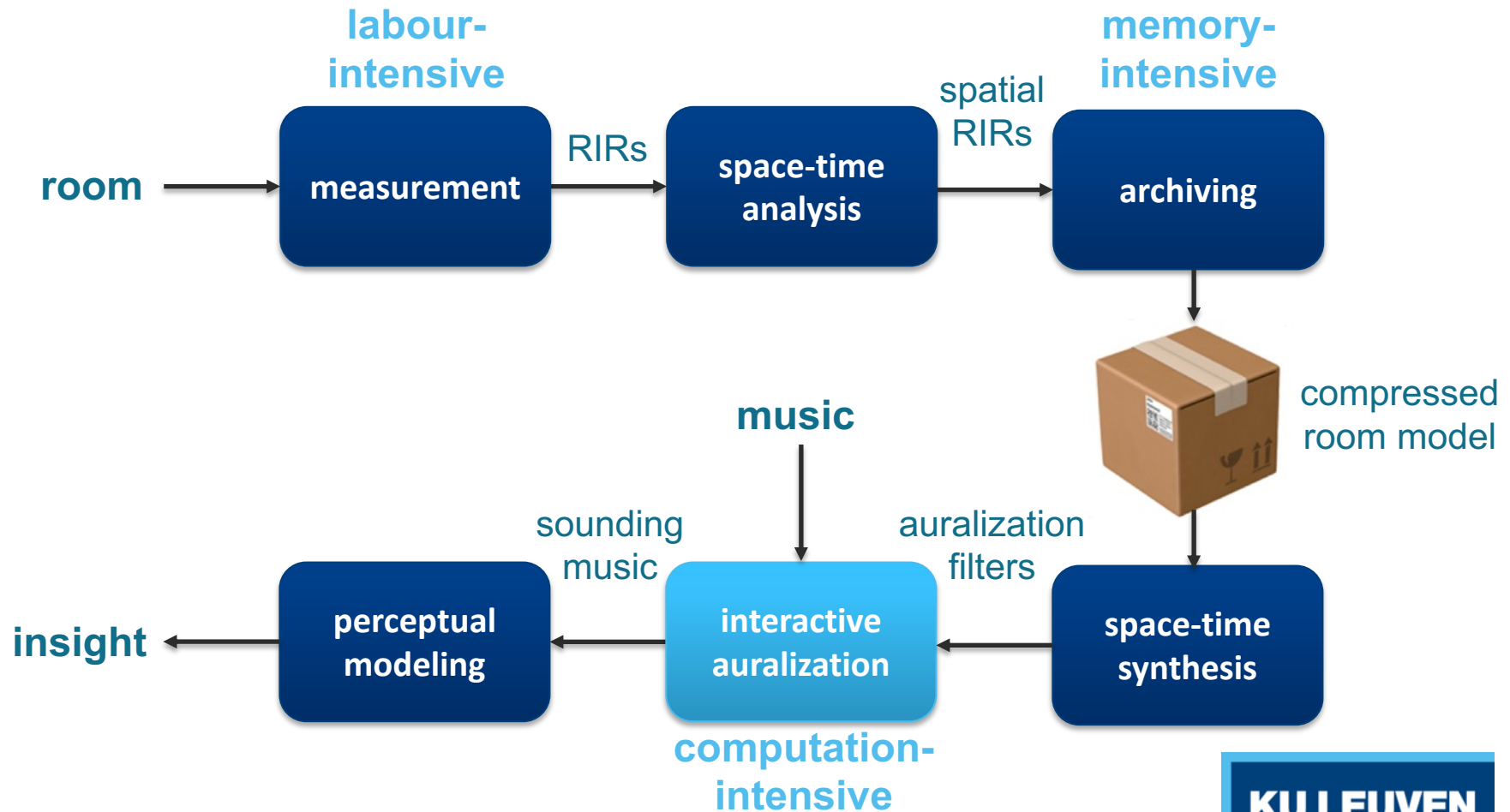
Space-time synthesis

- **Loudspeaker set-up: Library of Voices (Leuven)**
 - 3-D array of 20–24 Martin Audio 6,5” CDD loudspeakers
 - Reproduction room: RT60 = 0.5 s (with curtains closed)



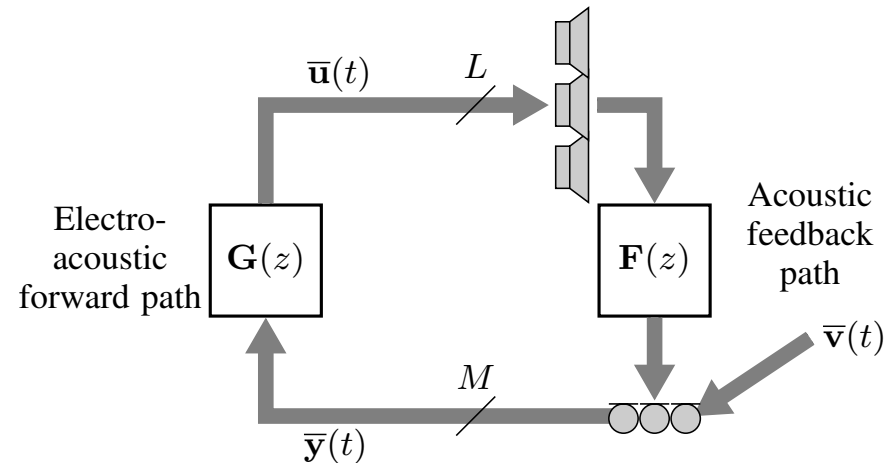
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Interactive auralization

- **Challenges:**
 - Multi-channel acoustic feedback control
 - Low-latency real-time multi-channel convolution



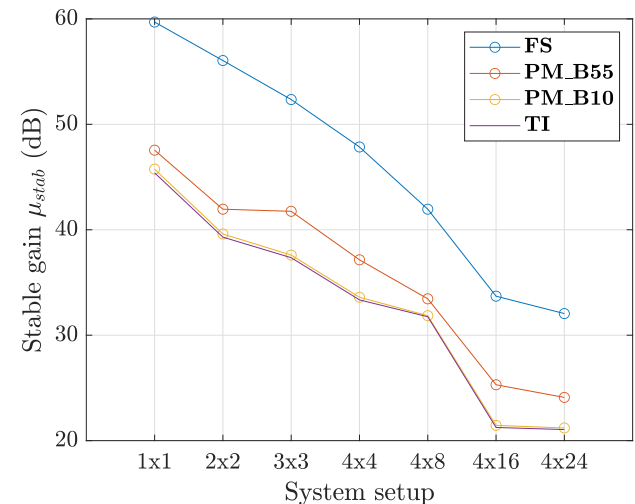
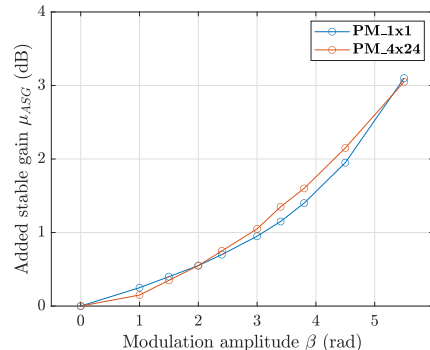
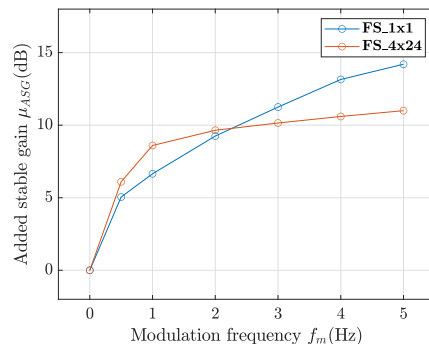
Interactive auralization

- **Multi-channel acoustic feedback control**

- Gain before instability decreases with increasing # channels
- Feedback-related artefacts: howling, ringing, coloration

- **Acoustic feedback control methods:**

- Phase modulation methods → *Reverberation enhancement systems*
- Gain reduction methods
- Spatial filtering methods
- Room modeling methods



FS = frequency shifting (5 Hz)

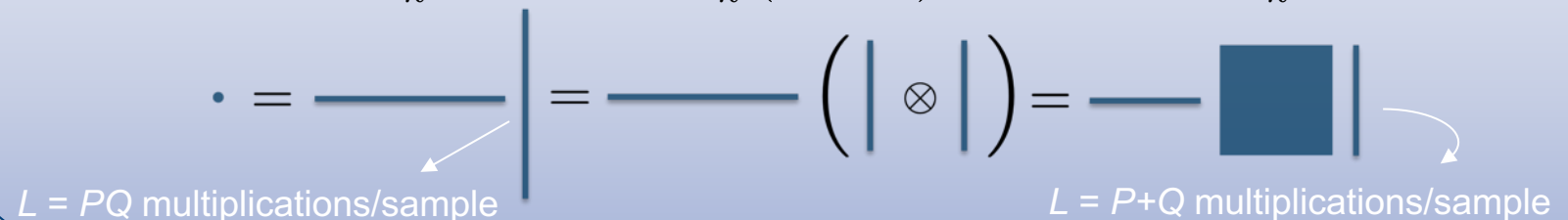
PM = phase modulation with modulation index $\beta=1.0$ or 5.5

Interactive auralization

- **Low-latency real-time multi-channel convolution**
 - **Key idea:** Rewrite convolution operation using compressed RIR
 - **Low latency:** no buffering required by staying in time domain
 - **Real-time:** reduced # input buffer reads and # FLOPS
 - Truncation $\rightarrow$ shorter FIR filter
 - Hard thresholding $\rightarrow$ FIR filter with fewer non-zero coefficients
 - Sparse approximation $\rightarrow$ short “warped” FIR filter (mixed-Kautz model)
 - Low-rank approximation $\rightarrow$ “low-rank convolution”

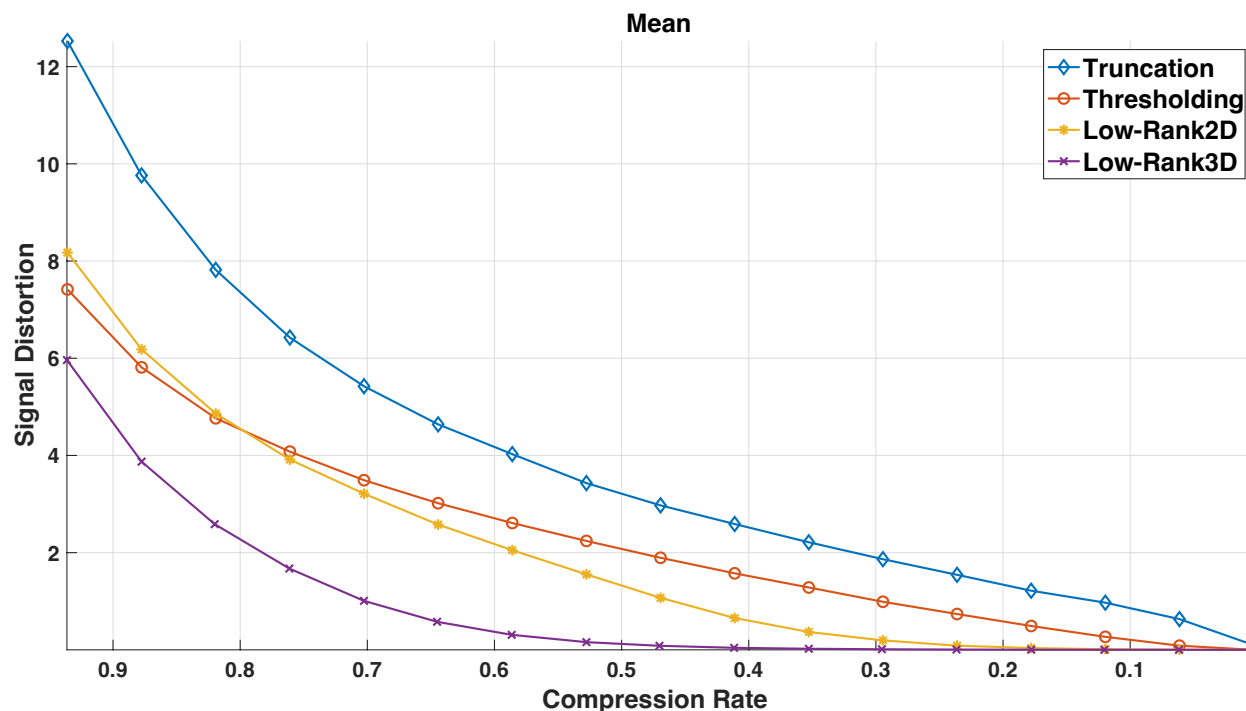
Low-rank convolution: illustration for rank-1 model $\mathbf{h}_{L \times 1} = \mathbf{u}_{P \times 1} \otimes \mathbf{v}_{Q \times 1}$

$$y_k = \mathbf{x}_k^T \mathbf{h} = \mathbf{x}_k^T (\mathbf{u} \otimes \mathbf{v}) = \mathbf{v}^T \mathbf{X}_k^T \mathbf{u}$$



Interactive auralization

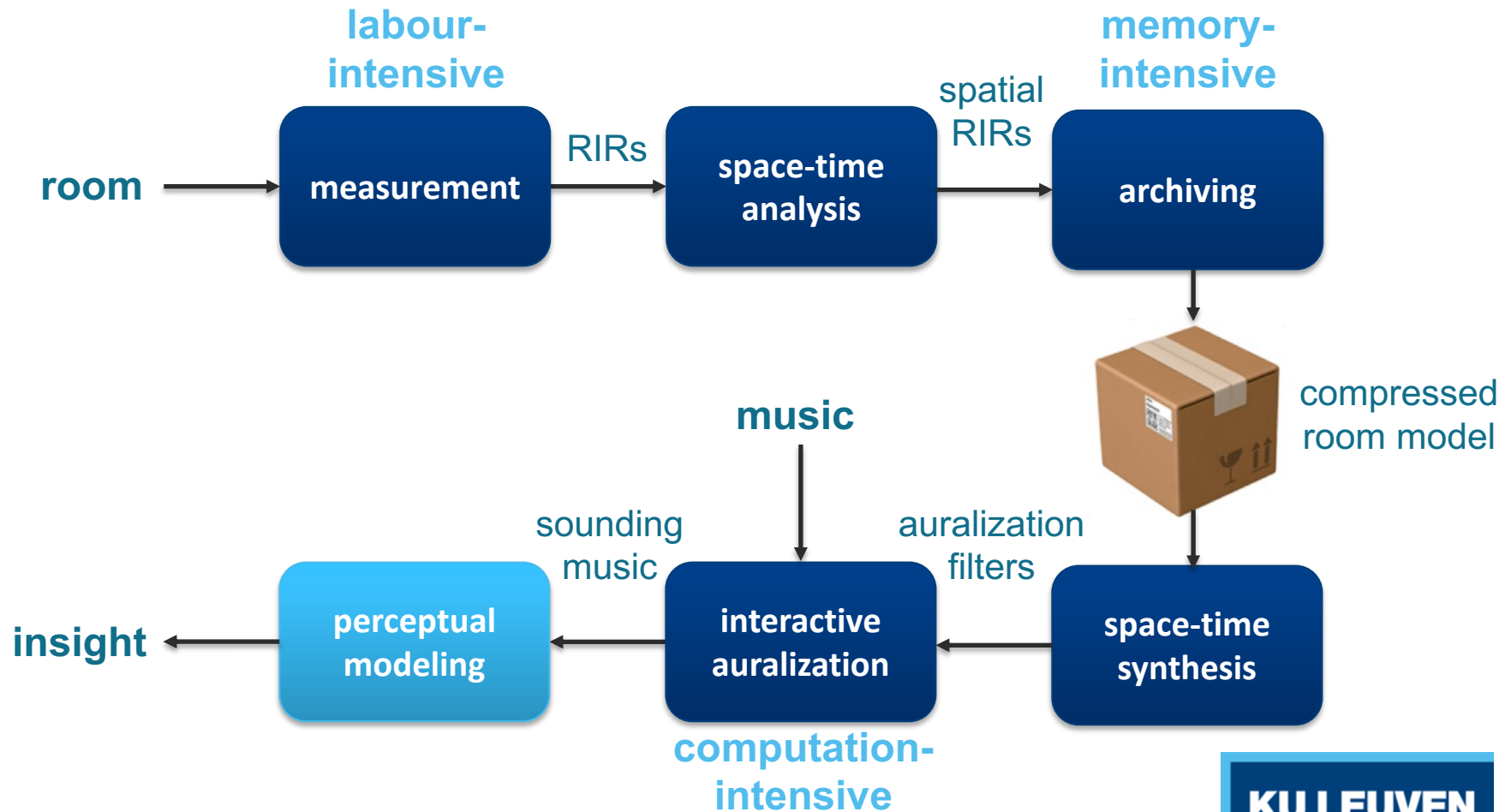
- **Signal distortion after compressed RIR convolution**
 - Methods: truncation, hard thresholding, low-rank approximation
 - Mean ERB-weighted signal distortion vs. RIR compression rate C



input = dry vocal music
RIR = Nassau Chapel RIR,
RT60 = 2.7 s, $L = 117649$

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Perceptual modeling

- **Aim:** understand relation between acoustic characteristics of church and their perceived quality (broadly defined)
- **Key methodology: Flash Profile rapid sensory analysis**
 - Originally developed in frame of **food tasting** experiments
 - Validated for perceptual modeling of virtual acoustics **by listening** (e.g. auralization of concert halls, car cabins, domestic rooms)
 - Evaluated in preliminary experiments for perceptual modeling of virtual church acoustics **by listening while singing**



Materials

- 6 virtual acoustic spaces
- 4 expert listeners: male singers from Cappella Pratensis
- 2 pieces: plainchant + polyphonic



Perceptual modeling

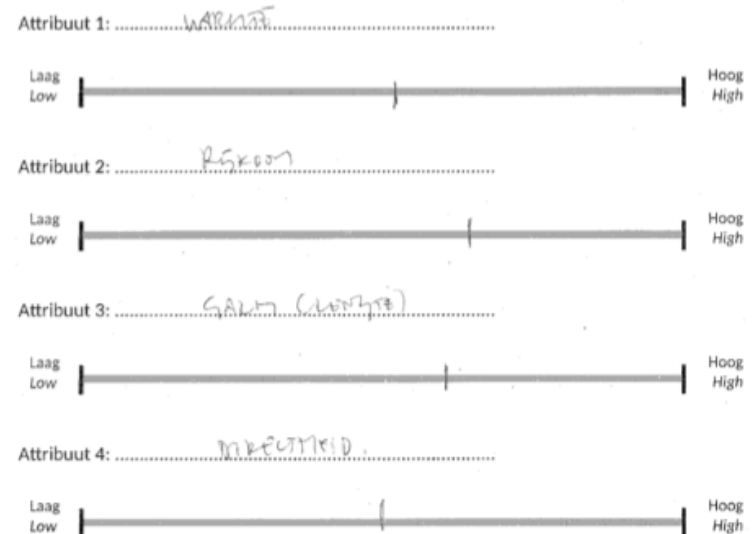
- **Flash Profile Methodology:** experiment design
 - **1) Elicitation phase:** individual semantic definition of perceptual attributes to characterize differences between virtual spaces
 - **2) Ranking phase:** continuous-scale (low-high) quantification of each perceptual attribute for each virtual space

Expert 1

1 DRYNESS
2 SHARPNESS
3 WARMTH
4 LOWER FREQUENCY FRIENDLINESS
5 FULLNESS

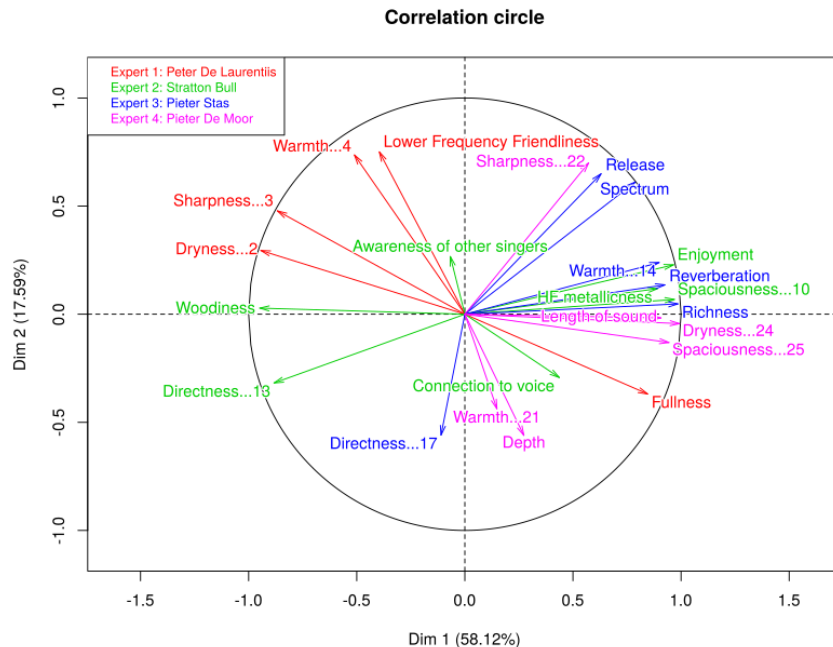
Expert 2

① enjoyment
② awareness of other signs
③ woodiness
④ spaciousness
⑤ connection to voice
⑥ high frequency-metallic
⑦ directness



Perceptual modeling

- **Flash Profile Methodology:** statistical data analysis
 - Multi-factor principal component analysis + clustering



- 75% of perceived variance among spaces is modeled by two largest principal components
- two largest principal components correlate to perceptual attributes of **spaciousness/reverberance** (Dim 1) and **spectral content** (Dim 2)
- six virtual spaces can then be mapped into this 2-D principal component subspace

Conclusion

- **Some take-home messages...**
 - **Acoustic signal processing in digital humanities:** case studies in humanities/musicology/cultural heritage/... may carry remarkably many research challenges relevant to our community
 - **Fundamental signal processing research** (sampling & interpolation, sparse & low-rank modeling, ...) can be highly relevant to address application-specific bottlenecks relating to labour, memory, and computational resources
 - **Interdisciplinary research** is much more than putting together diverse team – it requires to learn common language and mutual scientific understanding, attract funding, develop proper research methodology, ... which is not easy but very enriching
 - Looking for challenging research topic in acoustic signal enhancement? Consider **multi-channel acoustic feedback control**

Thank you...

- **This presentation is the result of joint work with:**
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- **Contact**

toon.vanwaterschoot@esat.kuleuven.be

<https://tvanwate.github.io>