

Acoustical Impulse Response Measurement with

Αλύκη

(Aliko)

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- What's all this impulse response and convolution stuff anyhow ?
- Use of impulse responses in acoustics and audio
- Acoustic IR capturing methods
- Measures derived from the IR
- Aliko — requirements
- Aliko — program structure
- [Demo](#)
- Aliko — the future
- The workshop

The impulse response of a system is its output when the input is a single *Dirac* pulse.

- In the analog world, a Dirac pulse is an infinitely short pulse with unit energy.
- In the digital world, it is a signal consisting of one nonzero sample preceded and followed by all zeros.

If a system is *linear* and *time invariant* then the IR tells all there is to know about it. Given the IR of such a system, we can compute its output for any input.

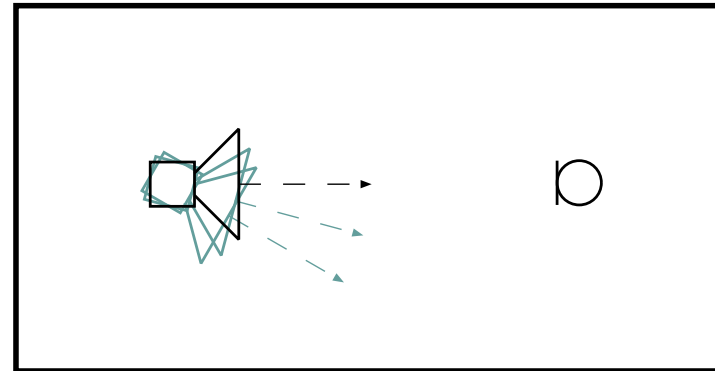
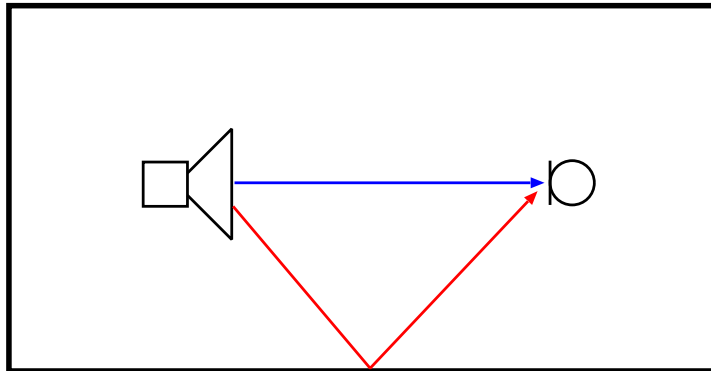
- Any input signal is the sum of a number of time-shifted Dirac pulses, one for each sample.
- Then the output is the sum of a number of time-shifted copies of the IR, one for each sample.

The process of combining an input signal and an IR to obtain the output is called *convolution*.

- A Finite Impulse Response (FIR) filter performs convolution. The coefficients of the the file *are* the IR. This requires one multiplication and addition per sample per coefficient.
- Convolution with long IR can be done more efficiently by FFT-based methods. They can easily be a factor 1000 or more faster than a simple FIR.
- This method is used by 'convolution engines' such as BruteFir, jackconvolve, and JACE.

Electro-acoustics: measurements of microphones and loudspeakers

- Frequency response, delay, directivity, ...
- IR techniques allow in many cases to eliminate the need for an anechoic room.
- Polar response plots require many measurements, and need to be automated.



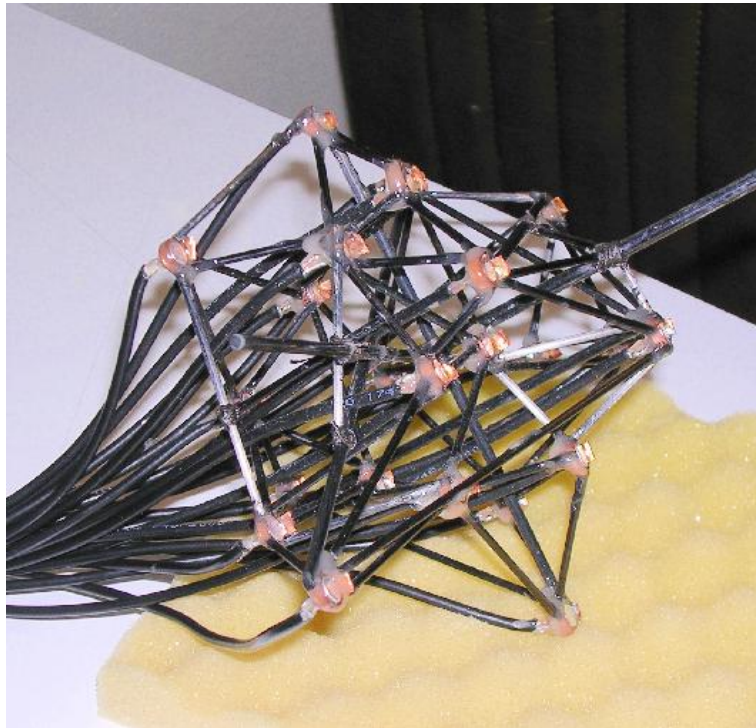
Architectural, musical and psychoacoustics:

- Evaluation of room character and qualities.
- 'Tuning' of concert halls.
- Measuring Head Related Transfer Functions.

Sound recording and reproduction:

- Convolution based reverb to reconstruct real spaces.
- Emulation of analog hardware, including non-linear.
- Processing for Ambisonics, Wave Field Synthesis, Crosstalk Cancellation, ...

Example: High Order Ambisonics microphones:

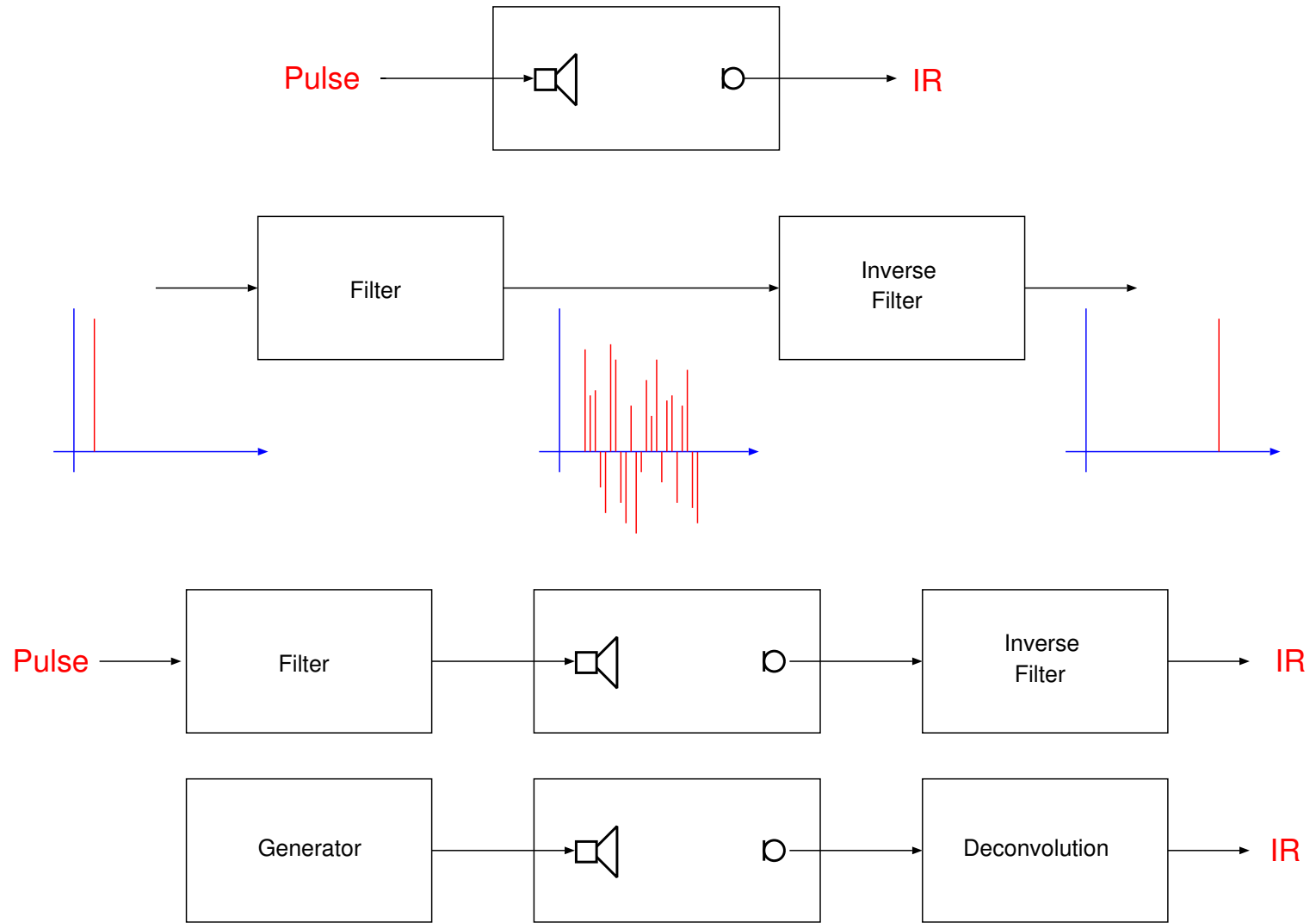


- Cluster of $N = 16$ or 25 microphones, at 'random' positions.
- Mic outputs are processed by a $N \times N$ convolution matrix, producing B-format.
- The encoding matrix must be computed from IR measurements.

Practical problem in acoustic IR measurement: a Dirac pulse is very short and hence has very low energy. Since there are always background noises, we have a bad signal to noise ratio.

- For measuring acoustic spaces, the Dirac pulse can be produced directly in the acoustic domain: bursting balloons, pistol shots, electric sparks, ...
- These methods are impractical, and provide limited repeatability.
- For measuring speakers and microphones, and to obtain IR for a convolution reverb we need more precision and another solution.

Acoustics IR capturing methods – (1)



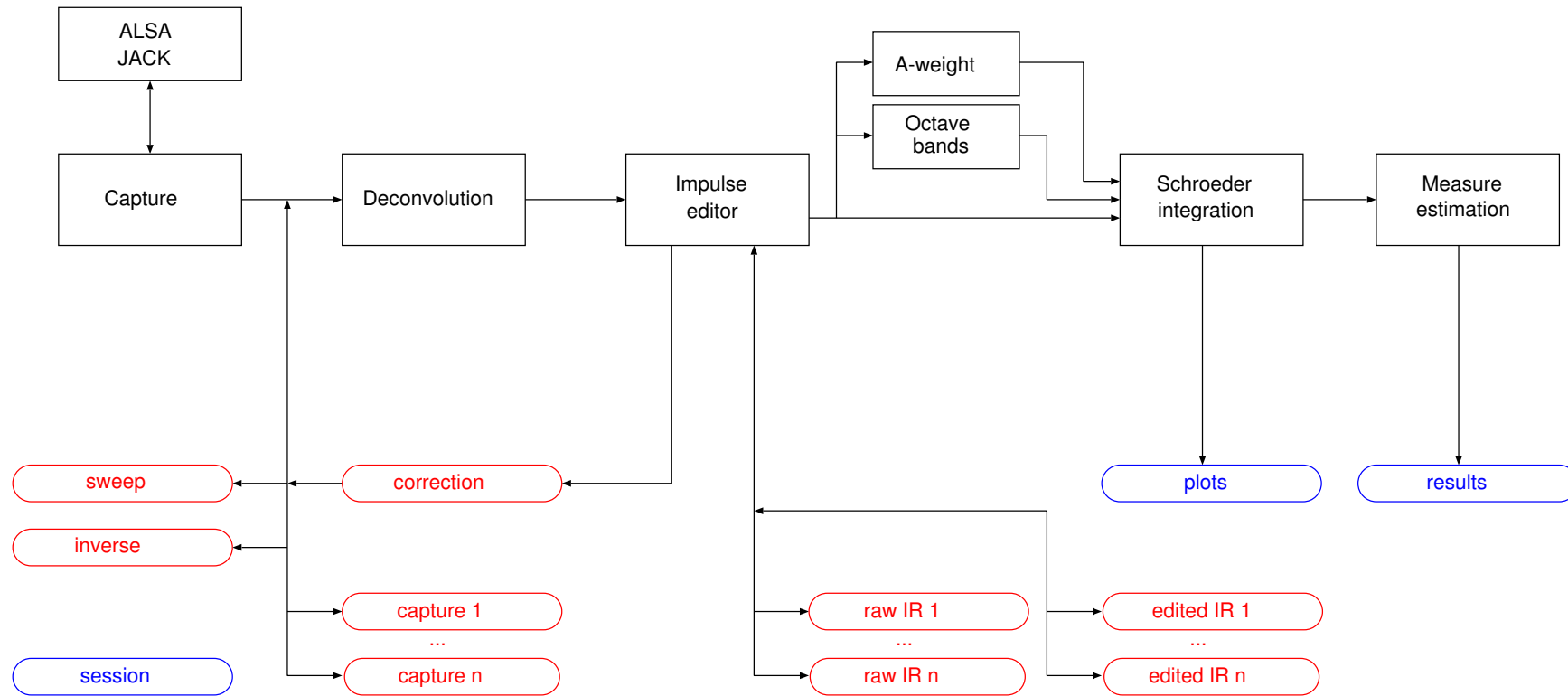
White or pink noise derived from binary MLS sequences.

- Deconvolution is performed by the Fast Hadamard Transform.
- Very sensitive to distortion in the measurement chain.

Linear and logarithmic frequency sweeps.

- Inverse filtering by fast convolution.
- Can be very long, giving good S/N ratio.
- Distortion appears in negative time and can be separated.
- Does not require precise synchronisation of playback and capture clocks.
- Logarithmic sweeps (pink spectrum) preferred in most cases.

- Integrated and easy to use for Linux dummies.
- Accurate and flexible.
- Support a variety of uses.
- Observe international standards for acoustics.
- Permit multichannel measurements.
- Support automated measurement series.
- Import and export standard audio files.
- Support report generation.
- Be expandable to a general purpose 'acoustics workbench'.



- Measuring a reverb program.
- Measuring the lecture room.
- An experiment.



Lots of work . . .

- Clean up the code.
- Manual and documentation.
- Add correction filter calculation.
- Add octave band filters.
- Add room acoustics measures calculation.
- Add polar plot output.
- . . .

Friday afternoon in the Kubus

- Provide hands-on experience.
- Measure the Kubus IR using a variety of microphones.
- Kick of the Free Impulse project.
- Listen to the result using 3-D Ambisonics playback (saturday).

I wish to thank:

- Prof. Angelo Farina (University of Parma, Italy) for reviewing the paper and providing helpful comments.
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