ACOUSTICS
part – 5
Sound Engineering Course

Angelo Farina

Dip. di Ingegneria Industriale - Università di Parma
Parco Area delle Scienze 181/A, 43100 Parma – Italy
angelo.farina@unipr.it
www.angelofarina.it

Impulse Response measurements

The Past

- Traditional time-domain measurements with pulsive sounds and omnidirectional transducers
- Electroacoustical measurements employing special computer-based hardware, a loudspeaker and an omnidirectional microphone

The Present

- Electroacoustical measurements employing standard sound cards, 2 or more loudspeakers and multiple microphones (2 to 8)

The Future

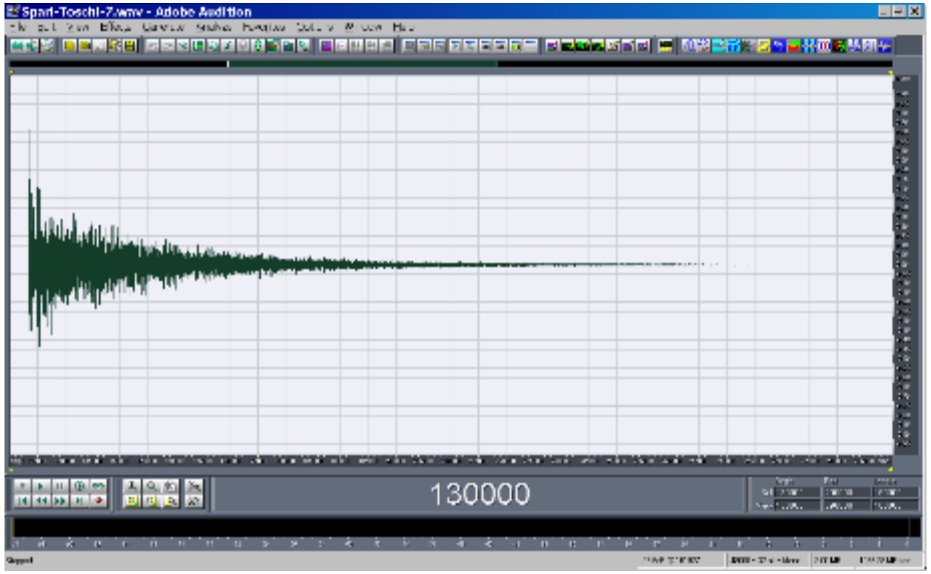
- Microphone arrays for capturing high-order spatial information
- Artificial sound sources employing a dense array of loudspeakers, capable of synthesizing the directivity pattern of any real-world source

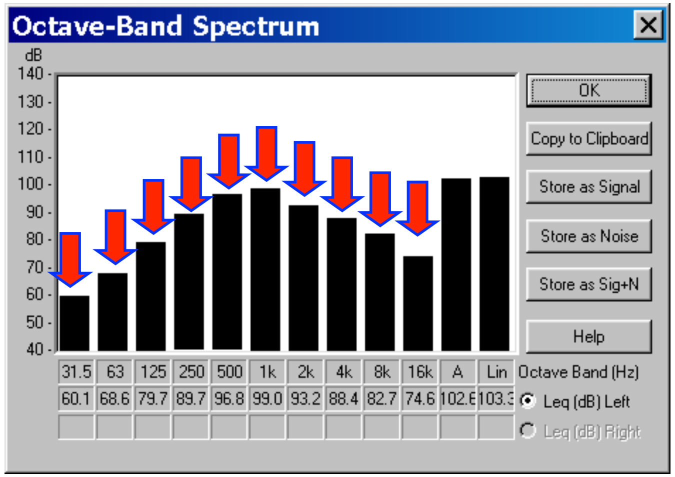
The Past

Traditional measurement methods

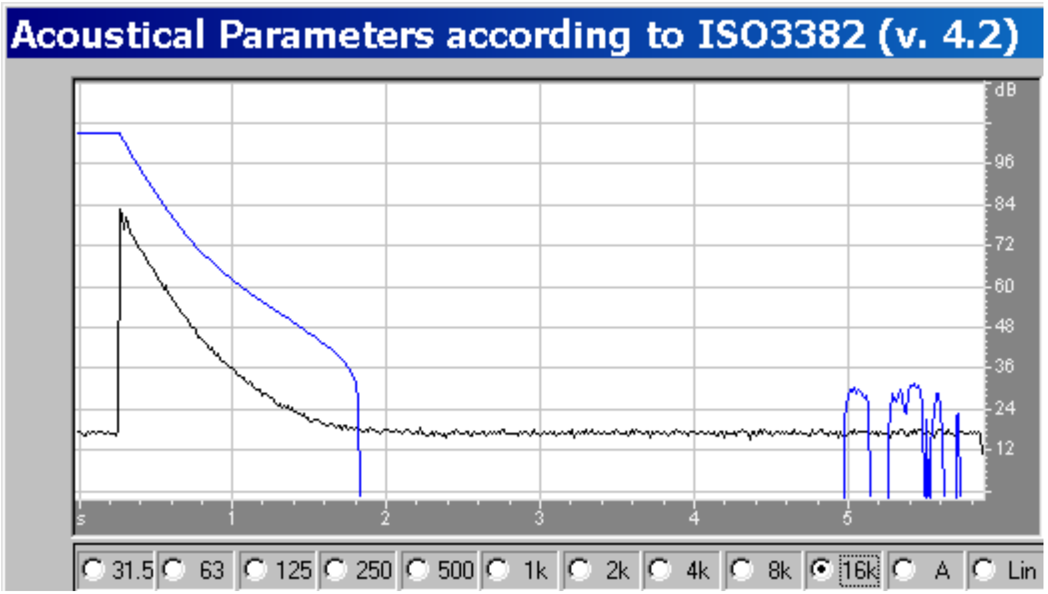
- Pulsive sources: ballons, blank pistol

Example of a pulsive impulse response



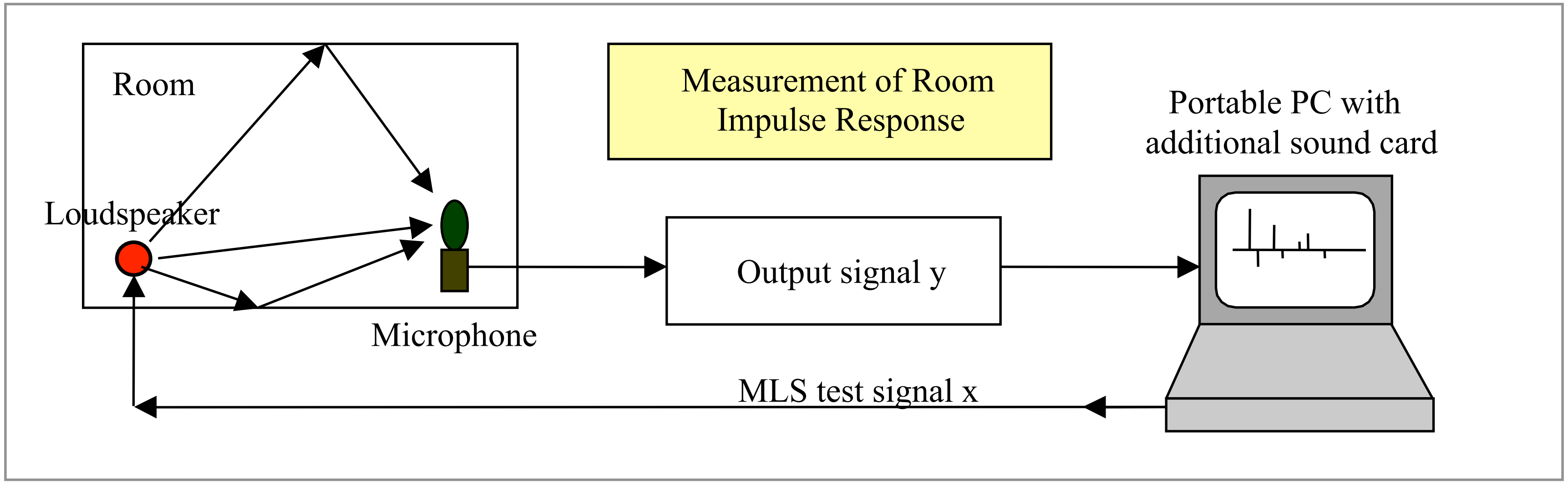


Note the bell-shaped spectrum



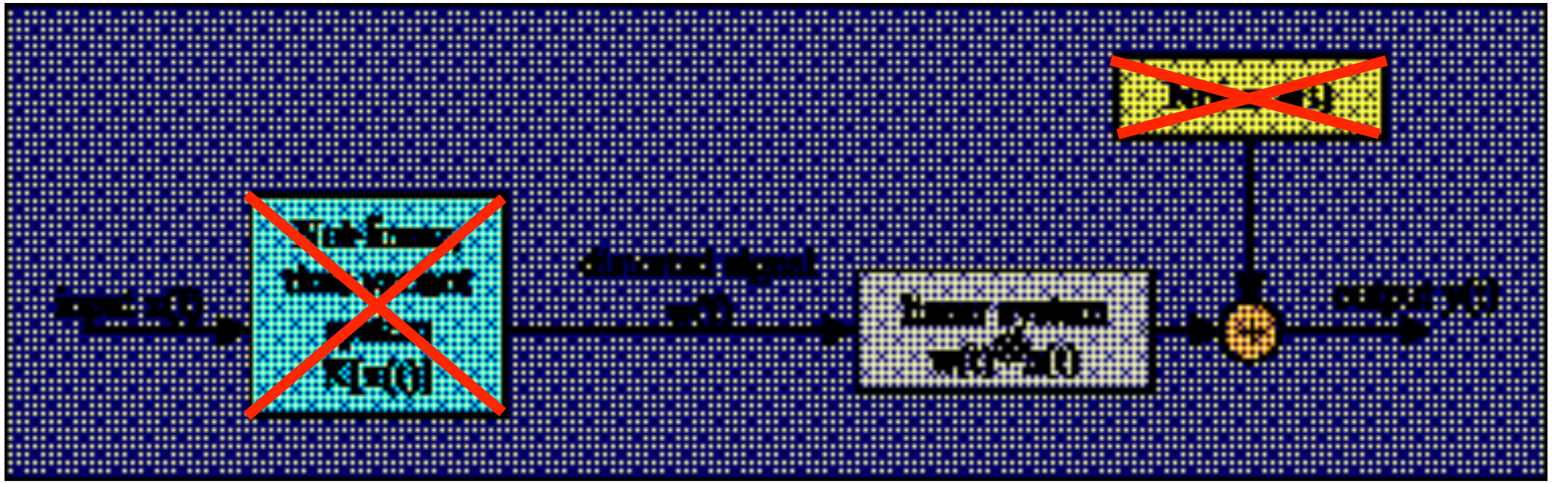
Electroacoustical measurements

Loudspeaker as sound source



- A loudspeaker is fed with a special test signal x(t), while a microphone records the room response
- A proper deconvolution technique is required for retrieving the impulse response h(t) from the recorded signal y(t)

Measurement process

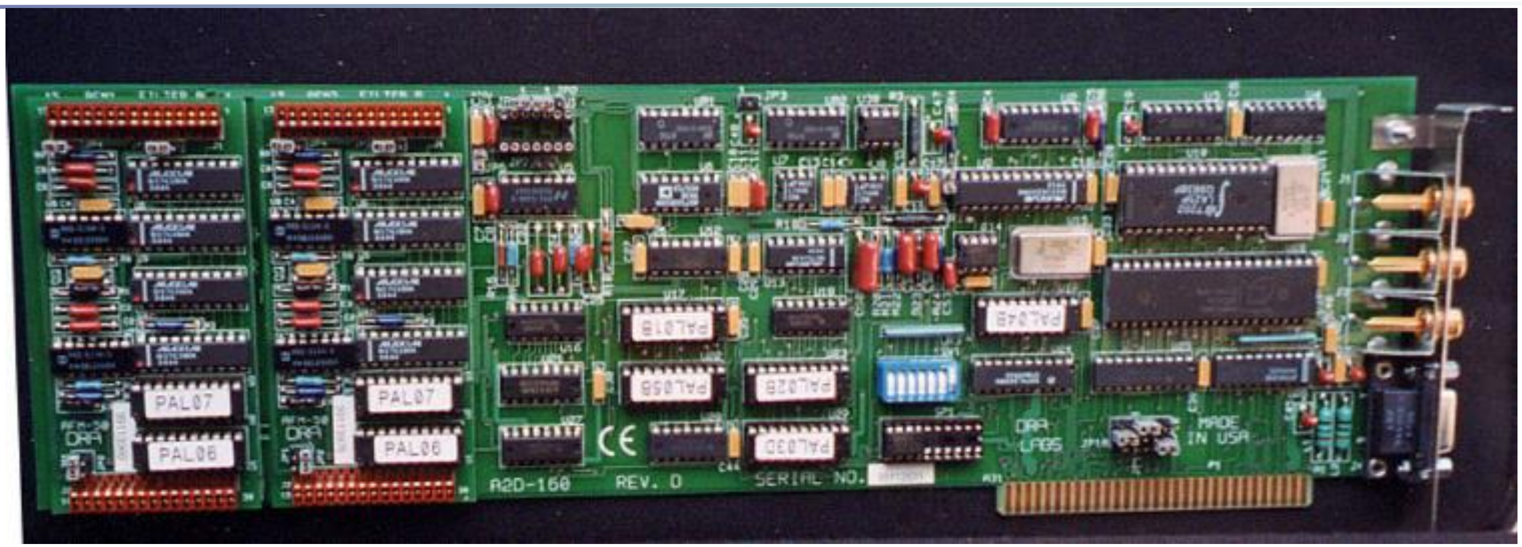


- The desidered result is the linear impulse response of the acoustic propagation h(t). It can be recovered by knowing the test signal x(t) and the measured system output y(t).
- It is necessary to exclude the effect of the not -linear part K and of the background noise n(t).

Electroacoustical methods

- Different types of test signals have been developed, providing good immunity to background noise and easy deconvolution of the impulse response:
  - MLS (Maximum Lenght Sequence, pseudo-random white noise)
  - TDS (Time Delay Spectrometry, which basically is simply a linear sine sweep, also known in Japan as "stretched pulse" and in Europe as "chirp")
  - **ESS (Exponential Sine Sweep)**
- Each of these test signals can be employed with different deconvolution techniques, resulting in a number of "different" measurement methods
- Due to theoretical and practical considerations, the preference is nowadays generally oriented for the usage of **ESS** with not-circular deconvolution

The first MLS apparatus - MLSSA



- MLSSA was the first apparatus for measuring impulse responses with MLS

More recently - the CLIO system

PB-4281

SC-01

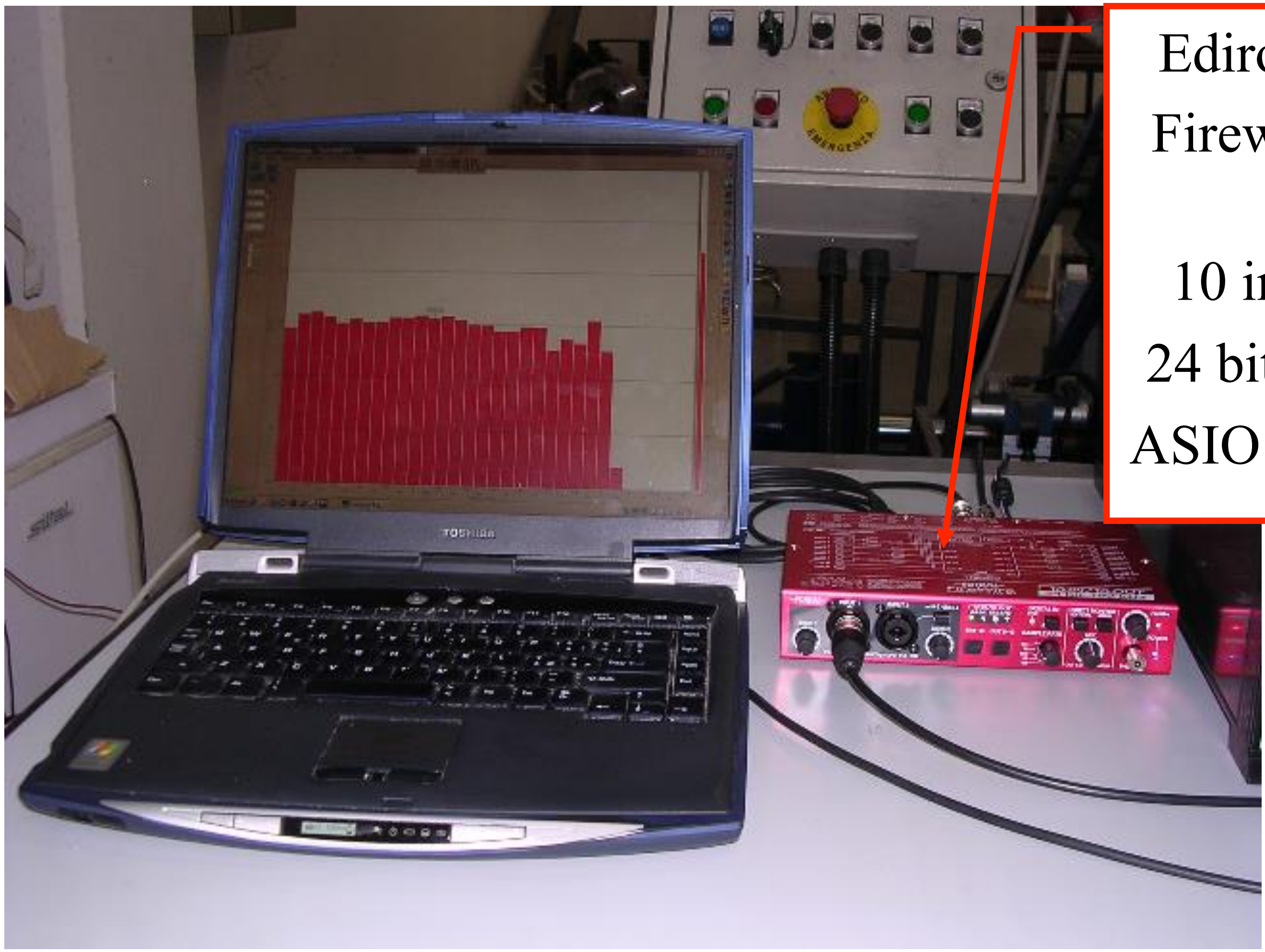
- The Italian-made CLIO system has superseded MLSSA for most electroacoustics applications (measurement of loudspeakers, quality control)

The first TDS apparatus - TEF

- Techron TEF 10 was the first apparatus for measuring impulse responses with TDS
- Subsequent versions (TEF 20, TEF 25) also support MLS

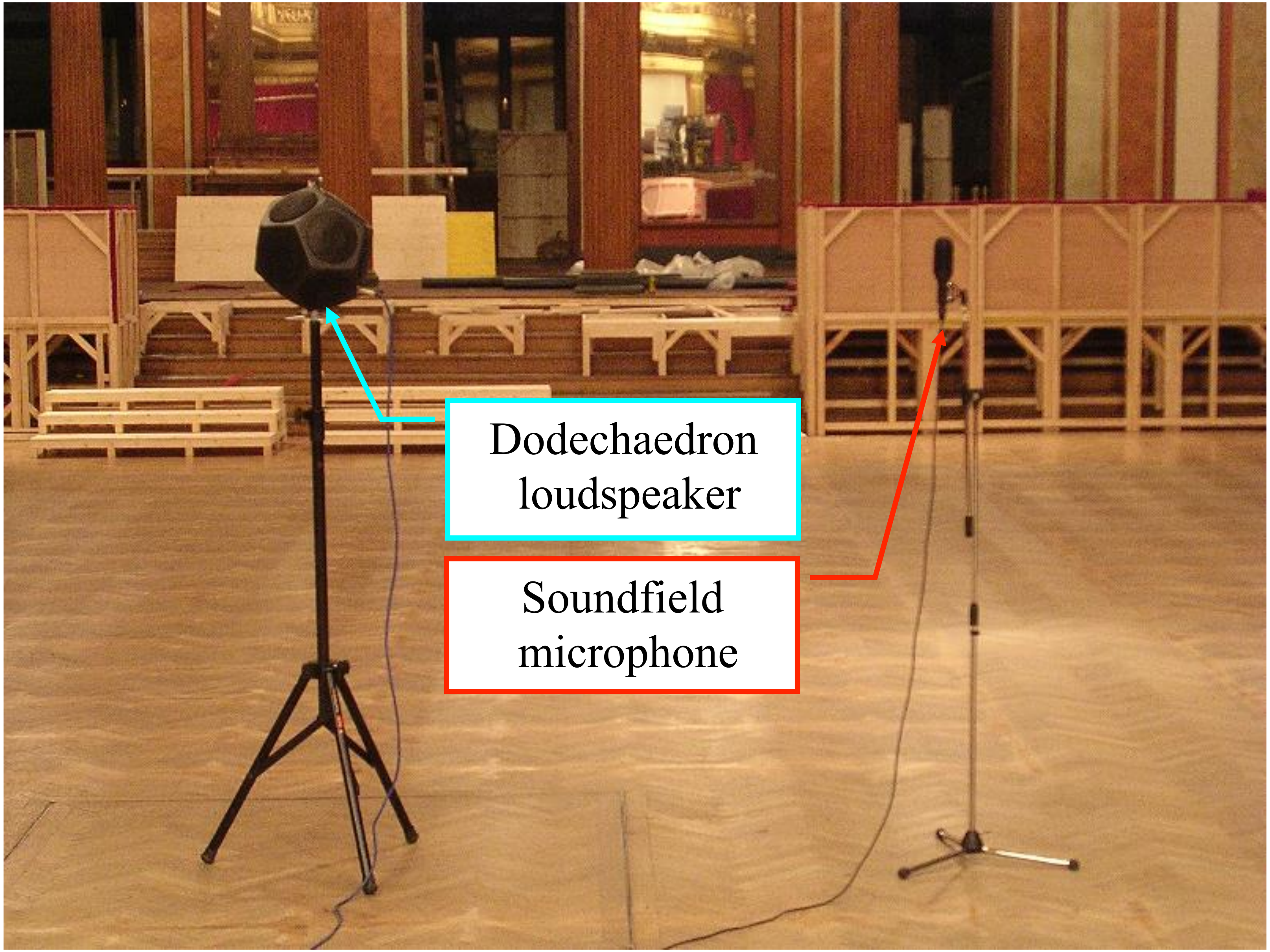
The Present

Today’s Hardware: PC and audio interface



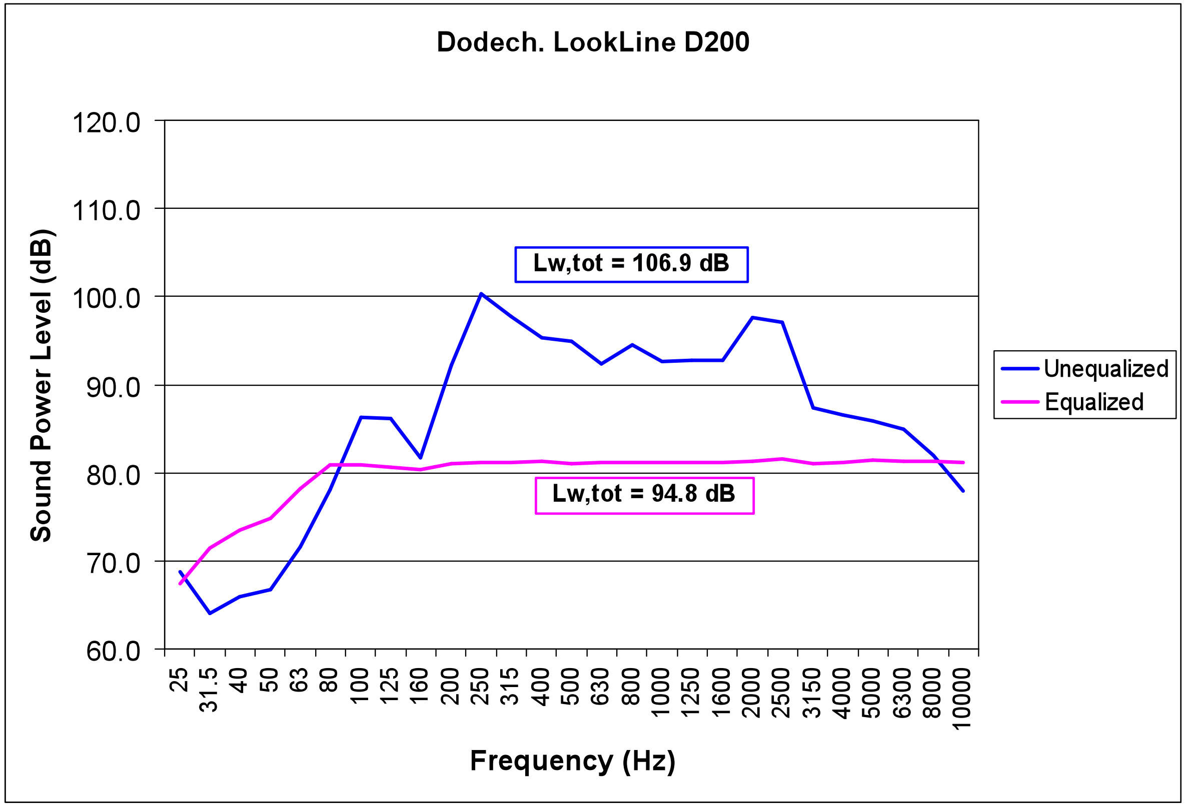
Edirol FA-101
Firewire sound card:
10 in / 10 out
24 bit, 192 kHz
ASIO and WDM

Hardware: loudspeaker & microphone



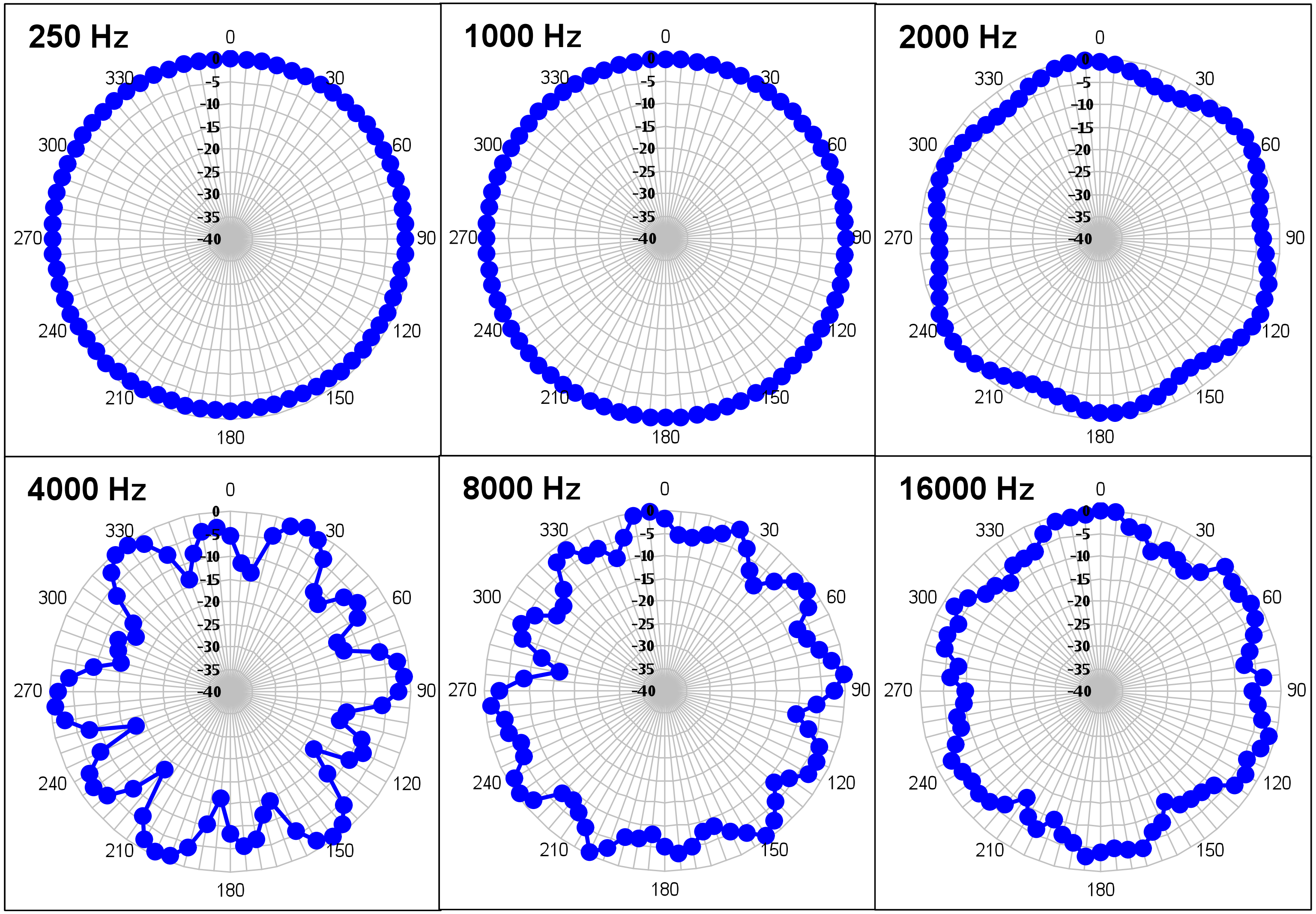
Loudspeakers

- Equalized, omnidirectional sound source:
  - Dodechaedron for mid-high frequencies
  - One-way Subwoofer (<120 Hz)



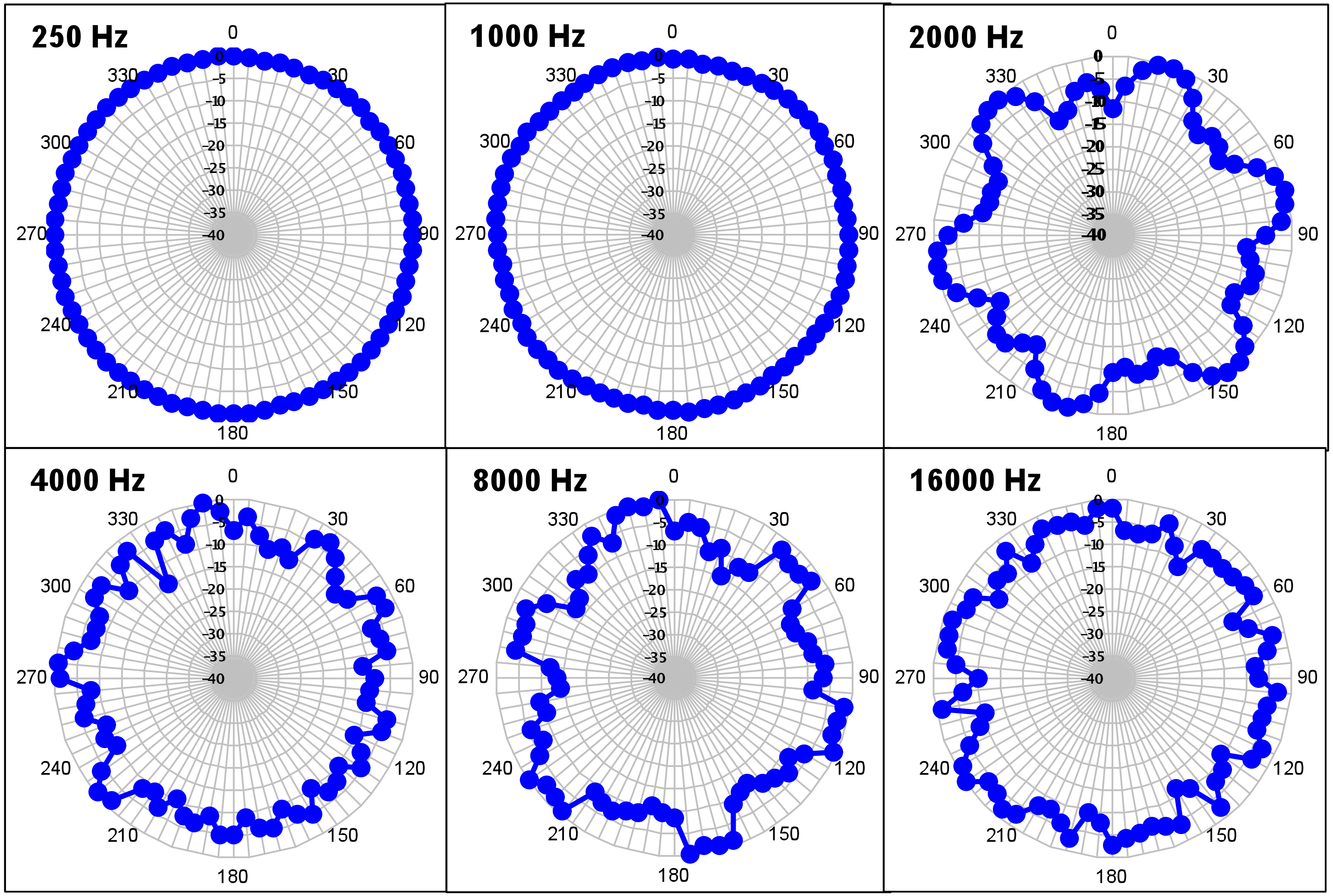
Directivity of loudspeakers

LookLine D-200 dodechaedron



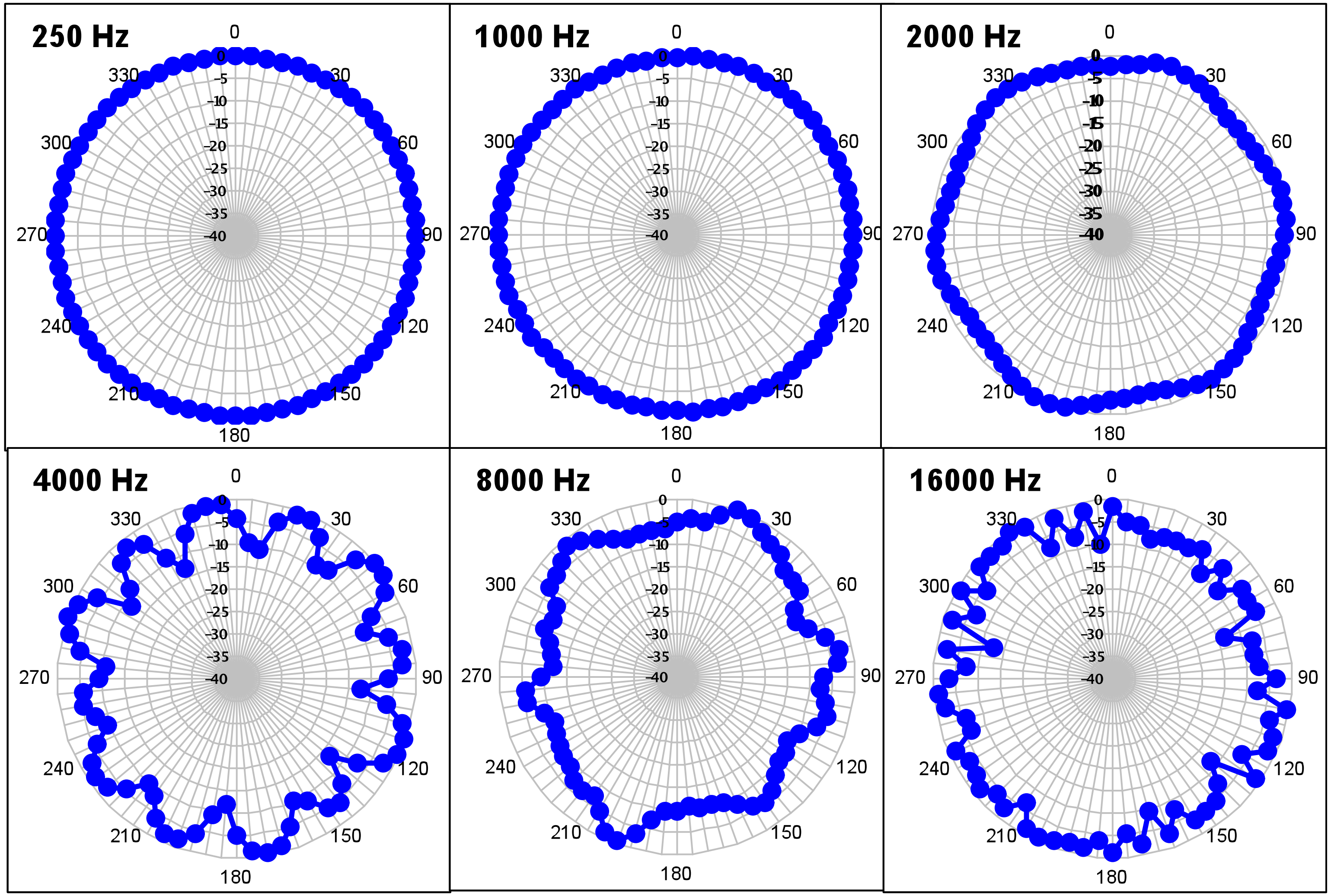
Directivity of loudspeakers

LookLine D-300 dodechaedron

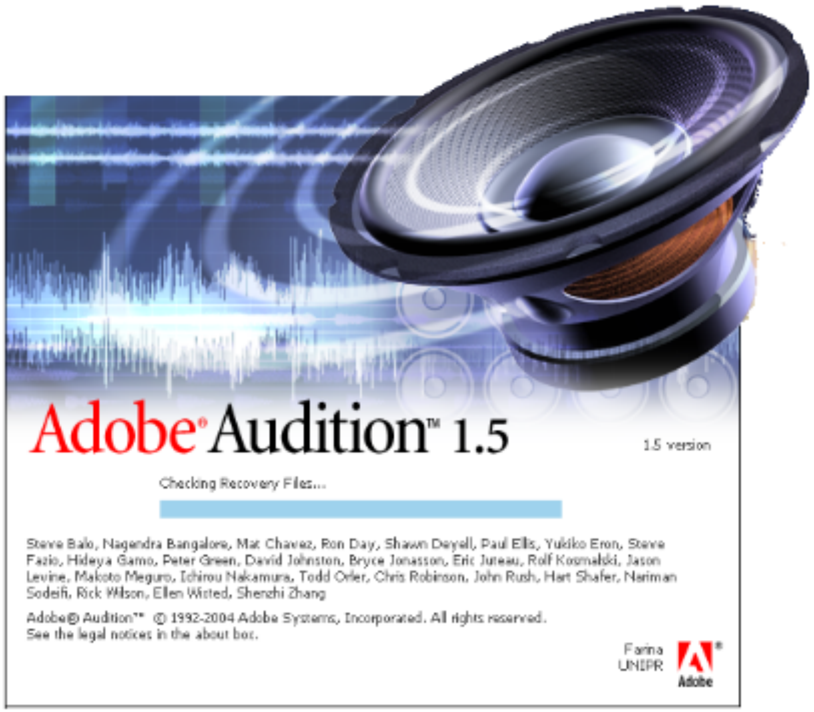


Directivity of loudspeakers

Omnisonic 1000 dodechaedron



The first ESS system - AURORA

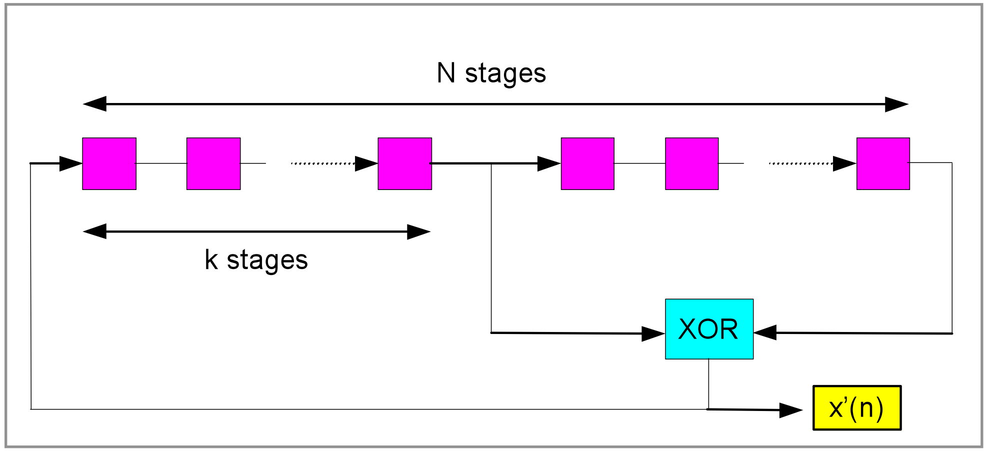


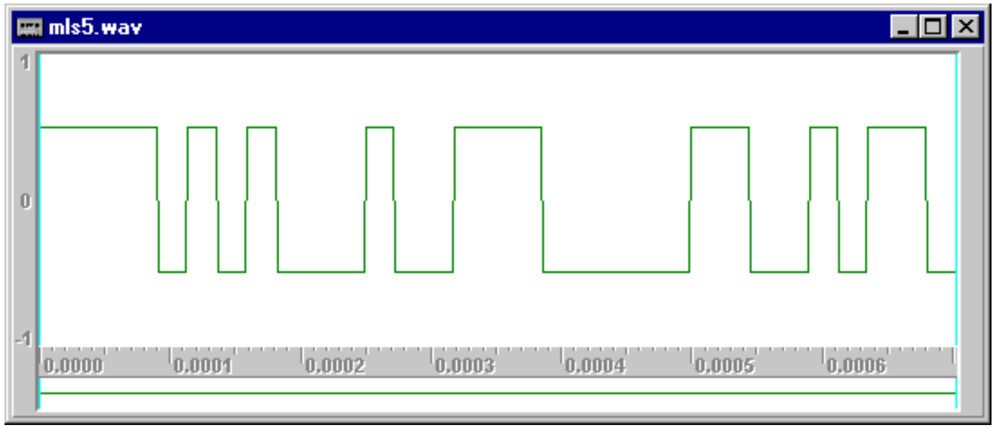
Aurora Plugins

Generate MLS	
Deconvolve MLS	
Generate Sweep	
Deconvolve Sweep	
Convolution	
Kirkeby Inverse Filter	
Speech Transm. Index	

- Aurora was the first measurement system based on standard sound cards and employing the Exponential Sine Sweep method (1998)
- It also works with traditional TDS and MLS methods, so the comparison can be made employing exactly the same hardware

MLS method





- X(t) is a periodic binary signal obtained with a suitable shift -register, configured for maximum lenght of the period

$$L = 2^N - 1$$

MLS deconvolution

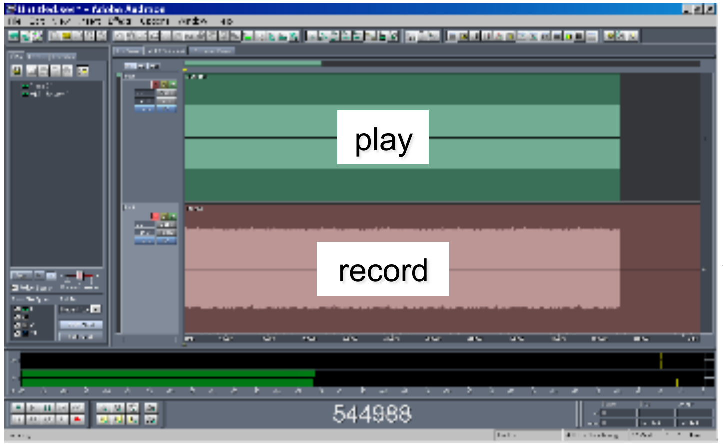
- The re-recorded signal y(i) is cross-correlated with the excitation signal thanks to a fast Hadamard transform. The result is the required impulse response h(i), if the system was linear and time -invariant

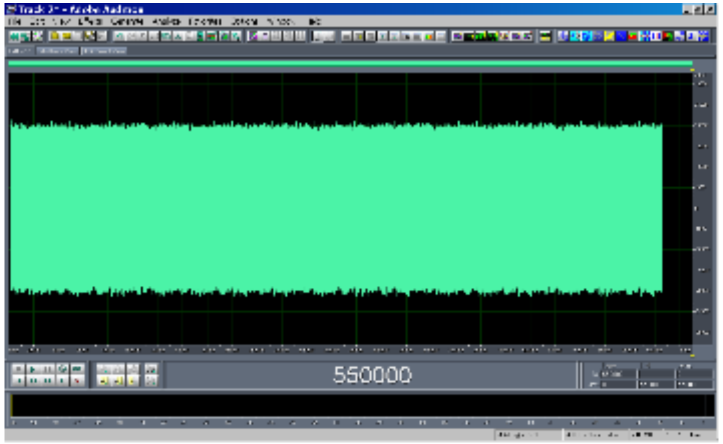
$$h = \frac{1}{L+1} \cdot \tilde{M} \cdot y$$

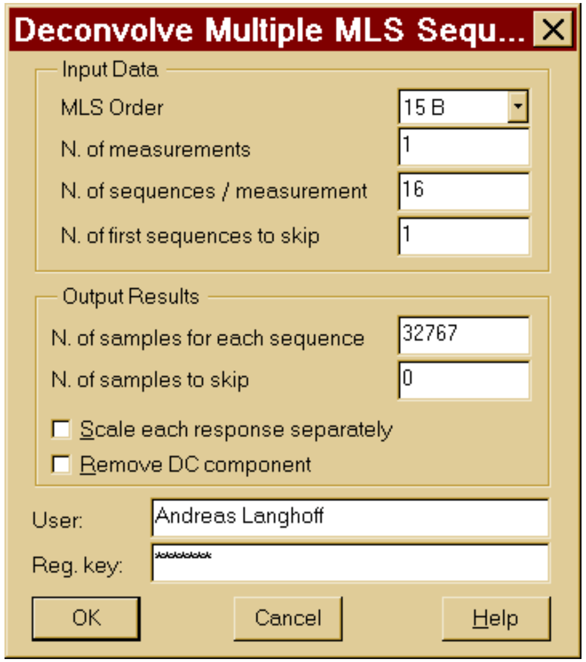
- Where M is the Hadamard matrix, obtained by permutation of the original MLS sequence m(i)

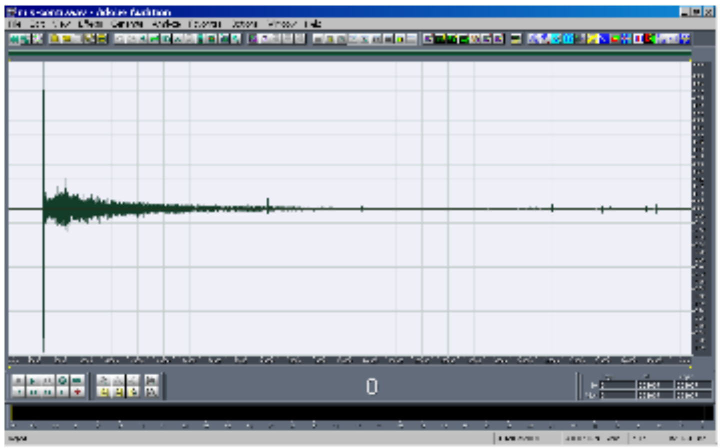
$$\tilde{M}(i,j) = m\left[\left(i + j - 2\right) \mathrm{mod}\, L\right] - 1$$

MLS measurement procedure

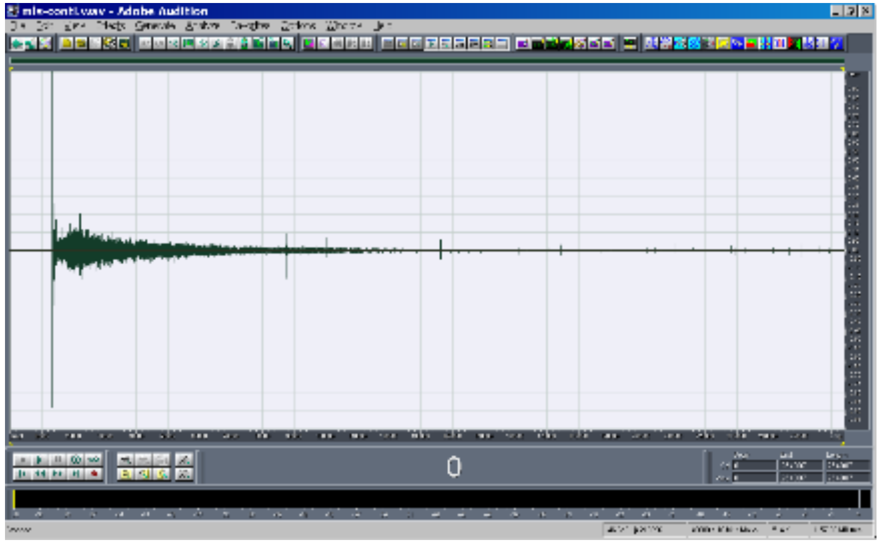


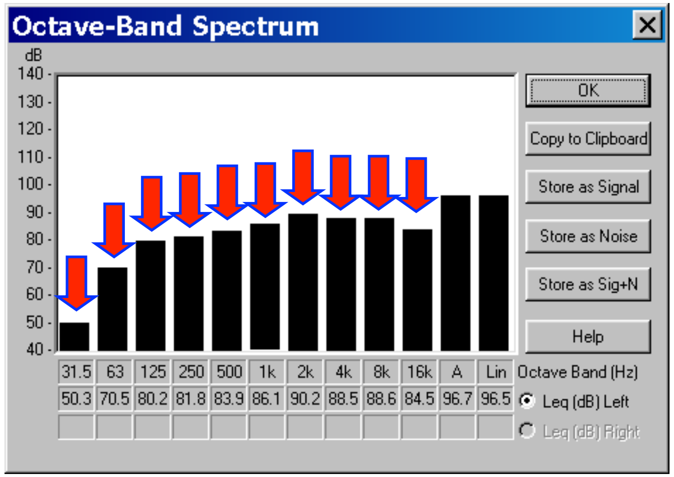


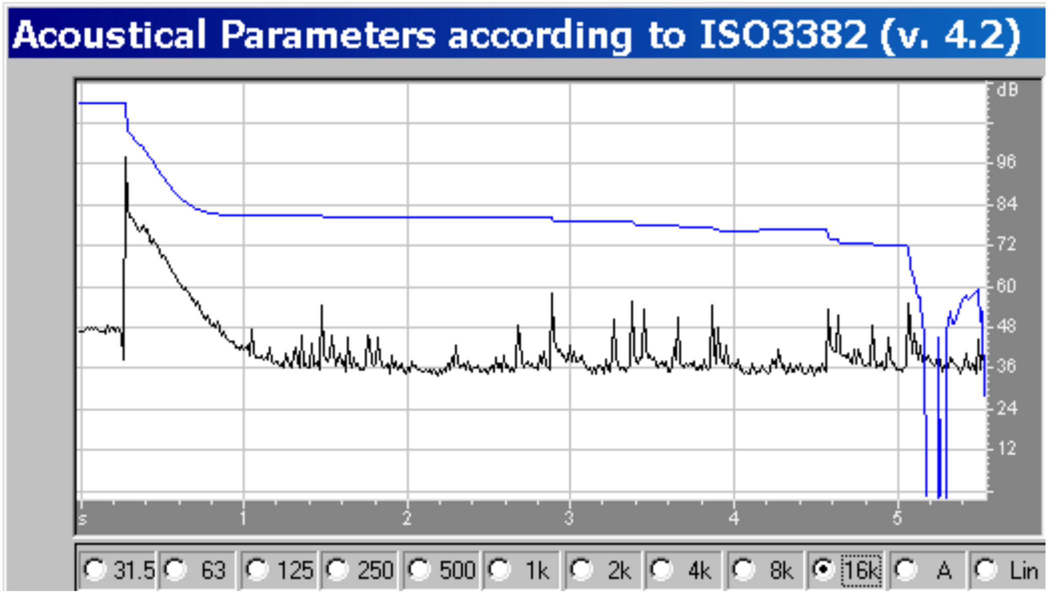




Example of a MLS impulse response





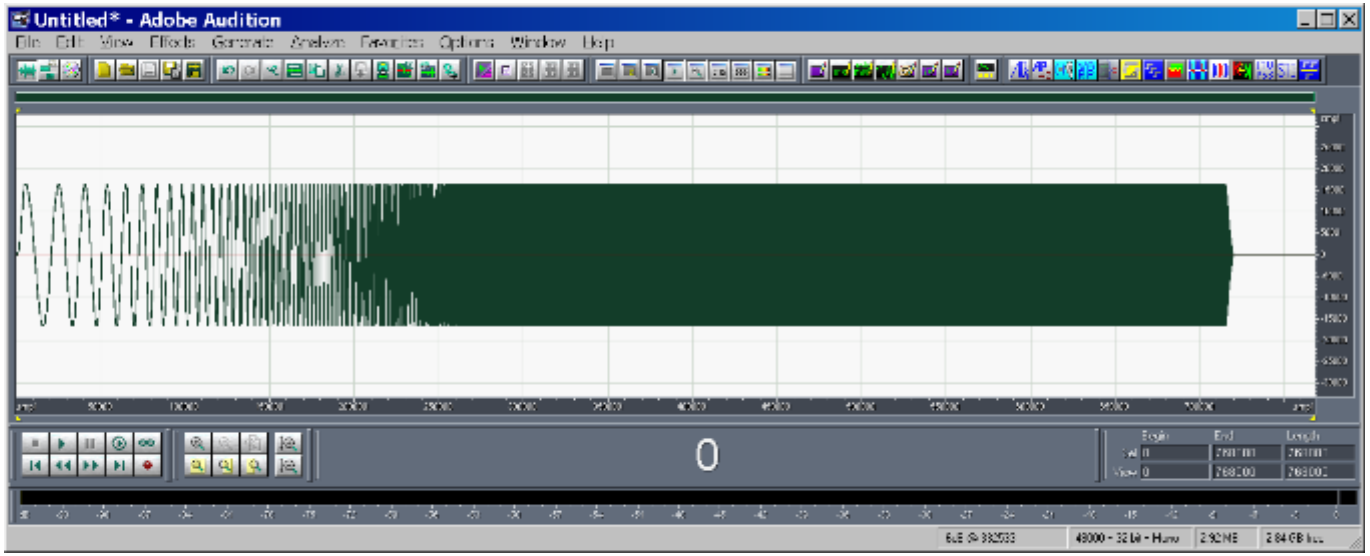


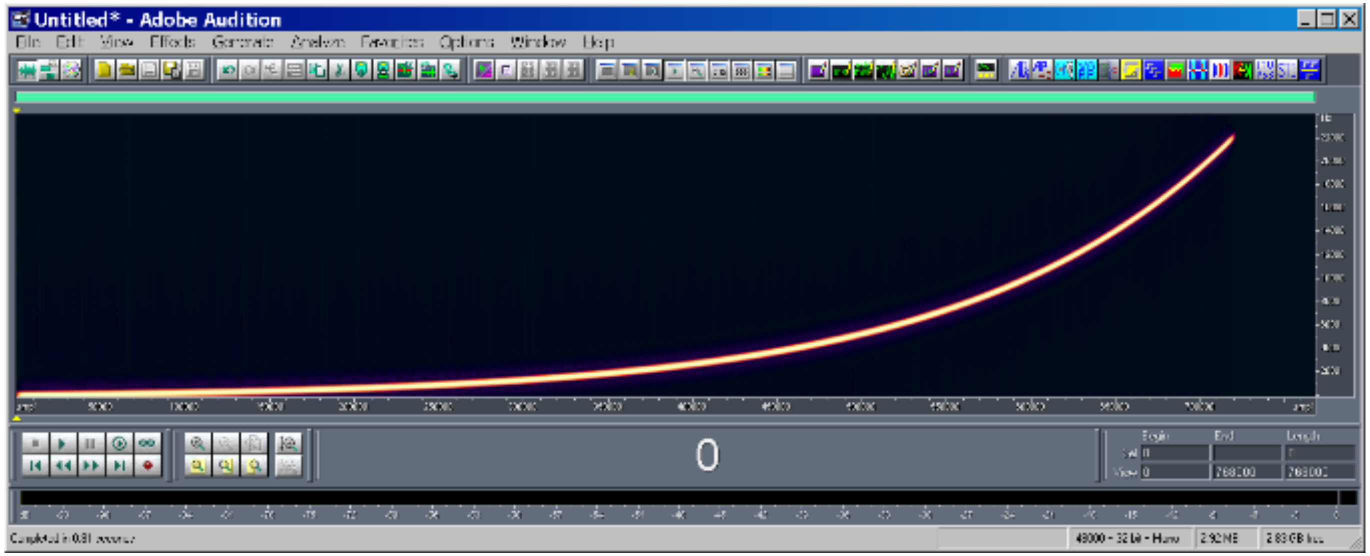
Exponential Sine Sweep method

- x(t) is a band-limited sinusoidal sweep signal, which frequency is varied exponentially with time, starting at f_1 and ending at f_2.

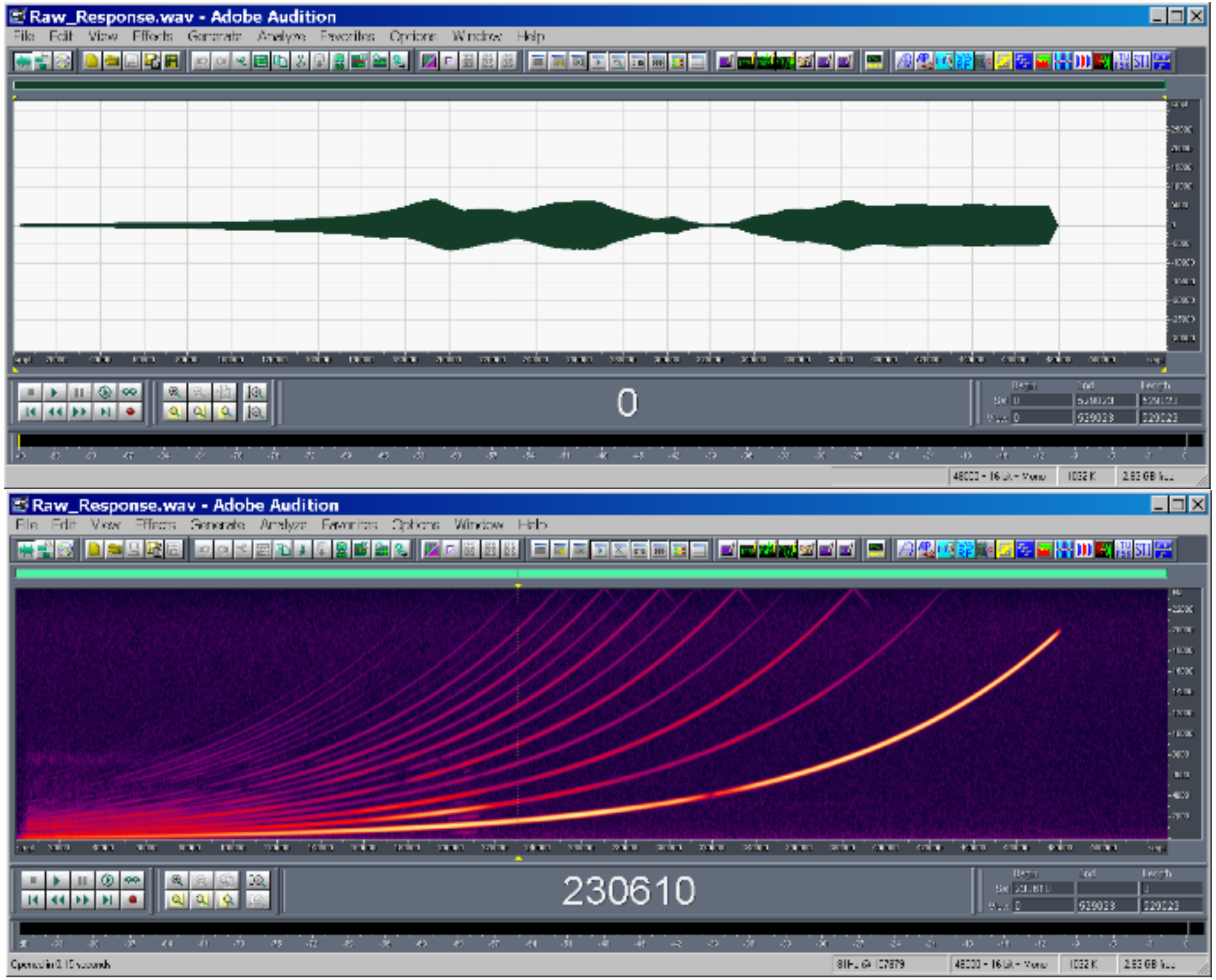
$$x(t) = \sin\left[\frac{2\cdot\pi\cdot f_1\cdot T}{\ln\left(\frac{f_2}{f_1}\right)}\cdot\left(e^{\frac{t}{T}\cdot\ln\left(\frac{f_2}{f_1}\right)}-1\right)\right]$$

Test Signal – x(t)



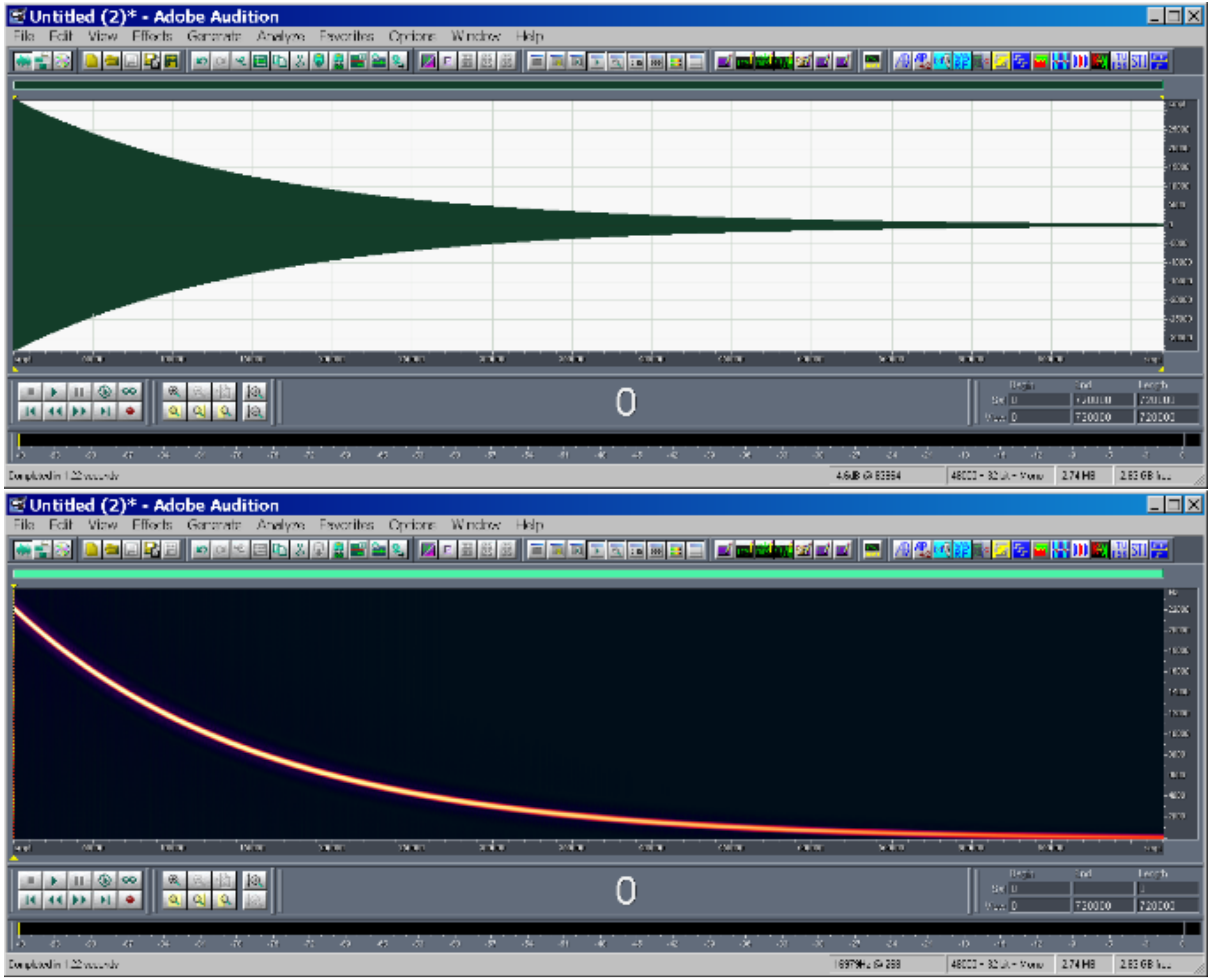


Measured signal - y(t)



- The not-linear behaviour of the loudspeaker causes many harmonics to appear

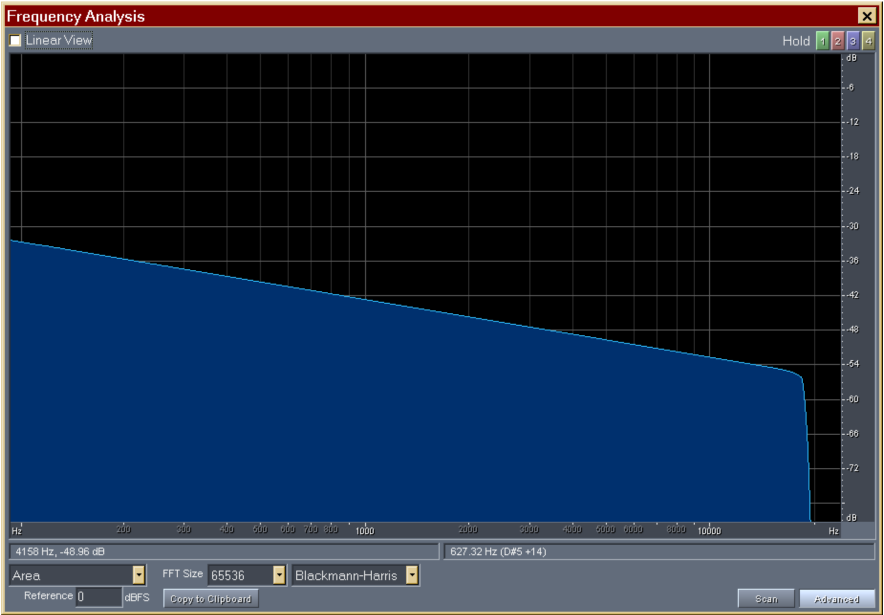
Inverse Filter – z(t)



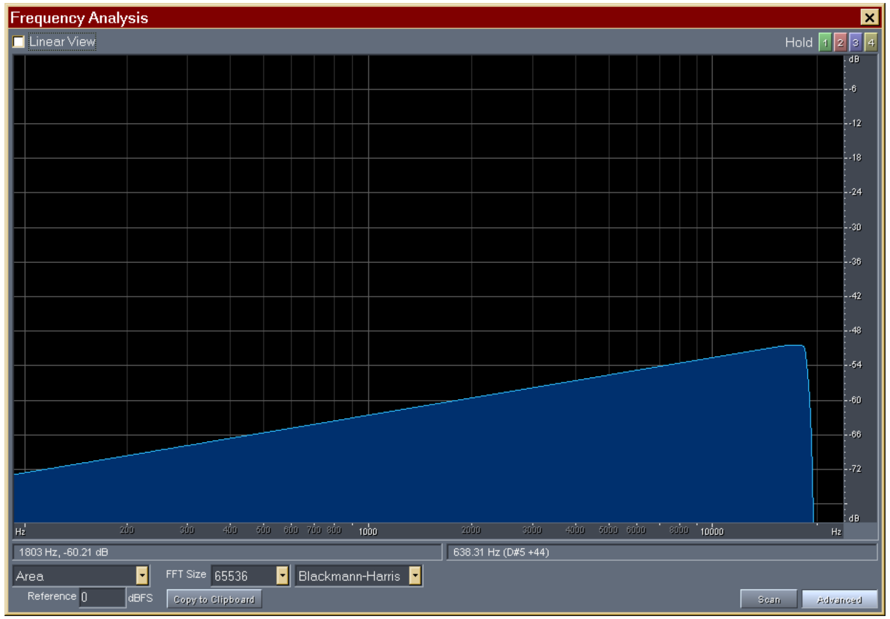
The deconvolution of the IR is obtained convolving the measured signal y(t) with the inverse filter z(t) [equalized, time-reversed x(t)]

Deconvolution of Exponential Sine Sweep

The "time reversal mirror" technique is employed: the system's impulse response is obtained by convolving the measured signal y(t) with the time-reversal of the test signal x(-t). As the log sine sweep does not have a "white" spectrum, proper equalization is required

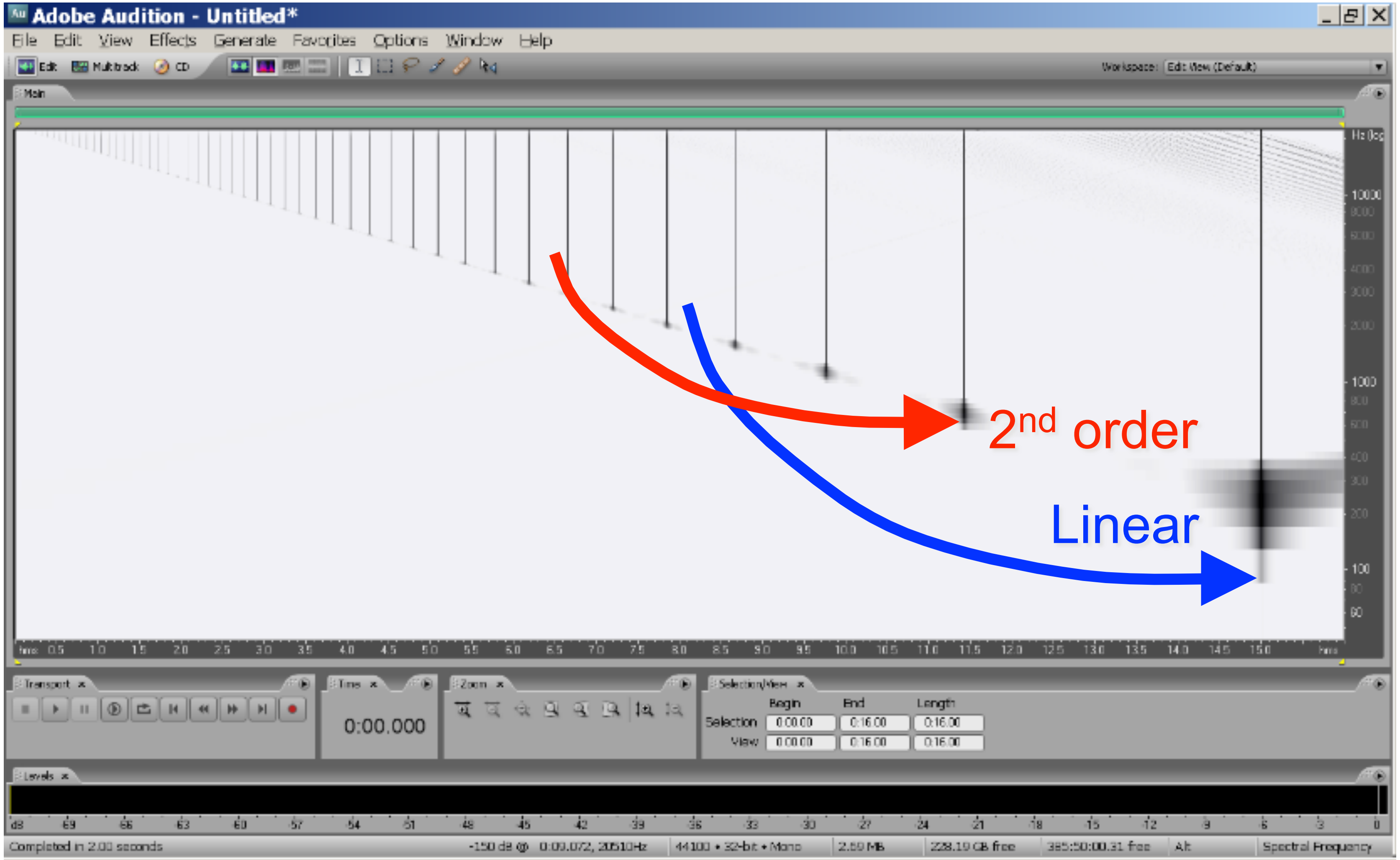


Test Signal x(t)



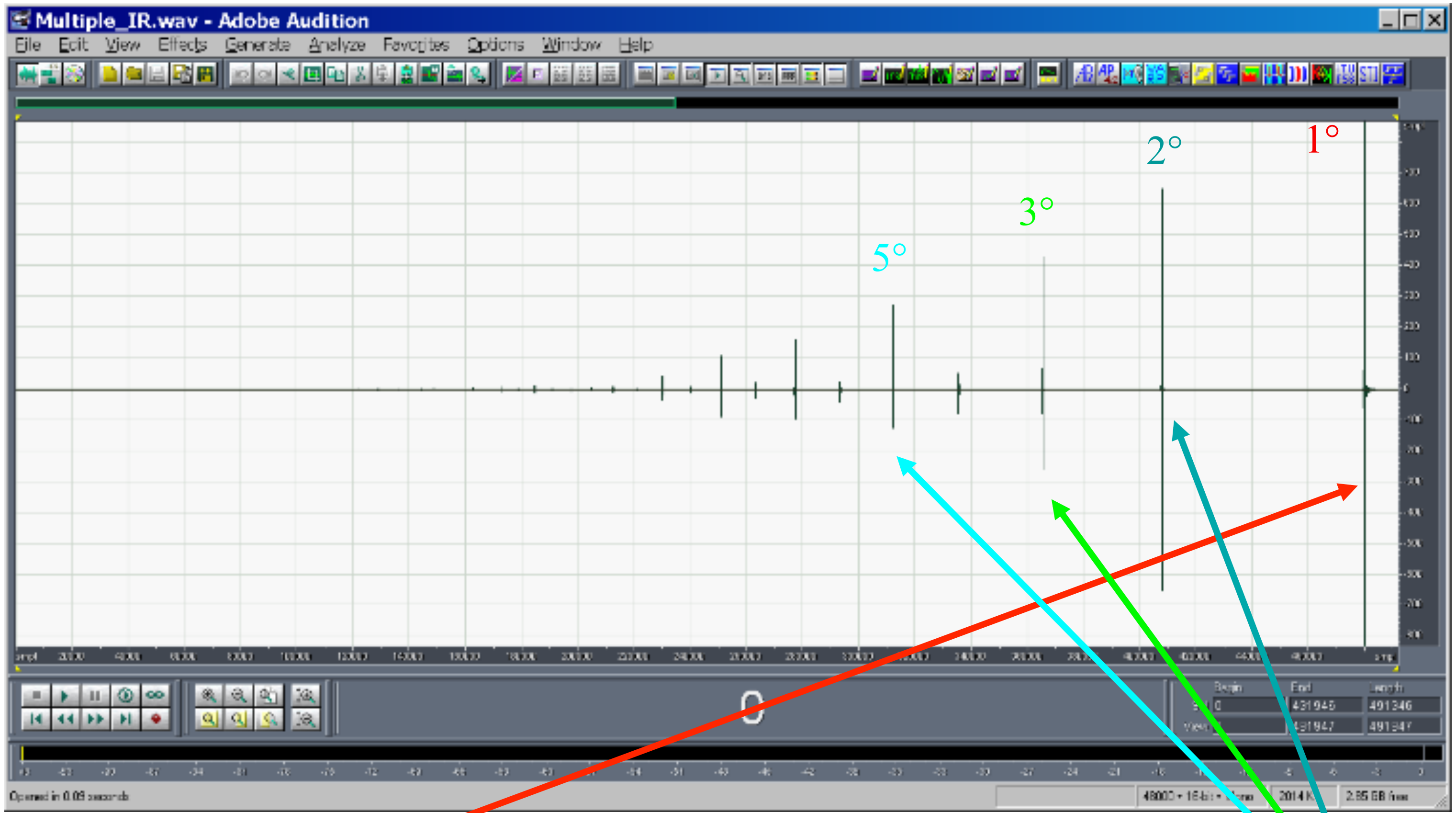
Inverse Filter z(t)

Deconvolution = sonogram rotation



- Convolution with inverse filter rotates the time-frequency plane counter-clockwise

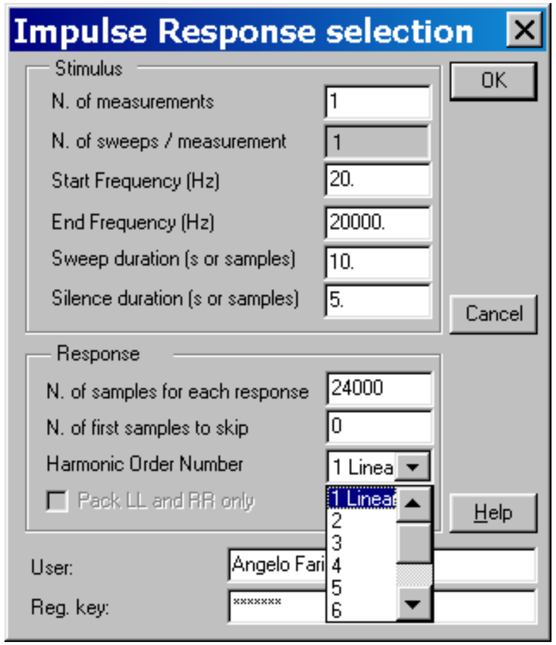
Result of the deconvolution

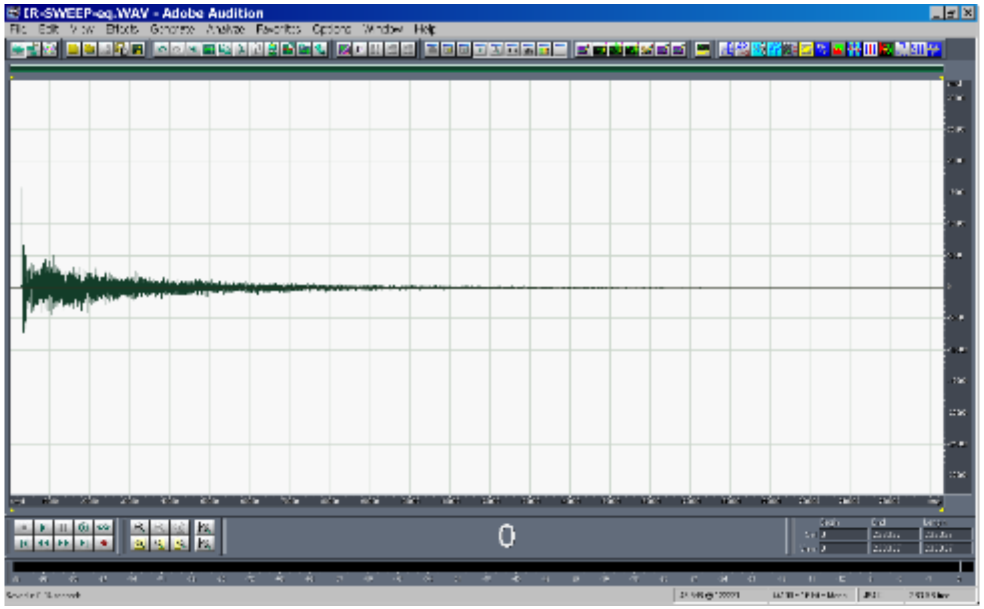


The last impulse response is the linear one, the preceding are the harmonics distortion products of various orders

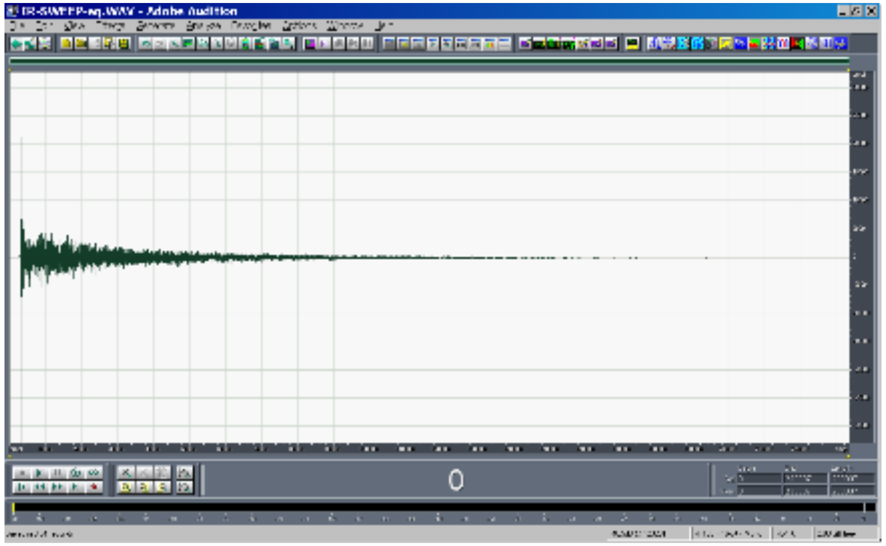
IR Selection

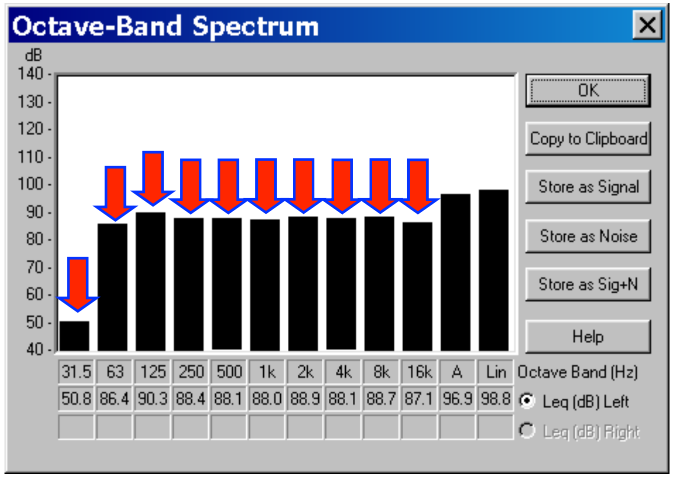
- After the sequence of impulse responses has been obtained, it is possible to select and extract just one of them (the 1°-order - Linear in this example):

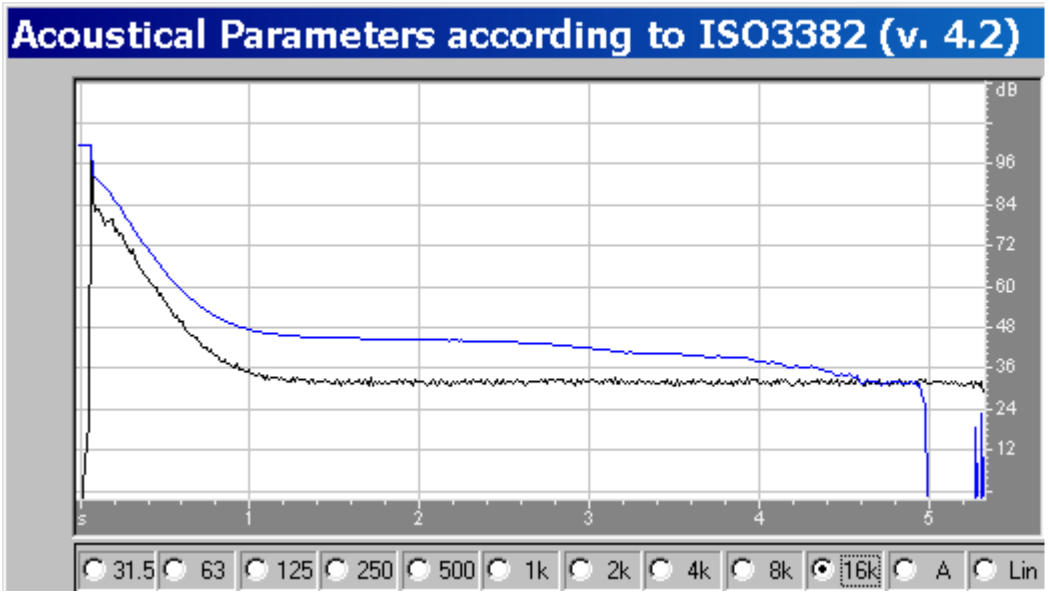




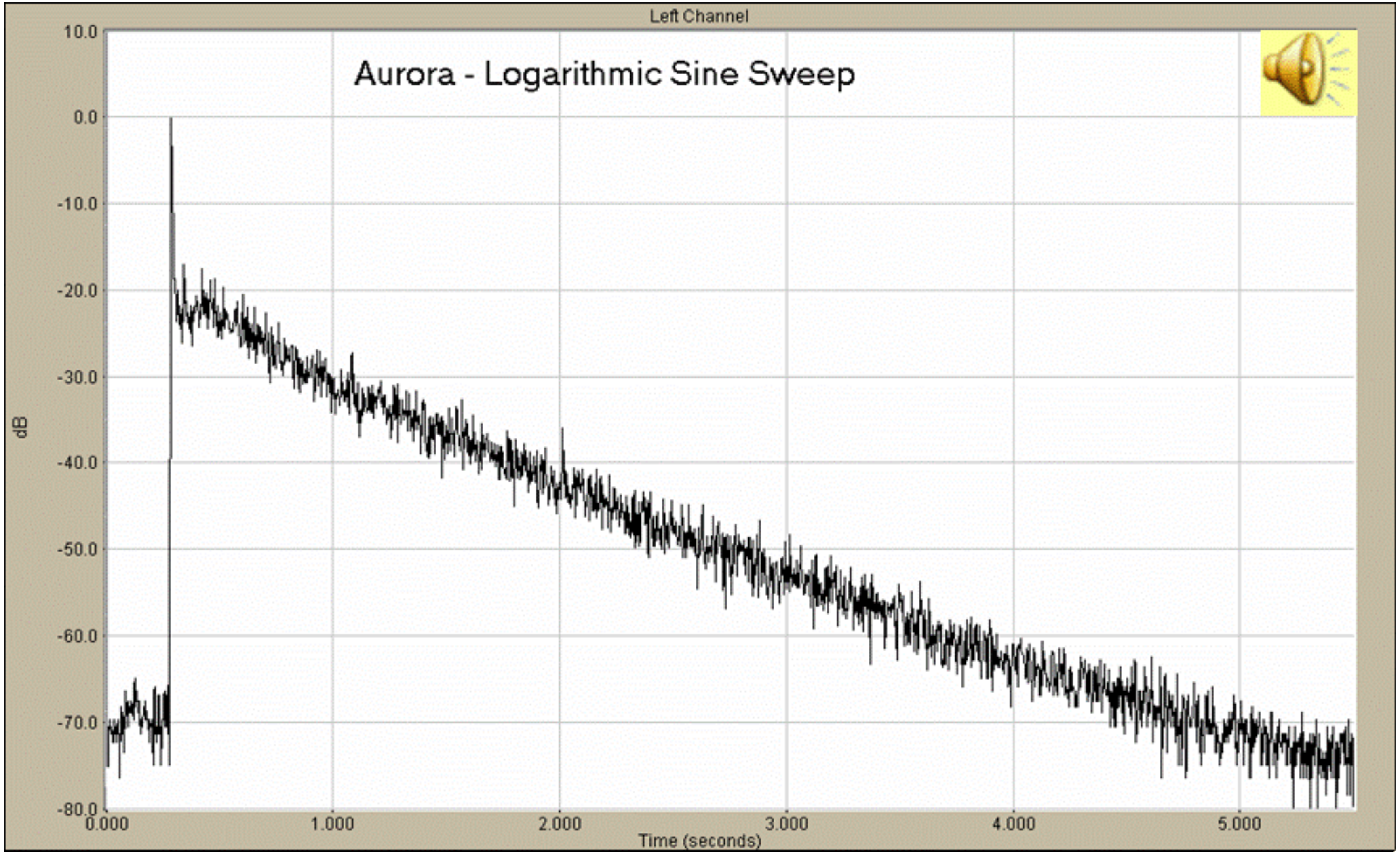
Example of an ESS impulse response







Maximum Length Sequence vs. Exp. Sine Sweep



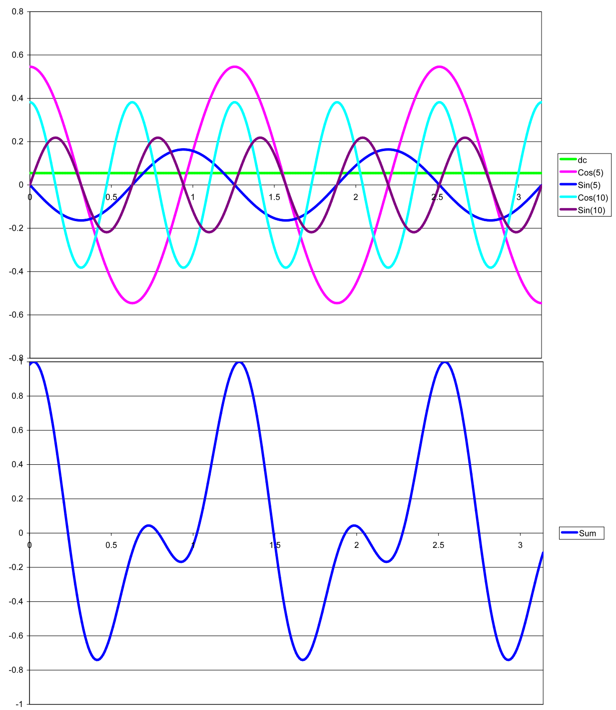
The Future

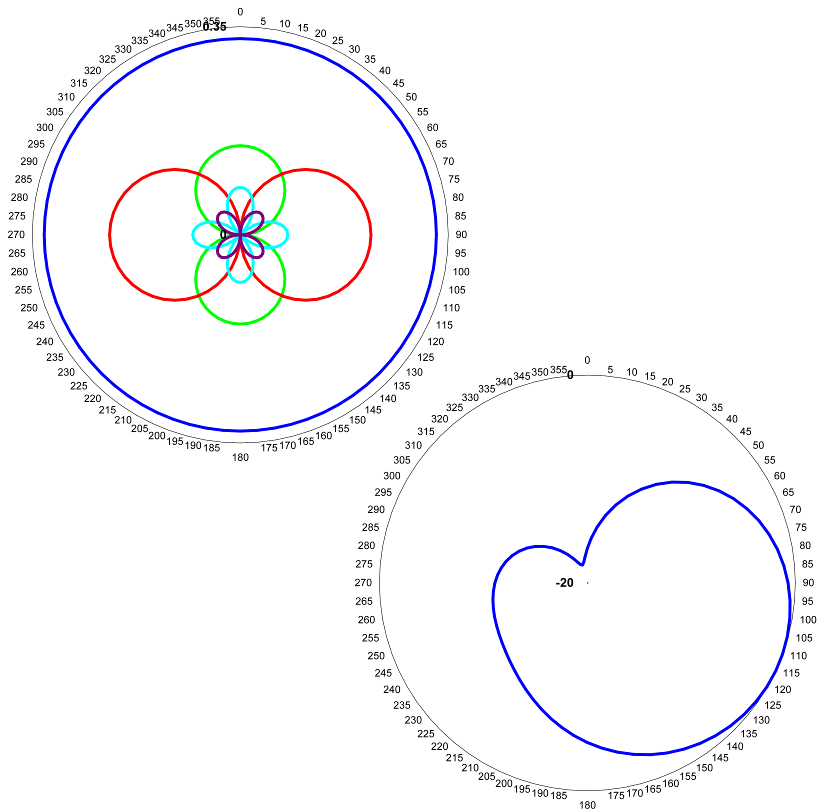
The Future

- Microphone arrays capable of synthesizing aribitrary directivity patterns
- Advanced spatial analysis of the sound field employing spherical harmonics (Ambisonics - 1° order or higher)
- Loudspeaker arrays capable of synthesizing arbitrary directivity patterns
- Generalized solution in which both the directivities of the source and of the receiver are represented as a spherical harmonics expansion

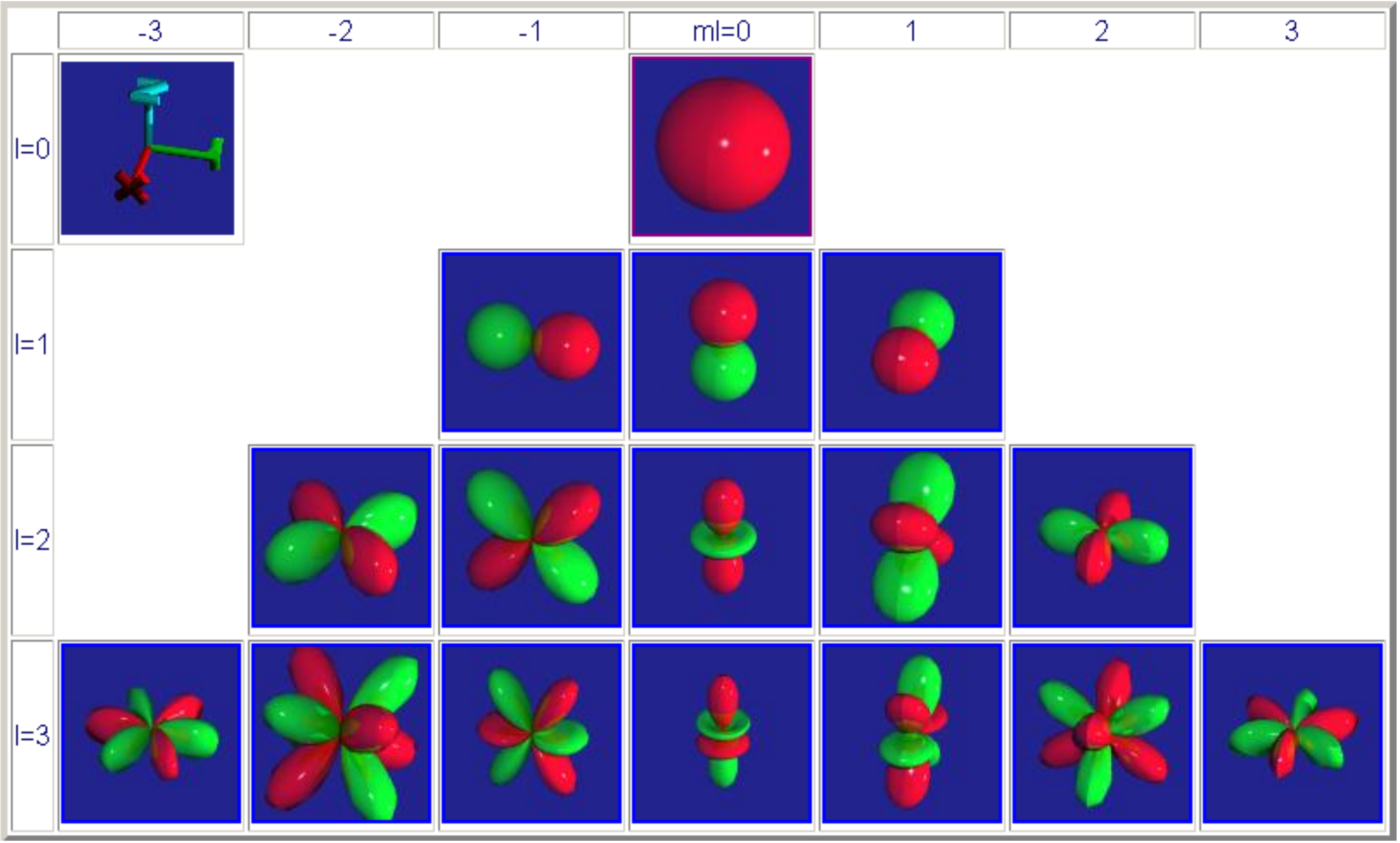
How to get better spatial resolution?

- The answer is simple: analyze the spatial distribution of both source and receiver by means of higher-order spherical harmonics expansion
- Spherical harmonics analysis is the equivalent, in space domain, of the Fourier analysis in time domain
- As a complex time-domain waveform can be though as the sum of a number of sinusoidal and cosinusoidal functions, so a complex spatial distribution around a given notional point can be expressed as the sum of a number of spherical harmonic functions



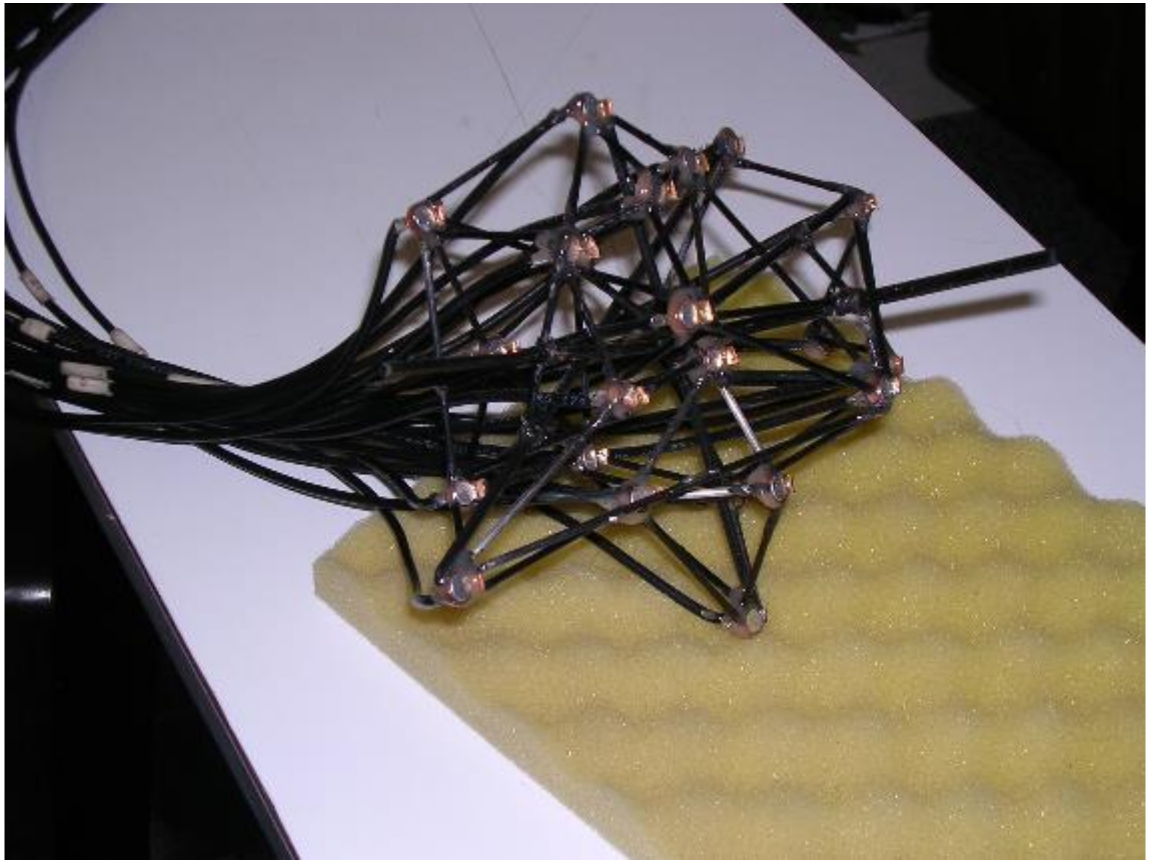


Higher-order spherical harmonics expansion



3°-order microphone (Trinnov - France)

- Arnoud Laborie developed a 24-capsule compact microphone array - by means of advanced digital filtering, spherical ahrmonic signals up to 3° order are obtained (16 channels)



4°-order microphone (France Telecom)

- Jerome Daniel and Sebastien Moreau built samples of 32-capsules spherical arrays - these allow for extractions of microphone signals up to 4° order (25 channels)

4°-order microphone (Italy)

- Angelo Farina's spherical mike (32 capsules)

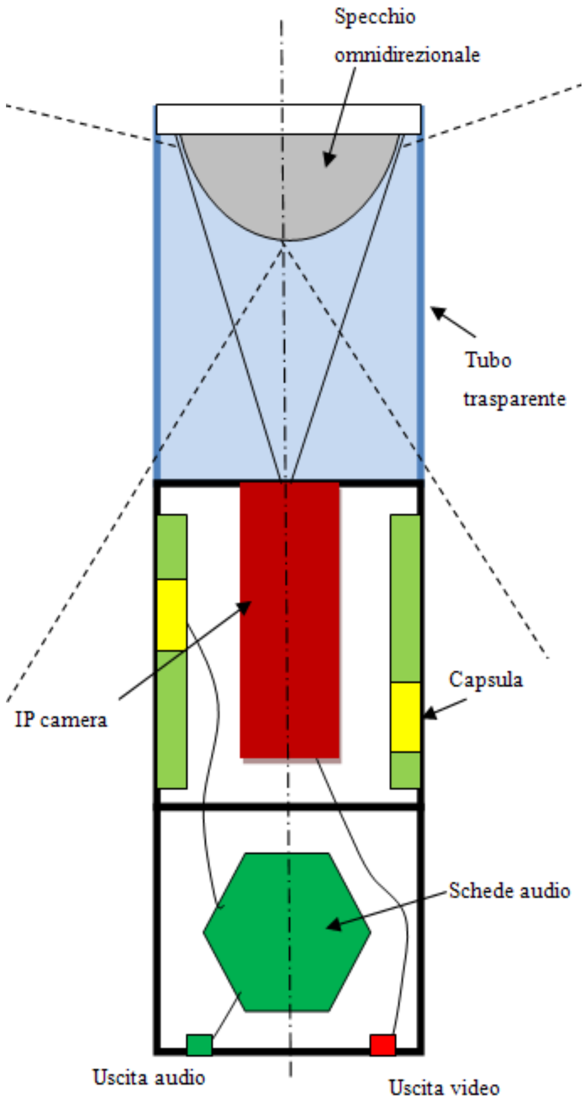
5°-order microphones (University of Sydney)

- Chris Craig's dual-sphere concentrical mike (64 capsules)
- And his 32-capsules cylindrical mike

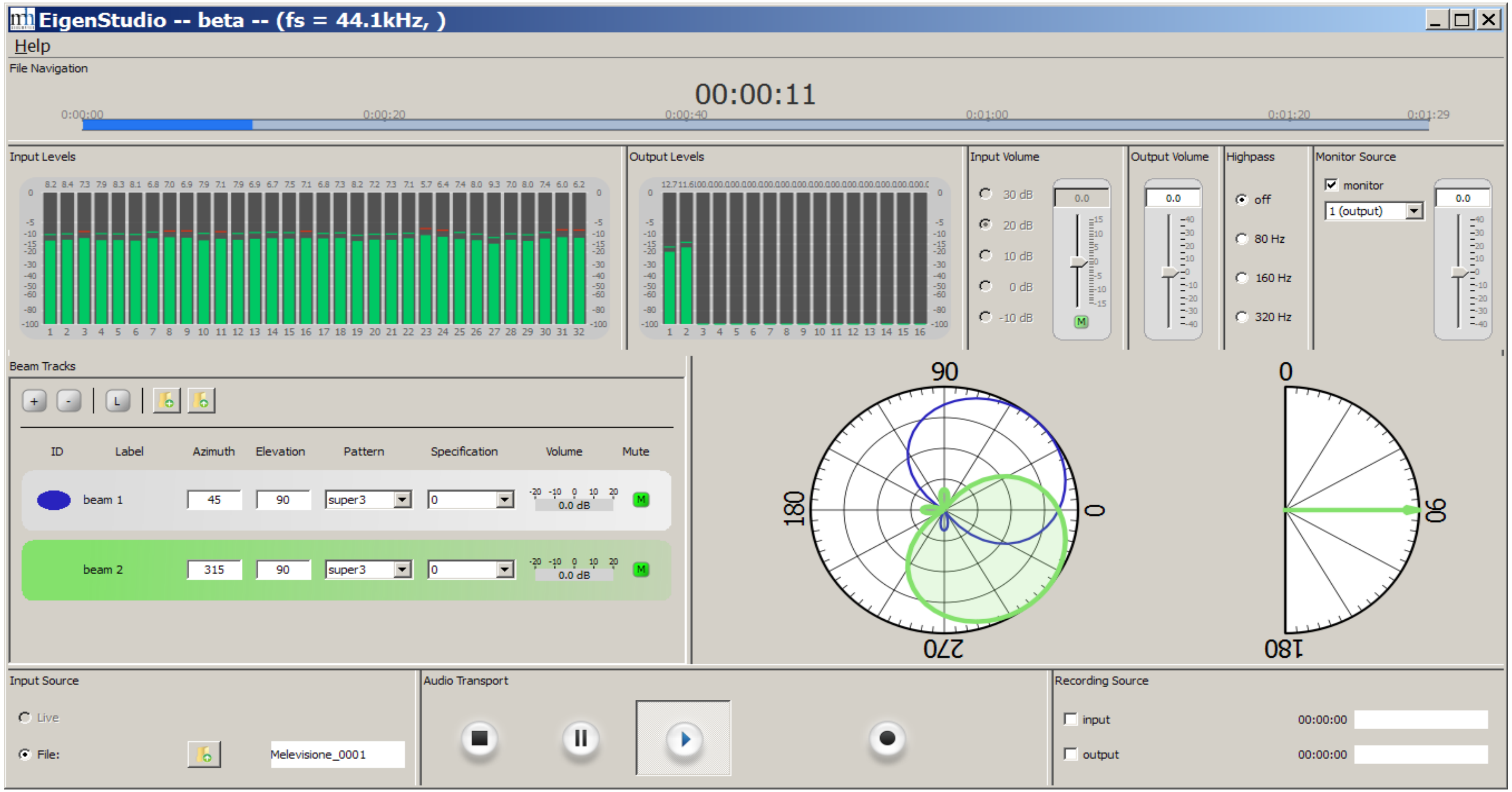
The Eigenmike$^{(TM)}$

- 32 high-quality Sennheiser capsules
- Embedded preamps wityh digital gain control
- Embedded Wolfson converters
- Embedded AV-net Audio-Over-Ethernet chip
- Power Over Ethernet
- EMIB Firewire interface with TCAT-II chipset
- Drivers for Windows, Mac, Linux
- "Virtual Microphone" Software

The 3D Cylindrical Microphone Array

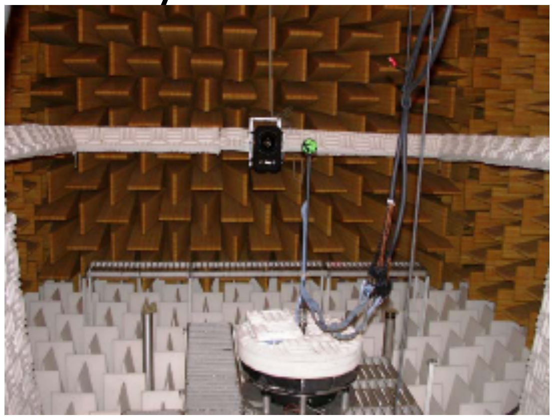


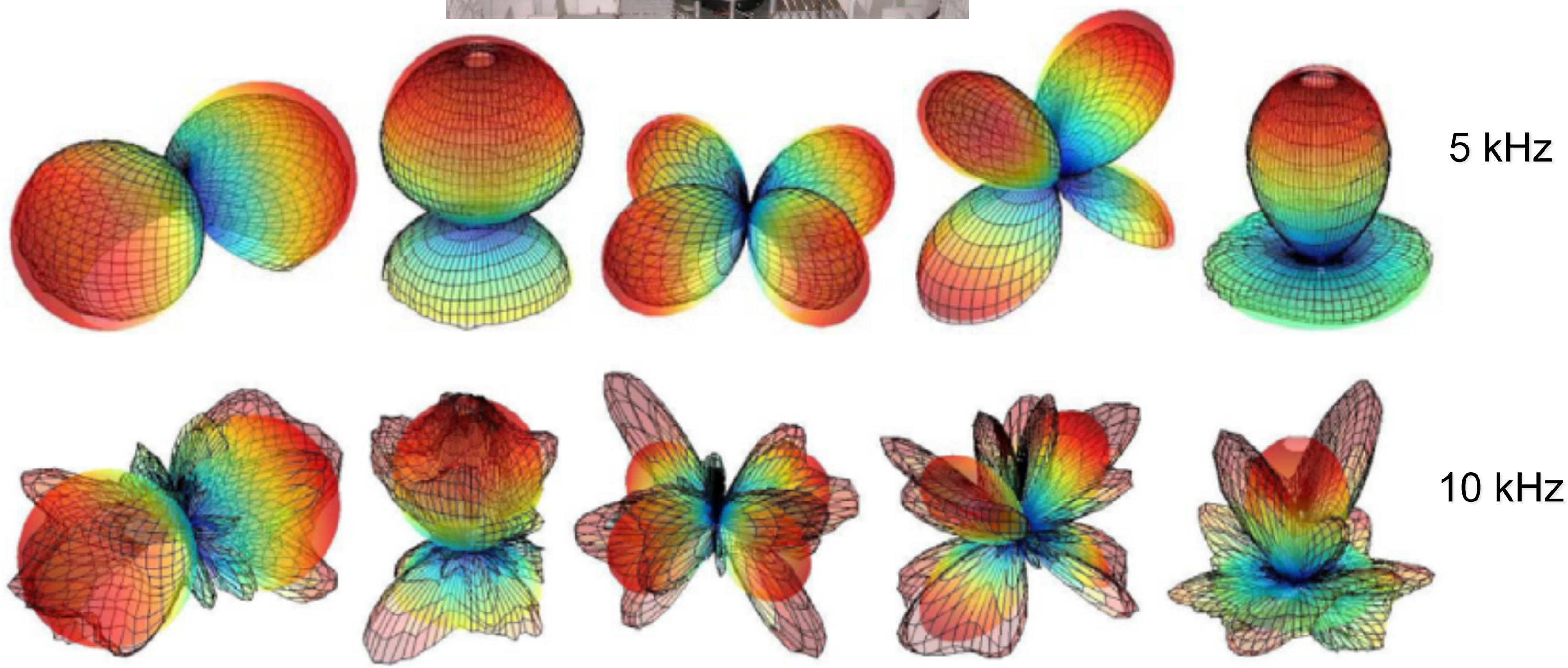
The EIGENMIKE™ software



Verification of high-order patterns

- Sebastien Moreau and Olivier Warusfel verified the directivity patterns of their 4°-order microphone array in the anechoic room of IRCAM (Paris)

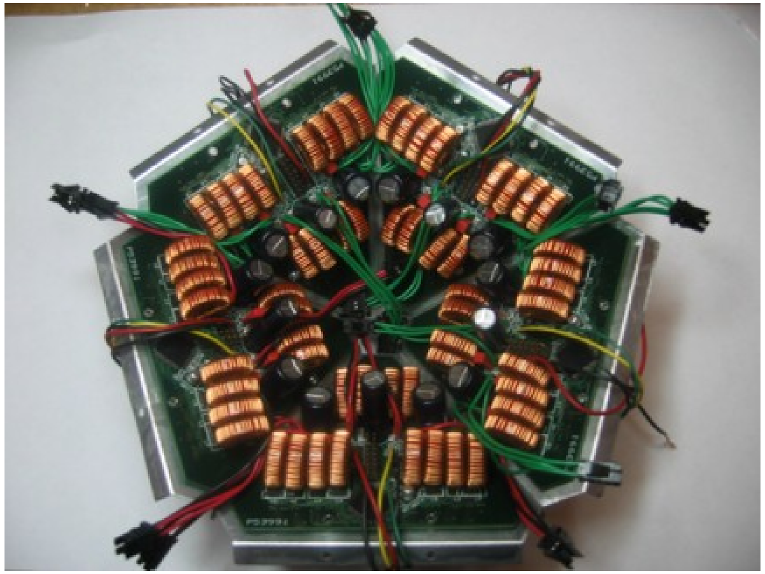




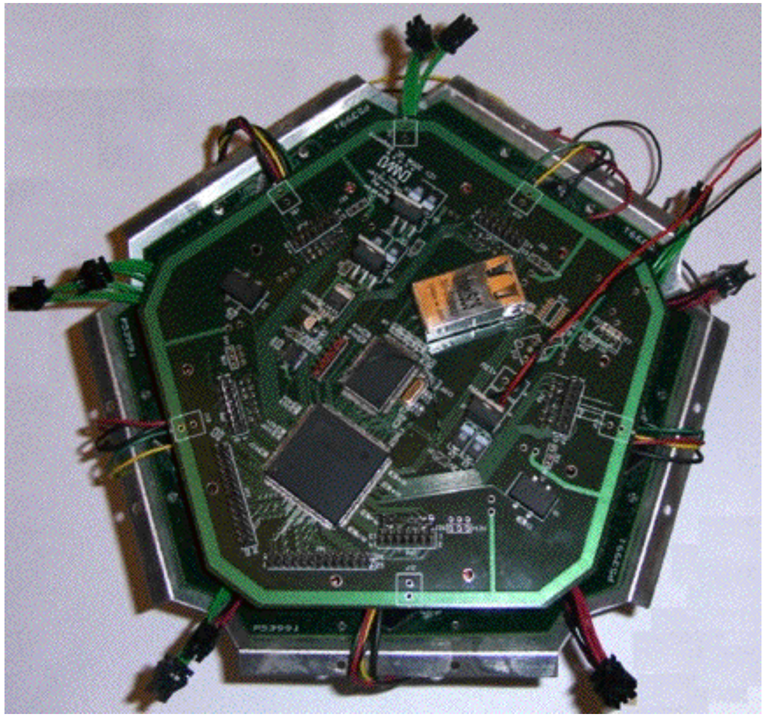
High-order sound source

- Adrian Freed, Peter Kassakian, and David Wessel (CNMAT) developed a new 120-loudspeakers, digitally controlled sound source, capable of synthesizing sound emission according to spherical harmonics patterns up to 5° order.

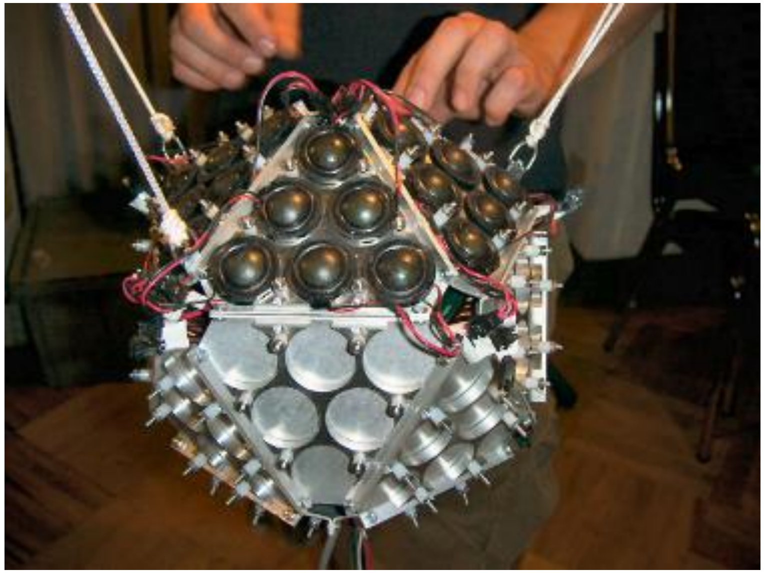
Technical details of high-order source



- Class-D embedded amplifiers



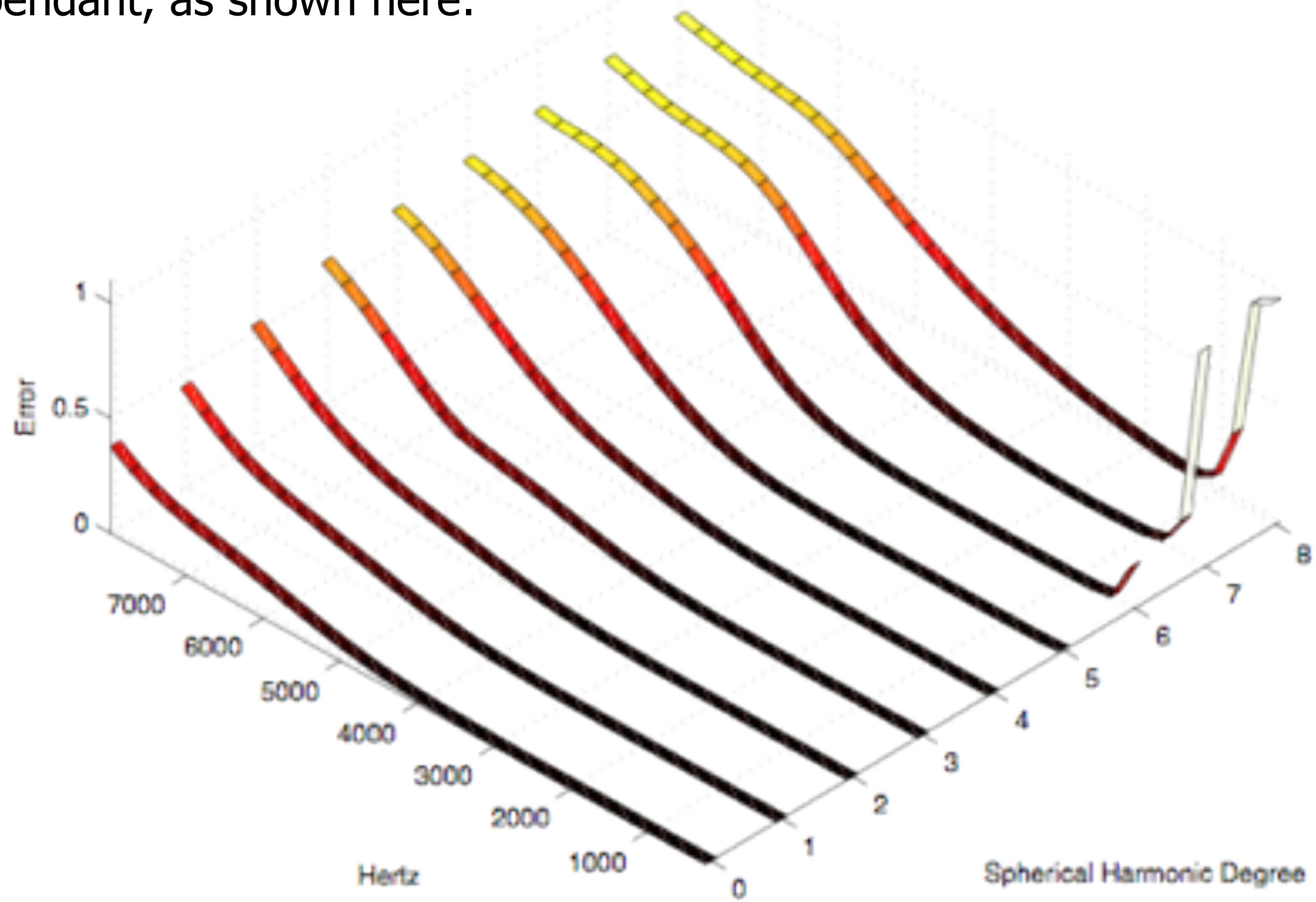
- Embedded ethernet interface and DSP processing



- Long-excursion special Meyer Sound drivers

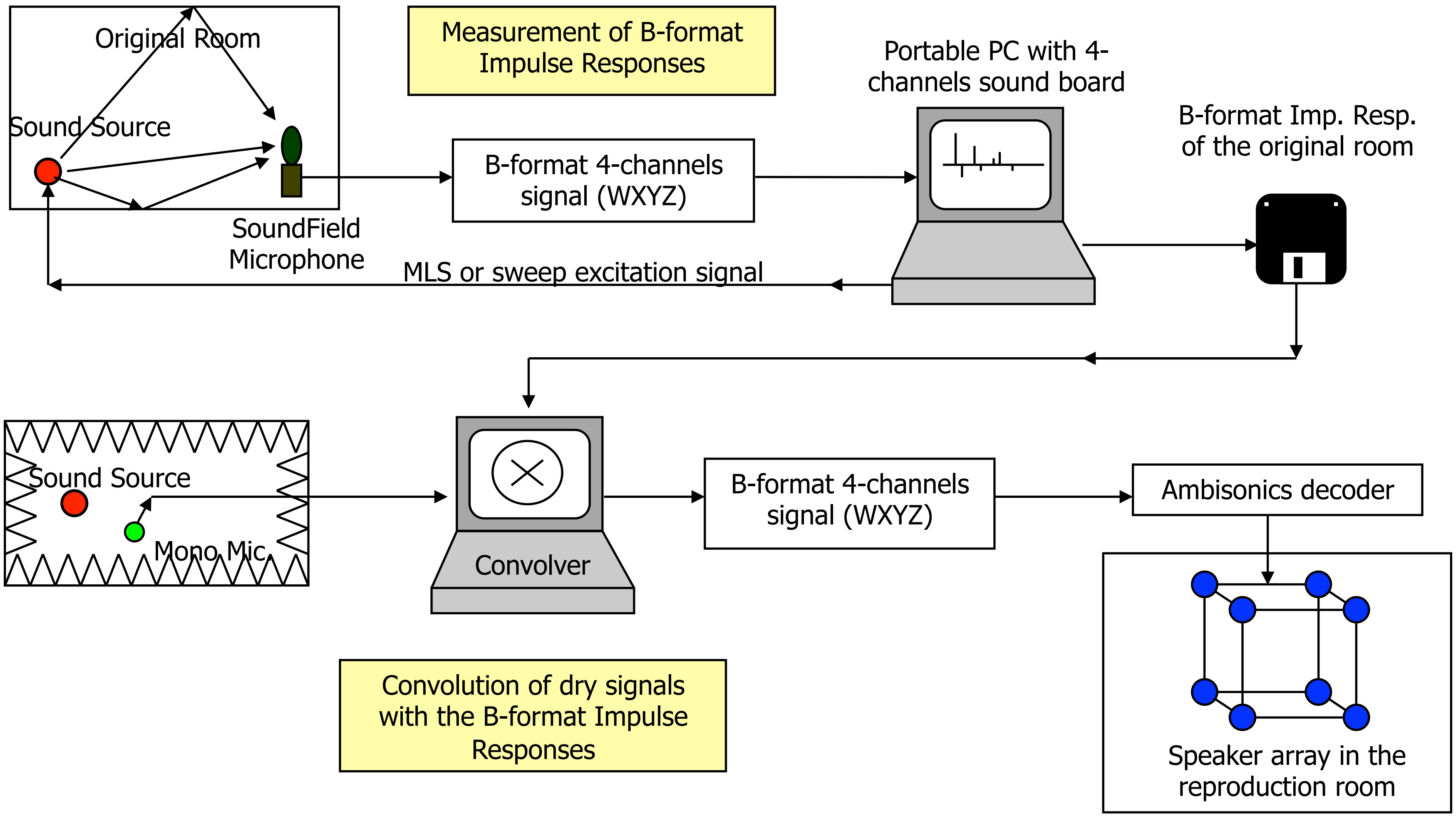
Accuracy of spatial synthesis

- The spatial reconstruction error of a 120-loudspeakers array is frequency dependant, as shown here:



- The error is acceptably low over an extended frequency range up to 5° -order

3D Impulse Response (Gerzon, 1975)

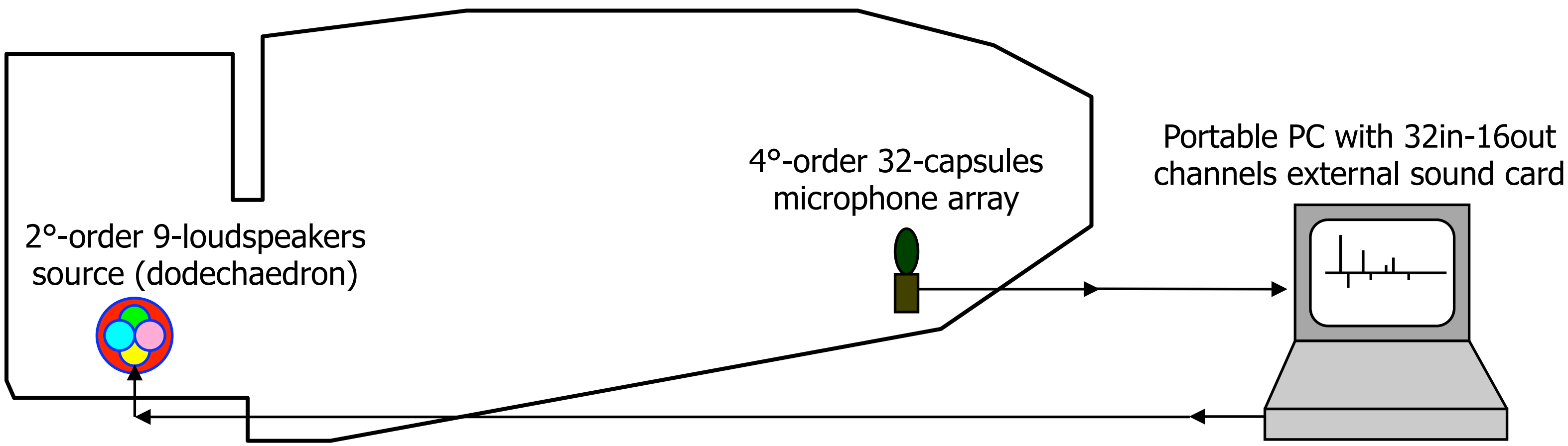


The Waves project (2003)

- The original idea of Michael Gerzon was finally put in practice in 2003, thanks to the Israeli-based company WAVES
- More than 100 theatres all around the world were measured, capturing 3D IRs (4 -channels B-format with a Soundfield microphone)
- The measurments did also include binaural impulse responses, and a circular-array of microphone positions
- More details on WWW.ACOUSTICS.NET

Complete high-order MIMO method

- Employing massive arrays of transducers, it will be feasible to sample the acoustical temporal-spatial transfer function of a room
- Currently available hardware and software tools make this practical only up to 4° order, which means 25 inputs and 25 outputs
- A complete measurement for a given source-receiver position pair takes approximately 10 minutes (25 sine sweeps of 15s each are generated one after the other, while all the microphone signals are sampled simultaneously)
- However, it has been seen that real-world sources can be already approximated quite well with 2°-order functions, and even the human HRTF directivites are reasonally approximated with 3°-order functions.



Teatro "La Scala": spatial analysis of reflections