ASSOCIAZIONE ITALIANA di ACUSTICA

Dipartimento di Energetica

Strumenti
e metodi di misura
per l'acustica e le vibrazioni
Seminario in ricordo di
Eugenio Mattei
Ancona, 21 settembre 2008

La misura della risposta all'impulso per la caratterizzazione di sistemi acustici e vibrazionali

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Time Line

The Past

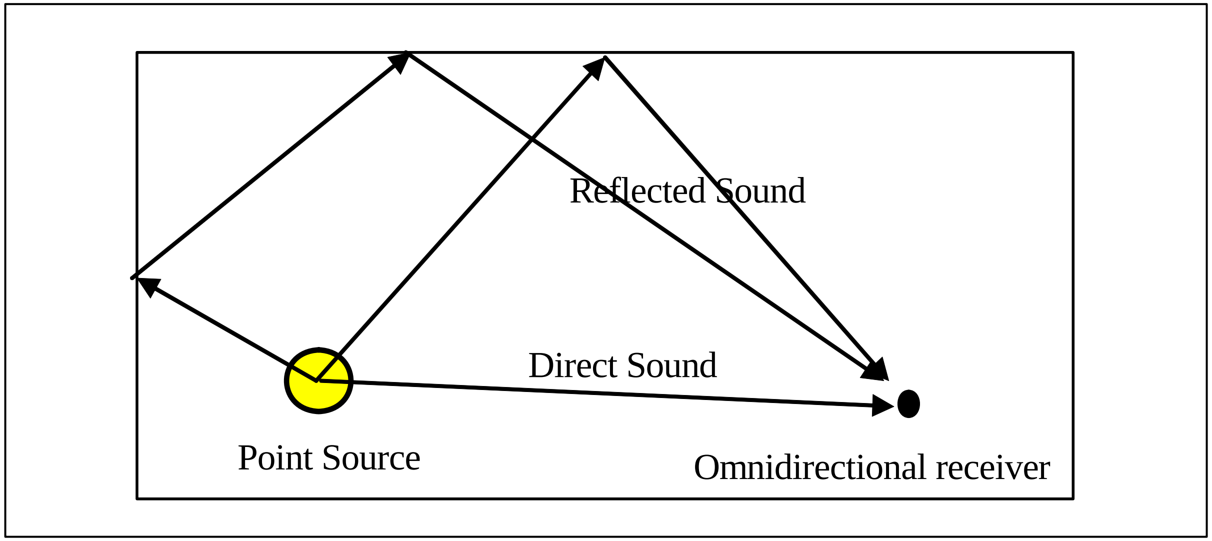
- Traditional time-domain measurements with pulsive sounds and omnidirectional transducers
- MLS and TDS methods for electroacoustical masurements

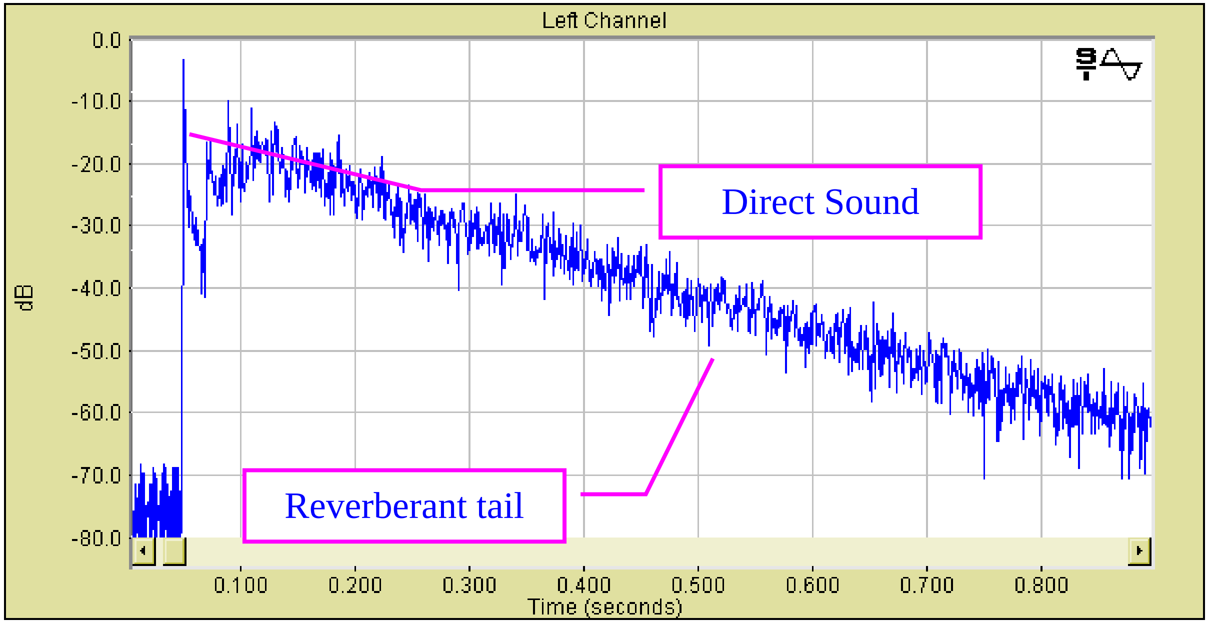
The Present

- Electroacoustical measurements employing the Exponential Sine Sweep method (ESS)
- Capturing spatial information with multichannel microphones
- Post processing of measured IRs for computing acoustical parameters and for auralization

The Past

Starting point: room impulse response

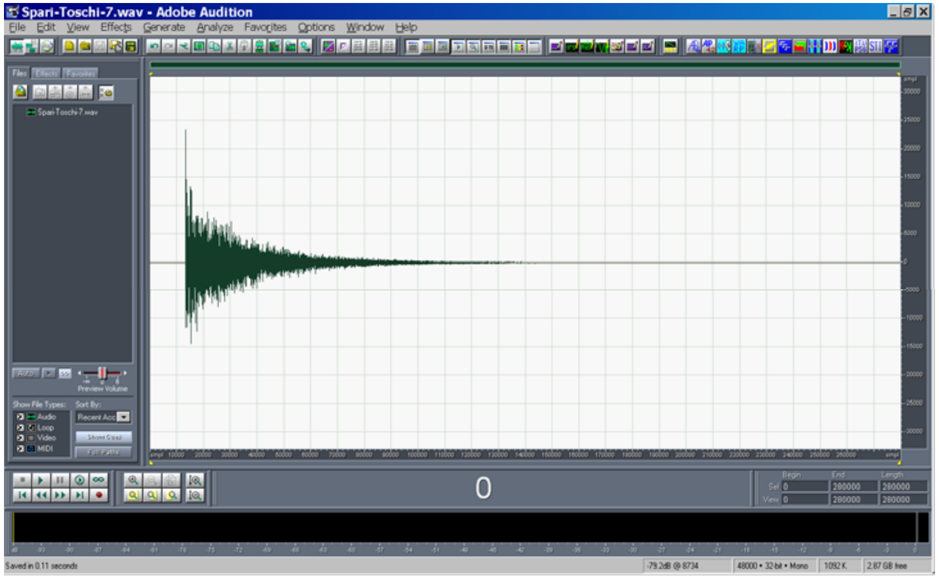


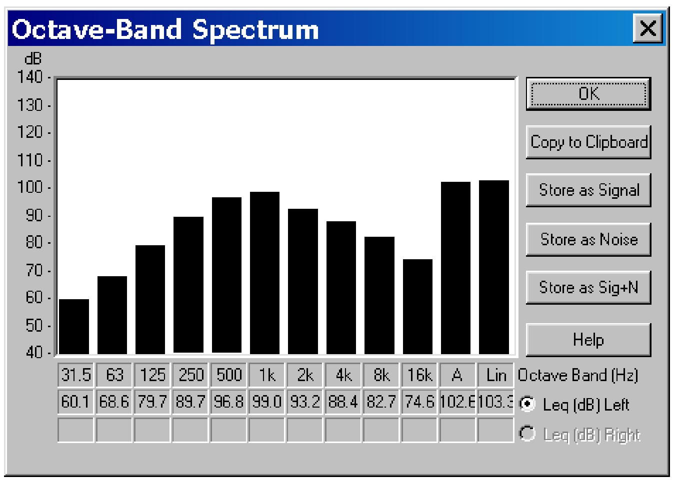


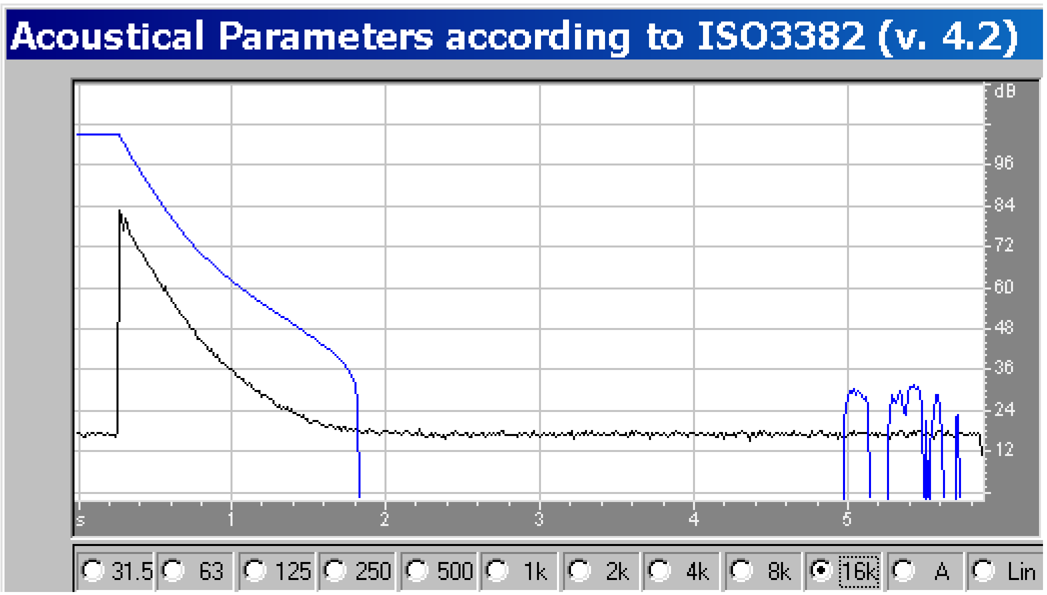
Traditional measurement methods

- **Pulsive sources: ballons, blank pistol**

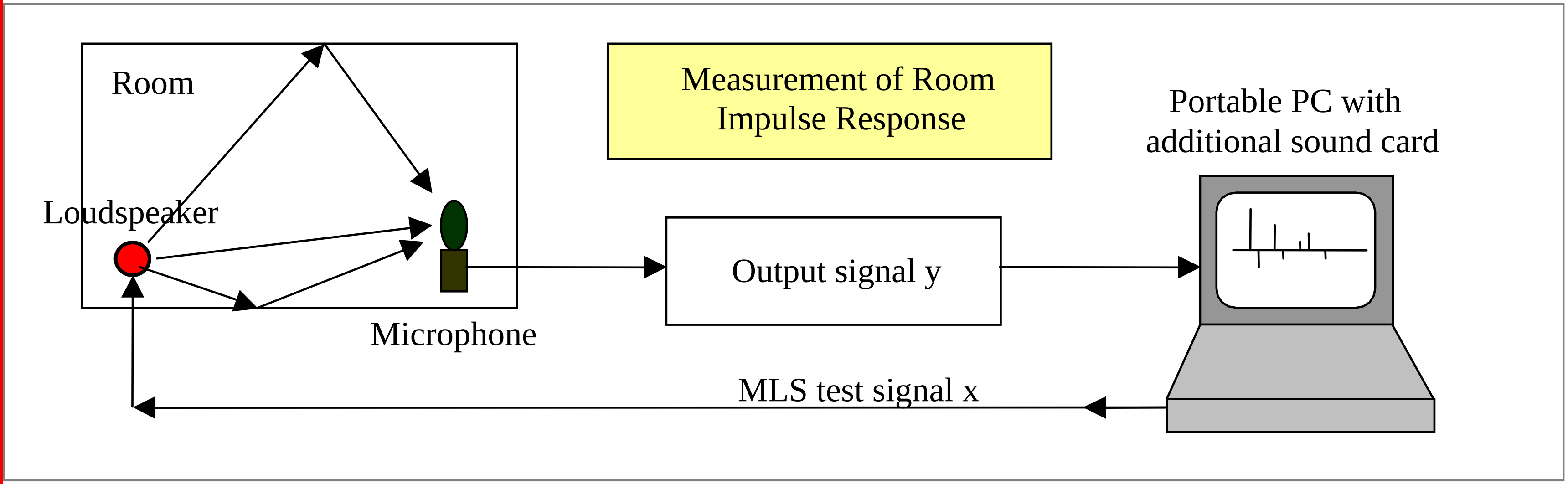
Example of a pulsive impulse response





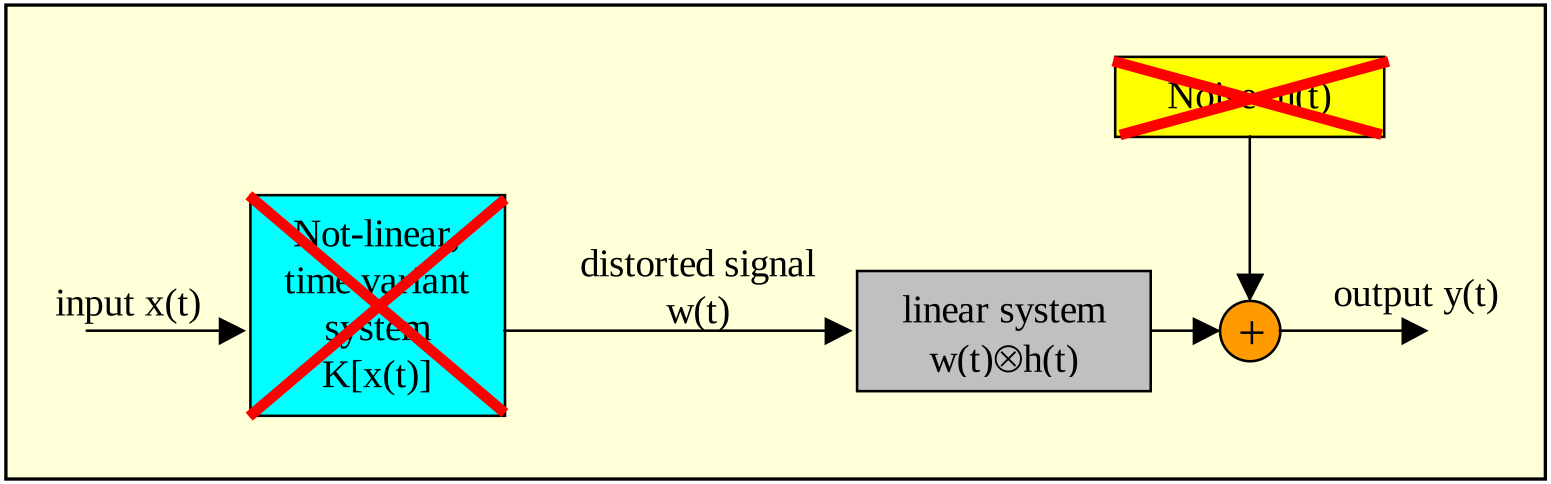


Loudspeaker as sound source



- 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

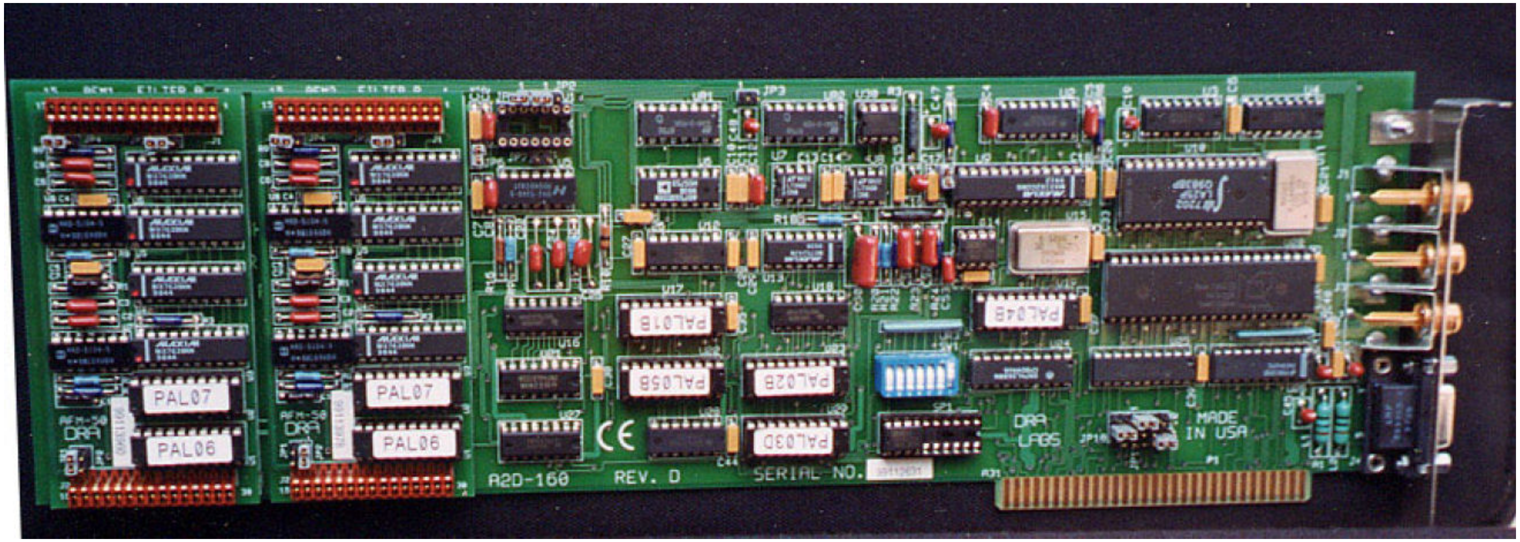


- The desidered result is the linear impulse response of the acoustic propagation h(t). It can be recovered by knowing the test signal x(t) and the measured system output y(t).
- It is necessary to exclude the effect of the not-linear part K and of the background noise n(t).

Electroacoustical methods

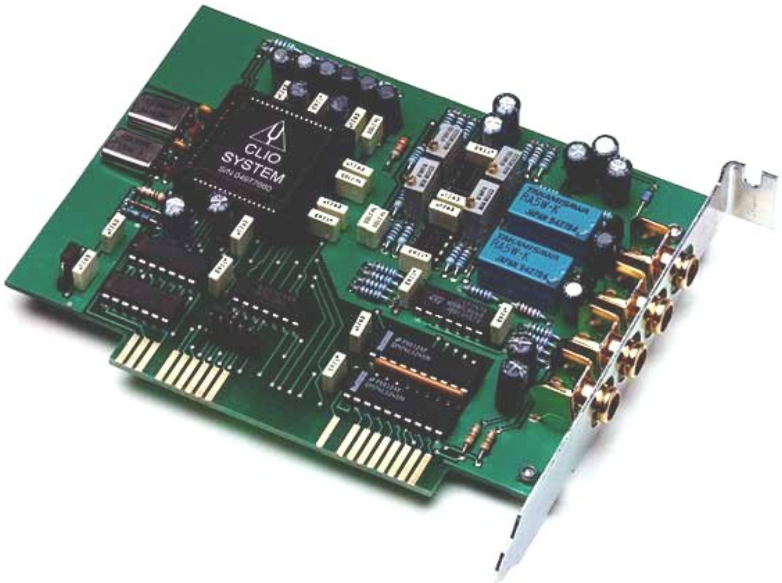
- **Different types of test signals have been developed, providing good immunity to background noise and easy deconvolution of the impulse response:**
  - MLS (Maximum Lenght Sequence, pseudo-random white noise)
  - TDS (Time Delay Spectrometry, which basically is simply a linear sine sweep, also known 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

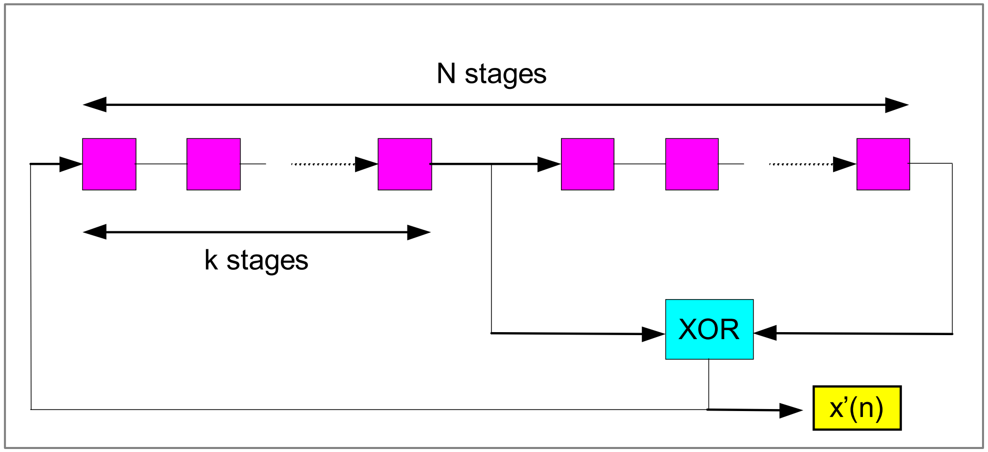
SC-01

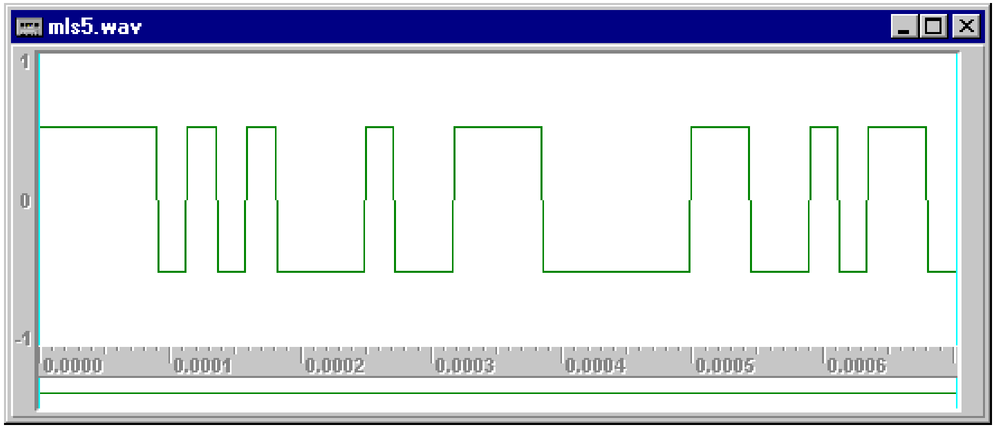
- **The Italian-made CLIO system has superseded MLSSA for most low-cost electroacoustics applications (measurement of loudspeakers, quality control)**

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**

Theory of 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

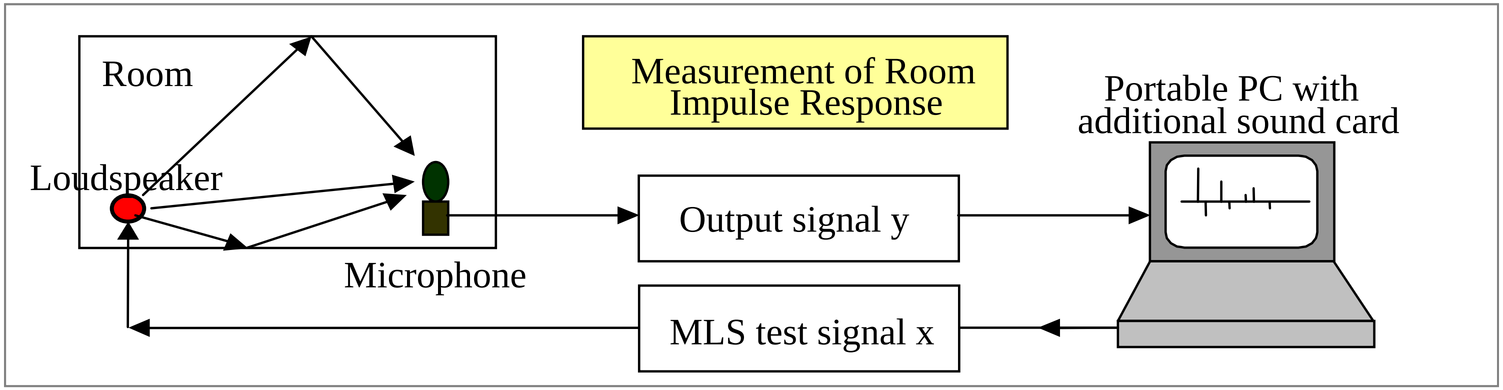
- **The re-recorded signal y(i) is cross-correlated with the excitation signal thanks to a fast Hadamard transform. The result is the required impulse response h(i), if the system was linear and time-invariant**

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

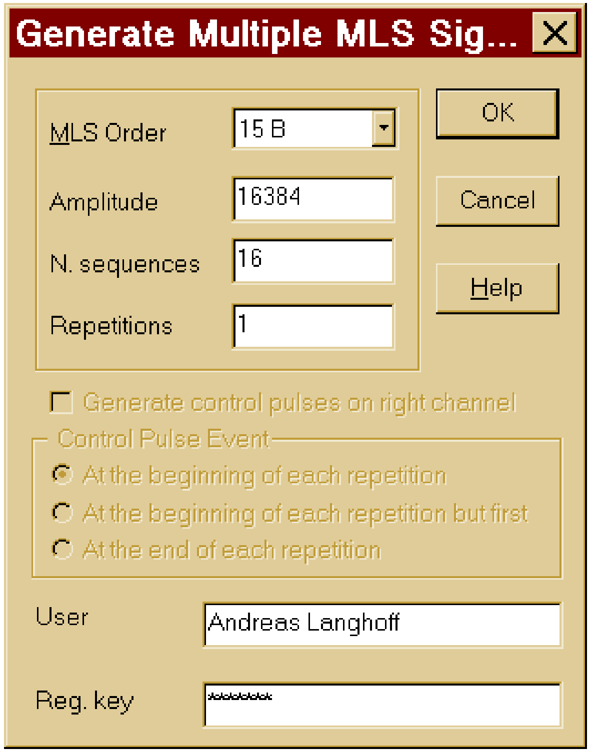
- **Where M is the Hadamard matrix, obtained by permutation of the original MLS sequence m(i)**

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

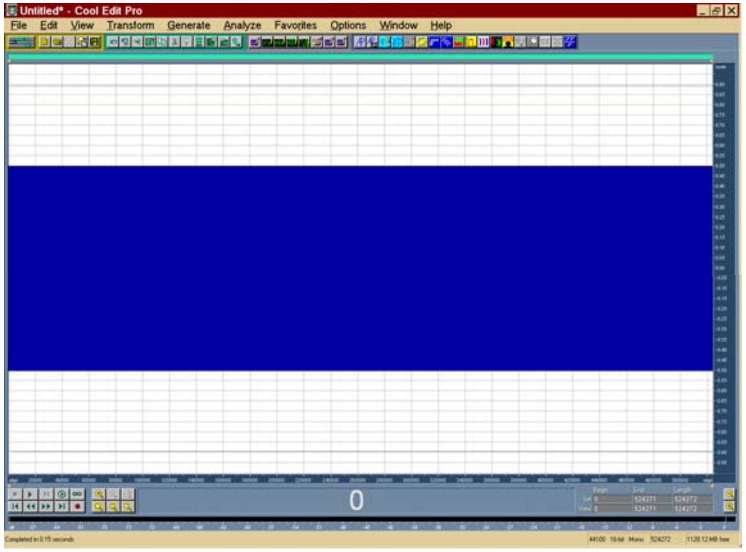
MLS example



Room
Loudspeaker
Microphone
Measurement of Room Impulse Response
Portable PC with additional sound card
Output signal y
MLS test signal x

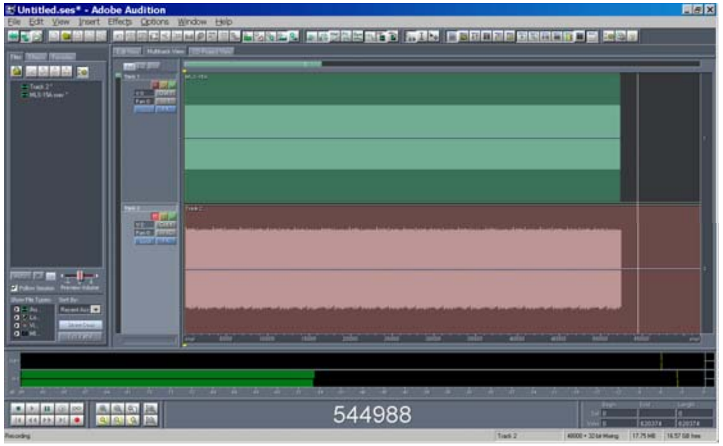


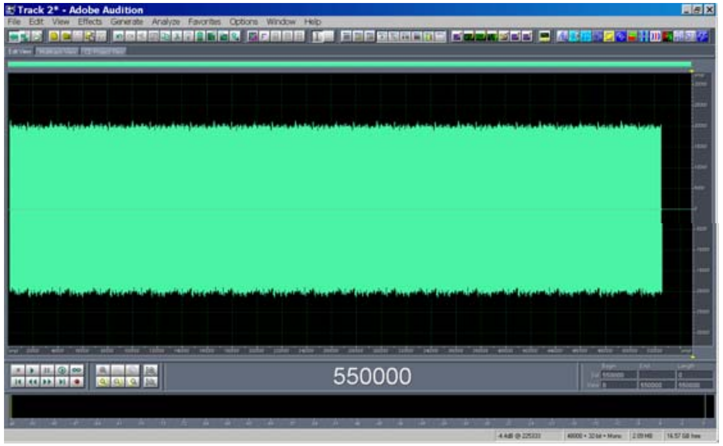
Generate Multiple MLS Sig...
MLS Order
15 B
OK
Amplitude
16384
Cancel
N. sequences
16
Help
Repetitions
1
Generate control pulses on right channel
Control Pulse Event
At the beginning of each repetition
At the beginning of each repetition but first
At the end of each repetition
User
Andreas Langhoff
Reg. key



Untitled* - Cool Edit Pro
File Edit View Transform Generate Analyze Favorites Options Window Help
0

MLS example


Untitled.ses* - Adobe Audition
544988


Track 2* - Adobe Audition
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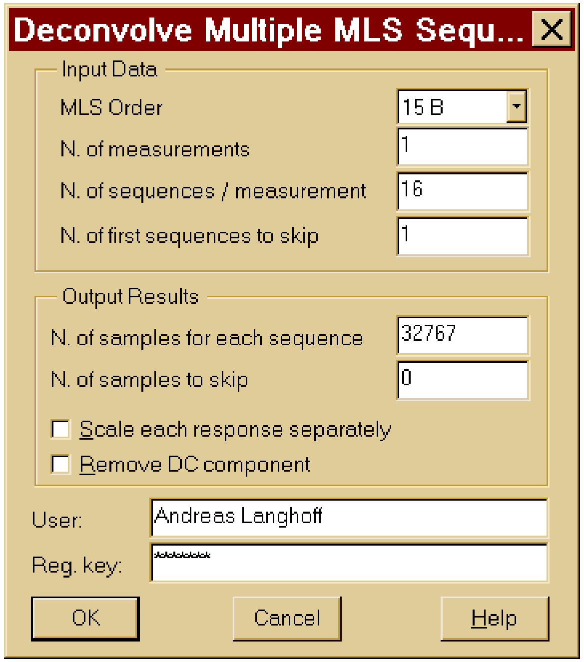
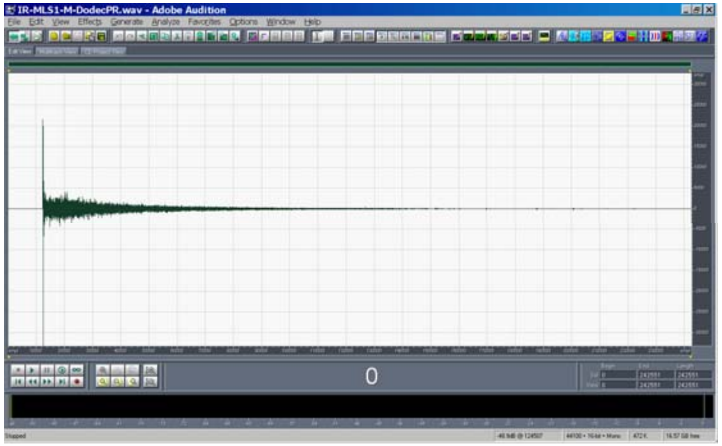

Deconvolve Multiple MLS Sequ...
Input Data
MLS Order
15 B
N. of measurements
1
N. of sequences / measurement
16
N. of first sequences to skip
1
Output Results
N. of samples for each sequence
32767
N. of samples to skip
0
Scale each response separately
Remove DC component
User:
Andreas Langhoff
Reg. key:
OK
Cancel
Help


IR-MLS1-M-DodecPR.wav - Adobe Audition
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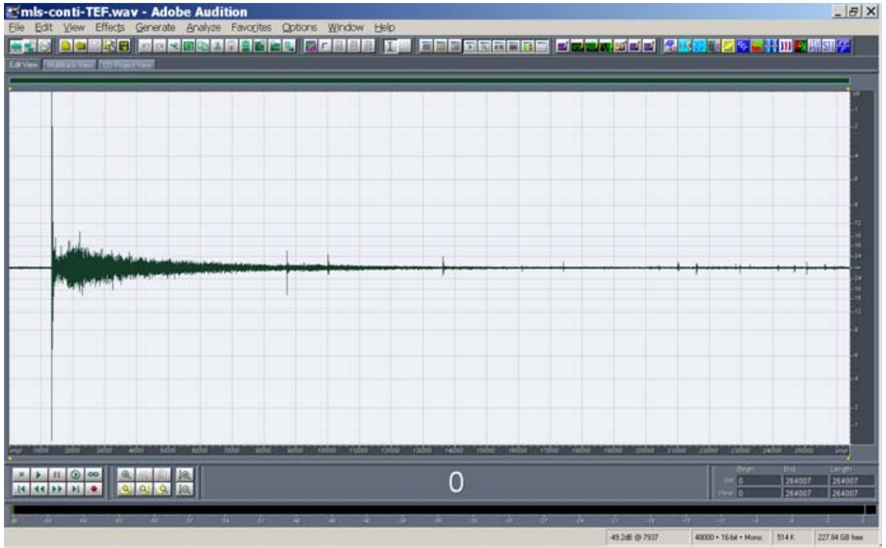
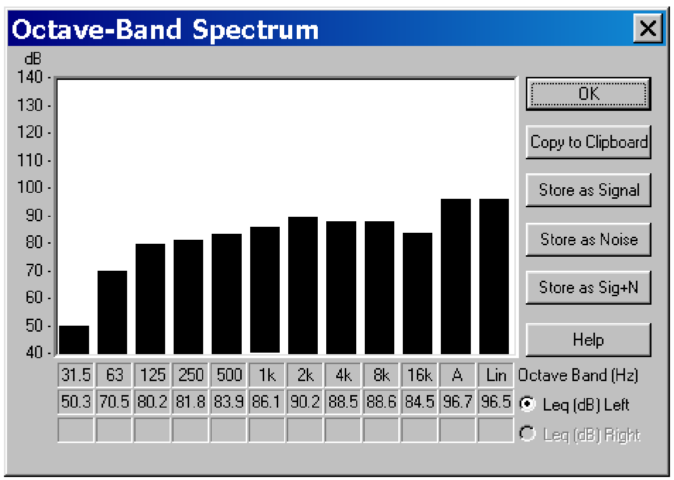
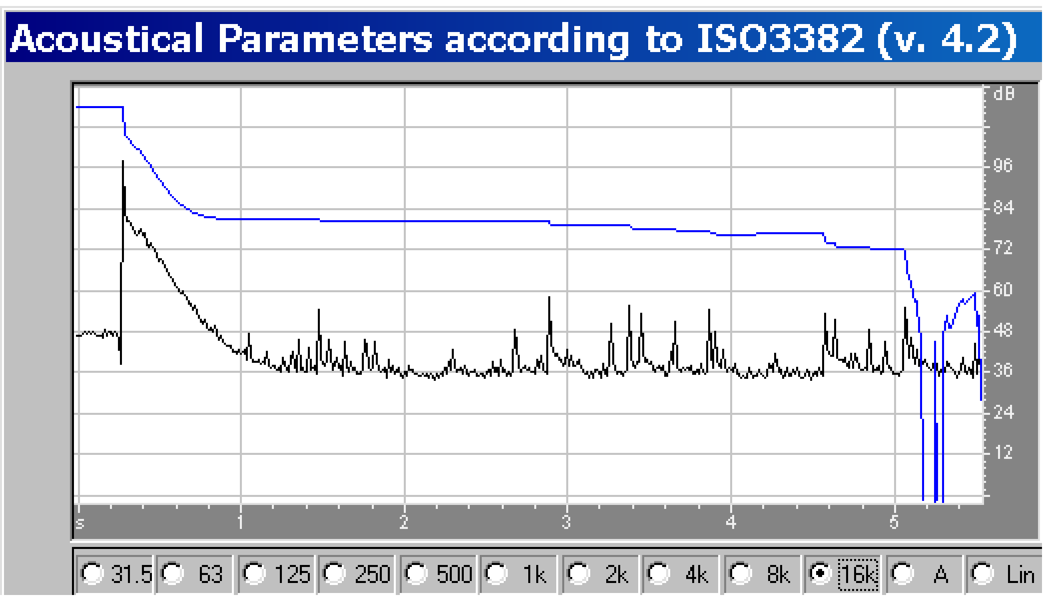
Example of a MLS impulse response

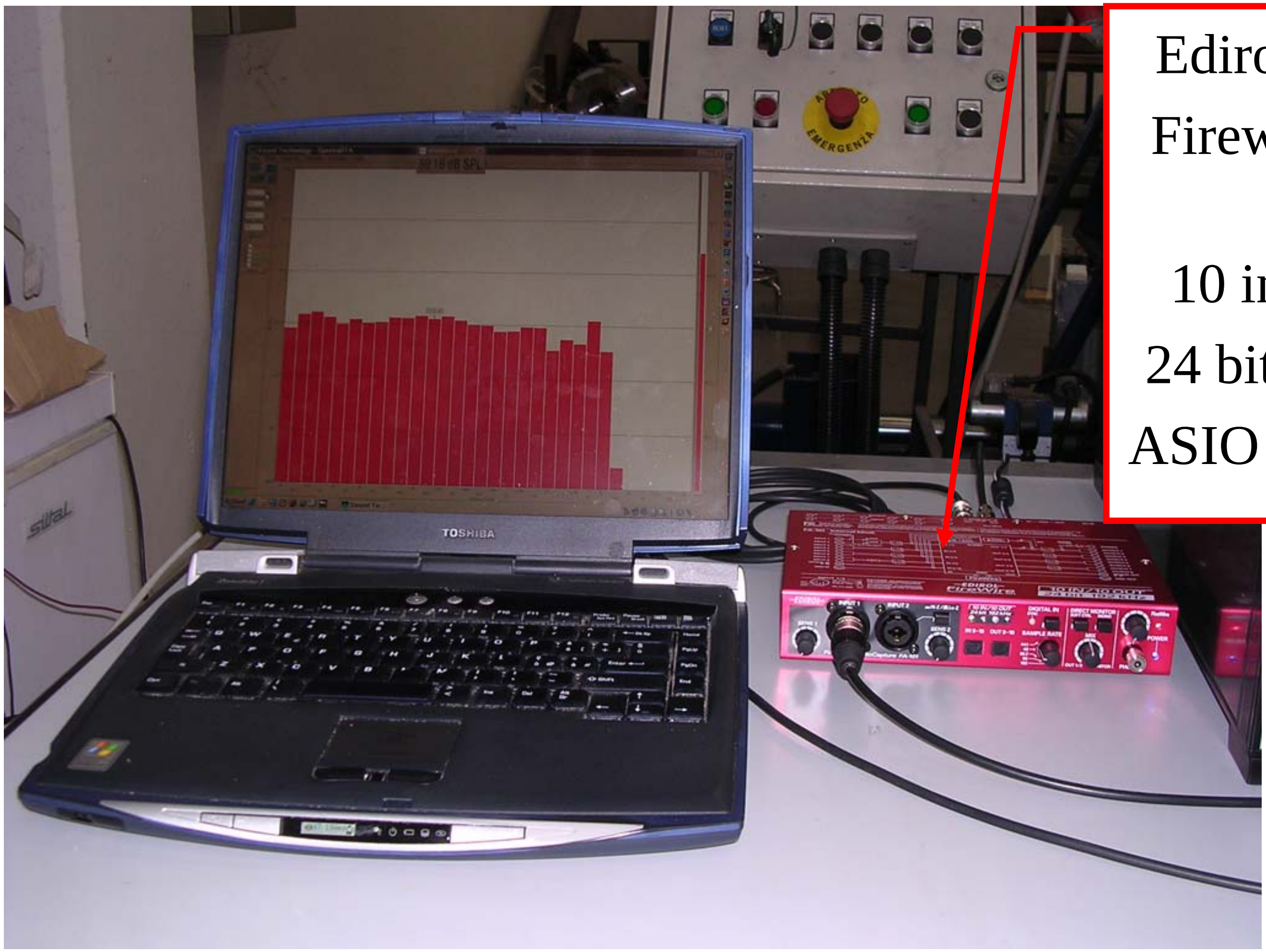






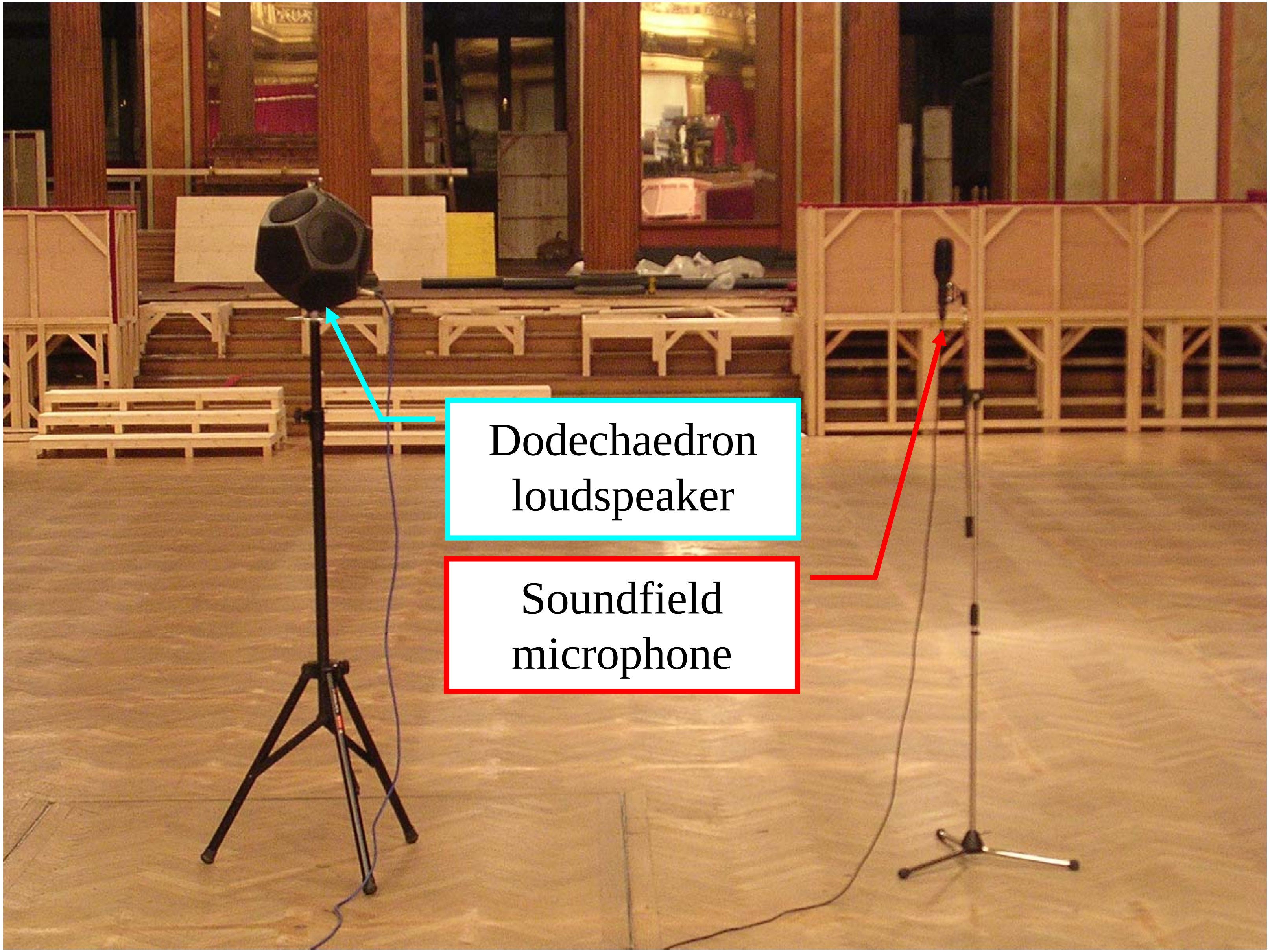
The Present

Today's Hardware: PC and audio interface

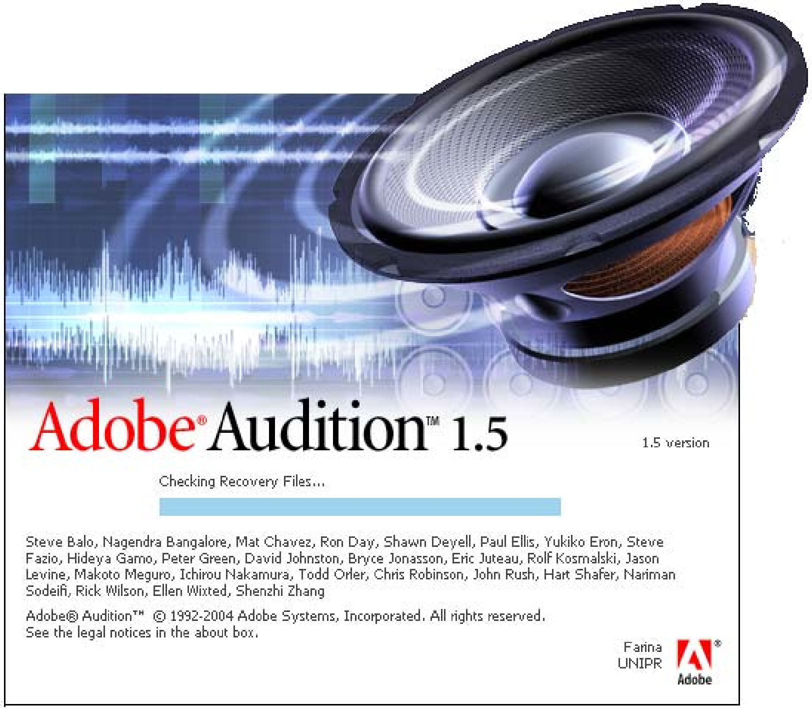


Edirol FA-101
Firewire sound card:
10 in / 10 out
24 bit, 192 kHz
ASIO and WDM

Hardware: loudspeaker & microphone



The first ESS system - AURORA



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Aurora Plugins

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

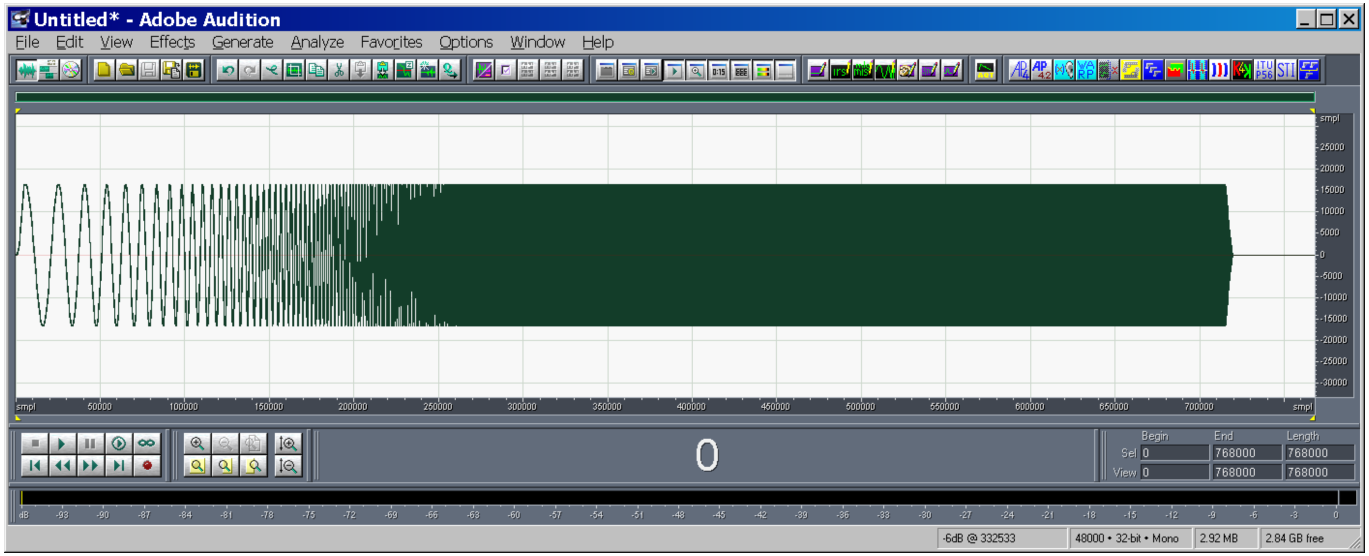
- **Aurora was the first measurement system based on standard sound cards and employing the Exponential Sine Sweep method**
- **It also works with traditional TDS and MLS methods, so the comparison can be made employing exactly the same hardware**

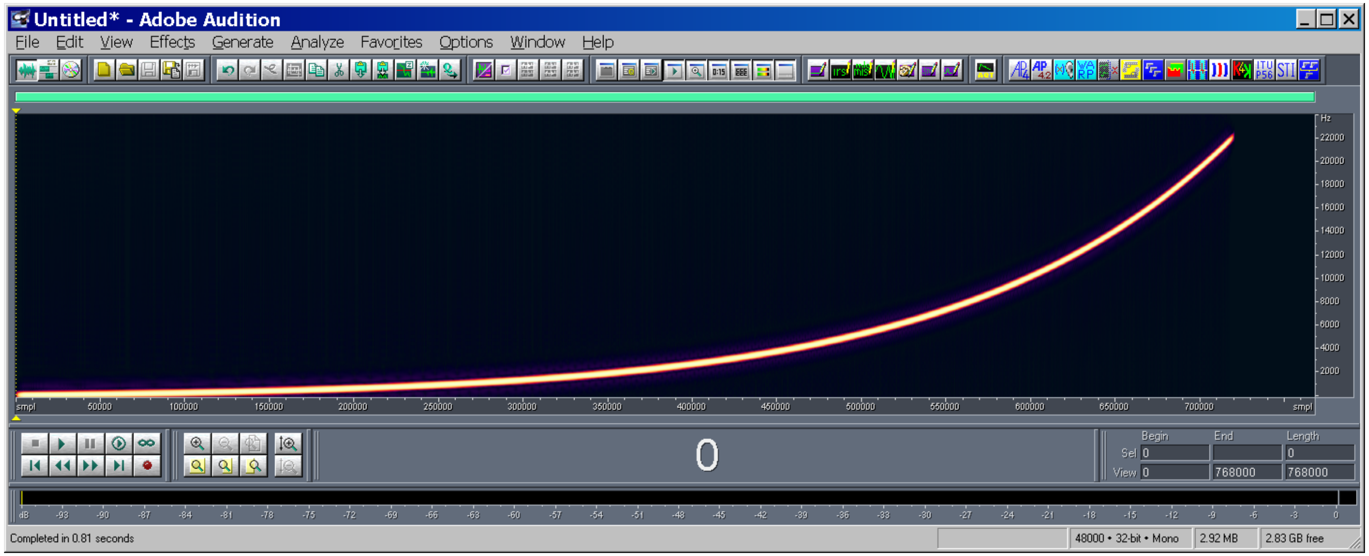
Exponential Sine Sweep method

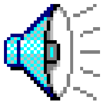
- x(t) is a band-limited sinusoidal sweep signal, which frequency is varied exponentially with time, starting at f_1 and ending at f_2.

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

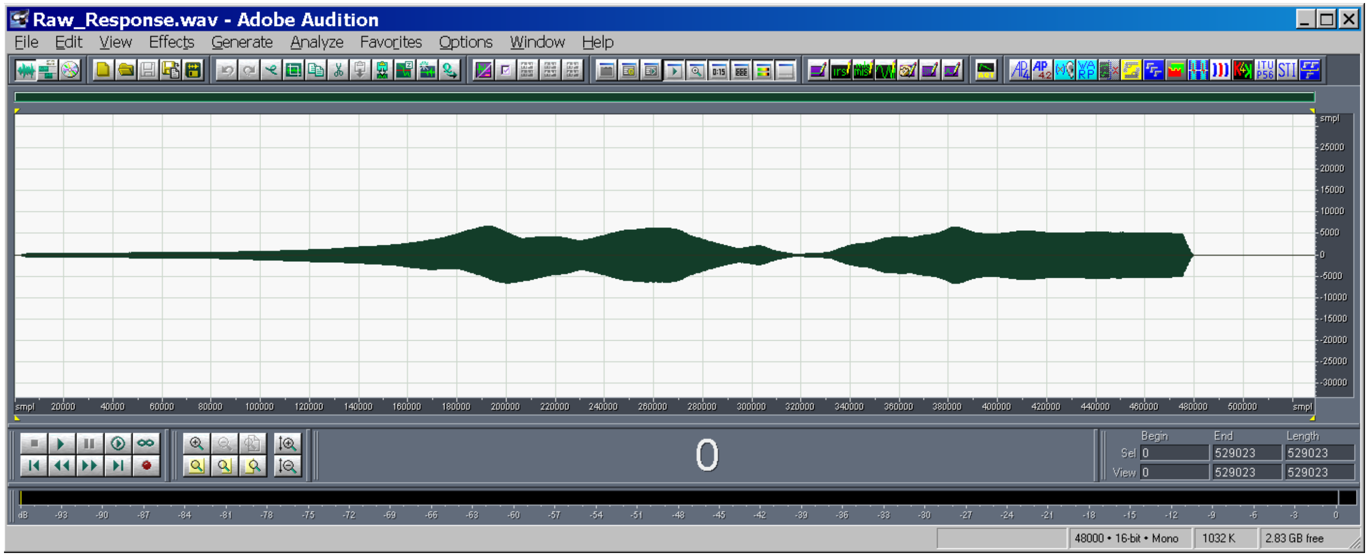
Test Signal – x(t)

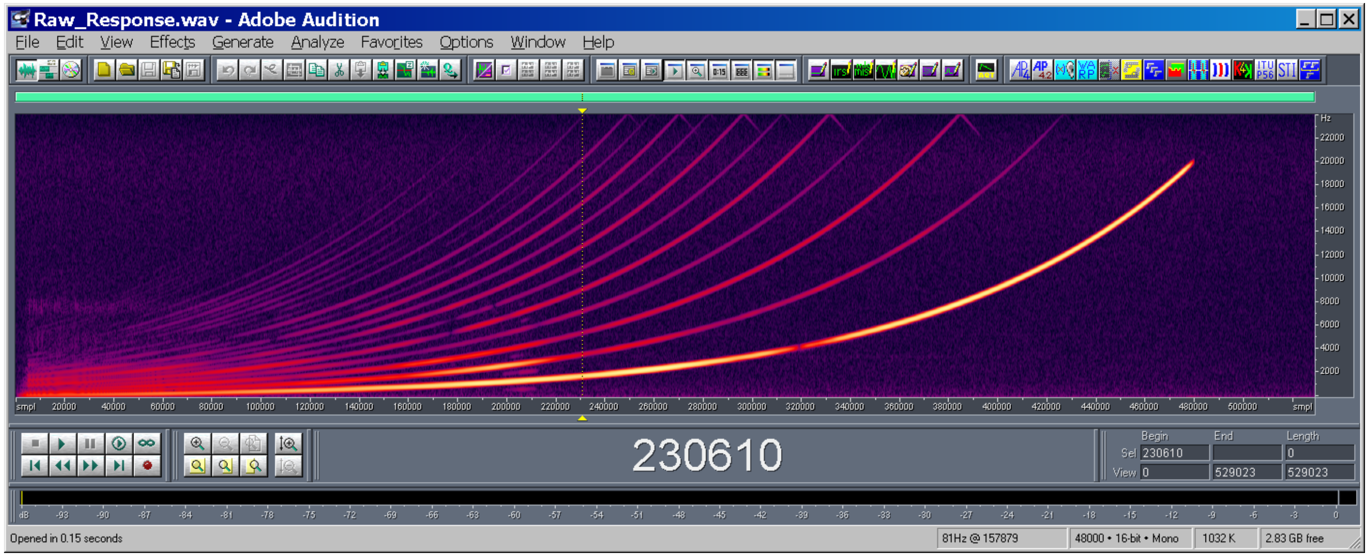





Stop

Measured signal - y(t)

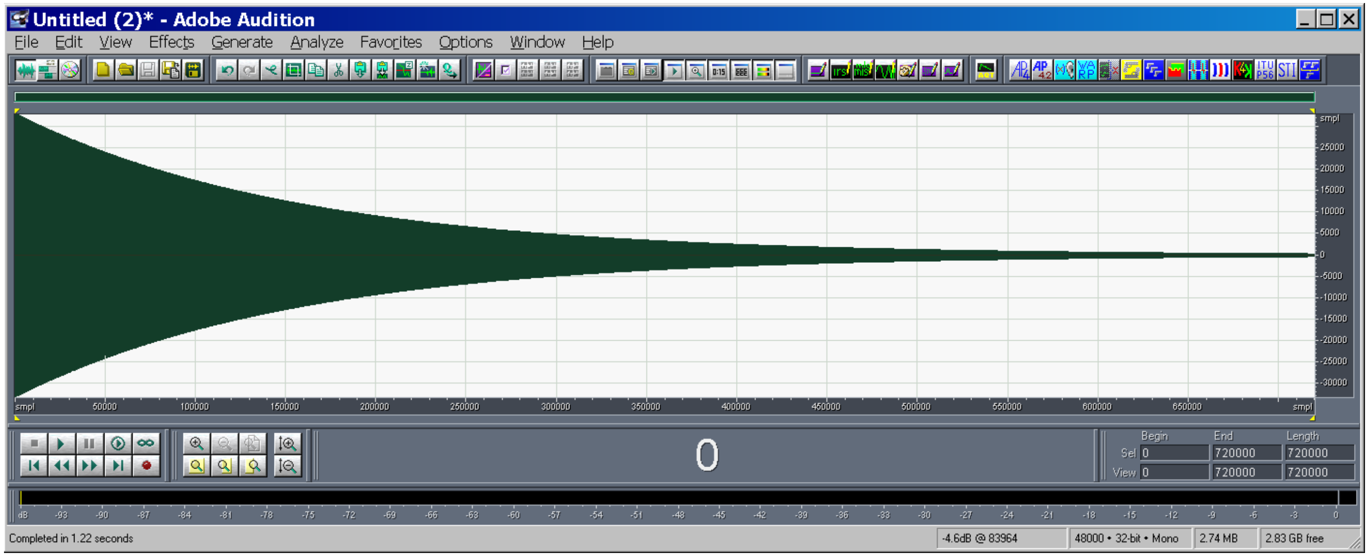


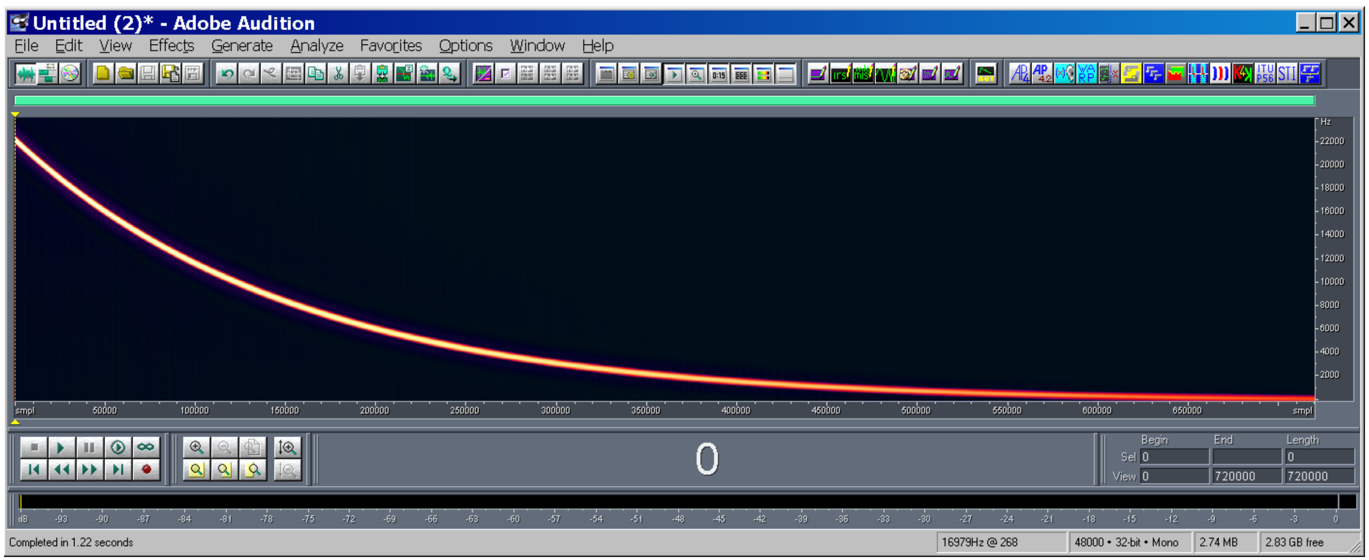


Stop

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

Inverse Filter – z(t)



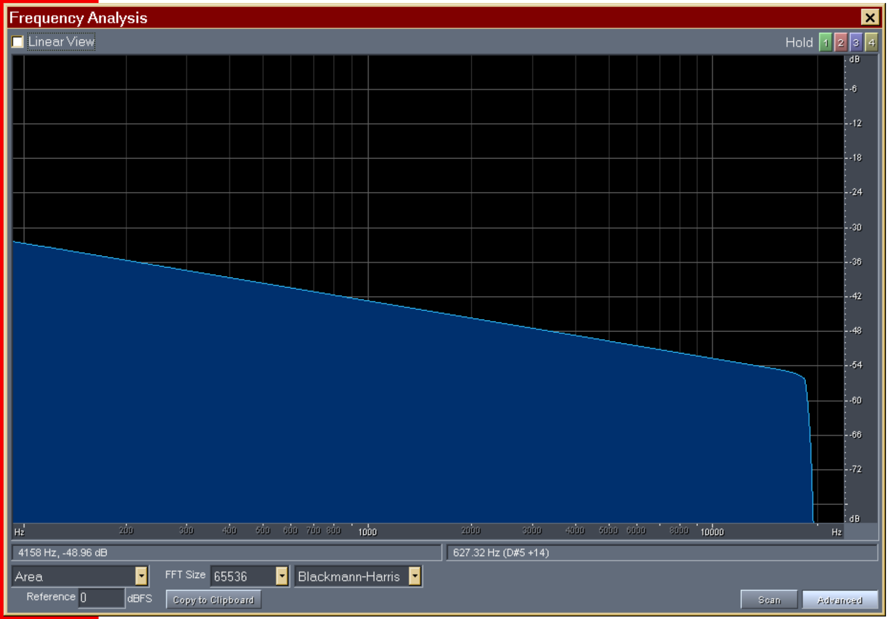


Stop

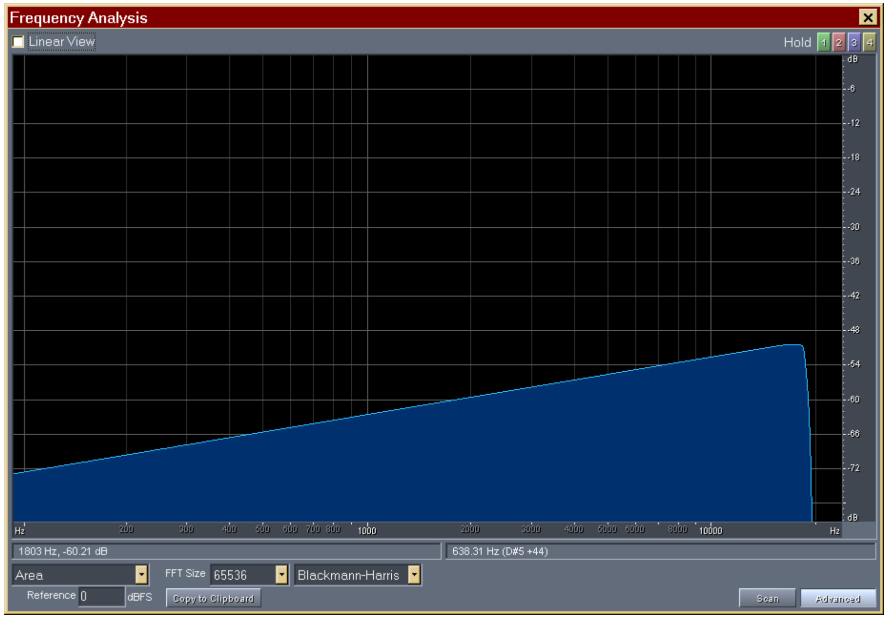
The deconvolution of the IR is obtained convolving the measured signal y(t) with the inverse filter z(t) [equalized, time-reversed x(t)]

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

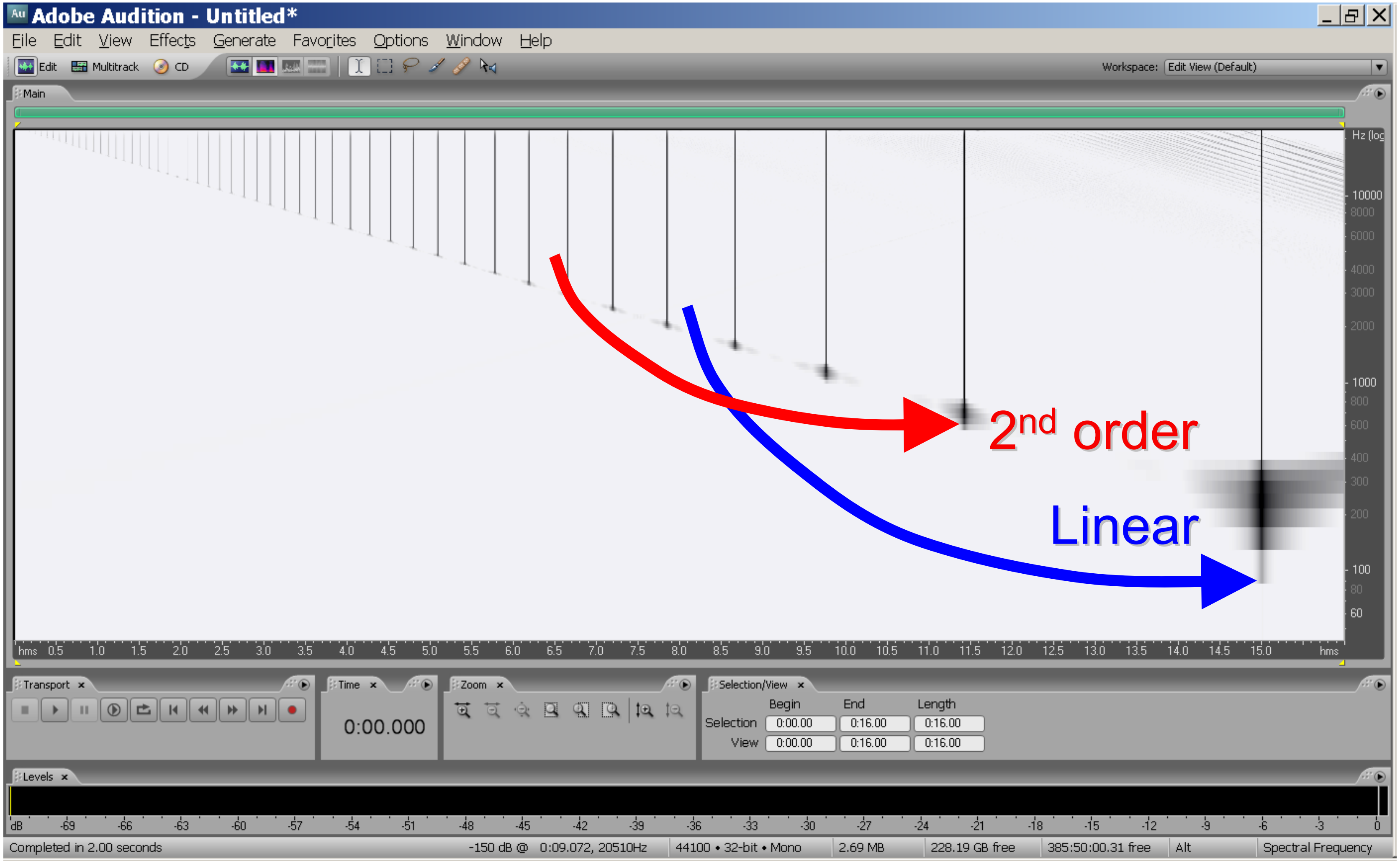


Test Signal x(t)



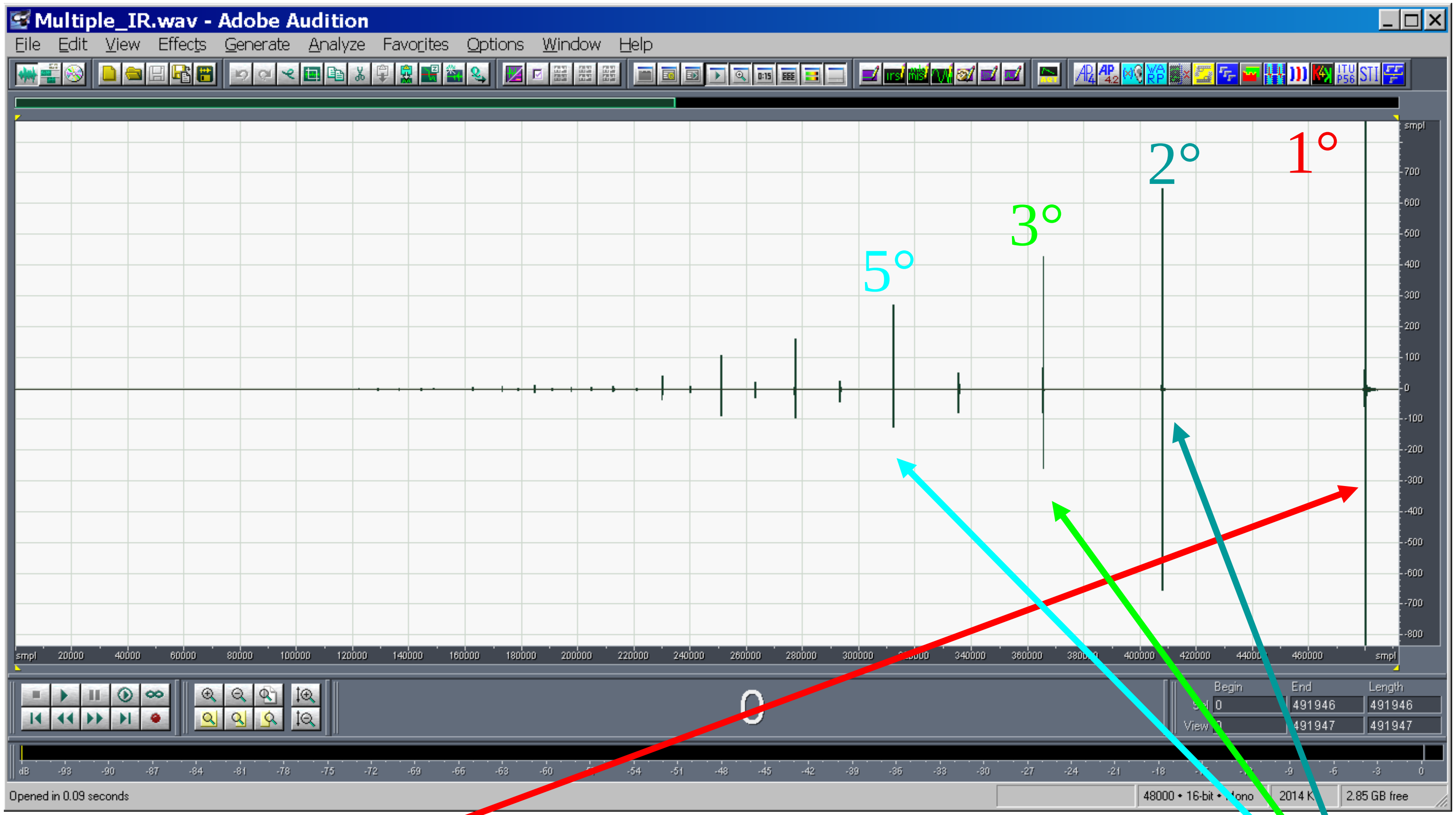
Inverse Filter z(t)

Deconvolution = rotation of the sonograph



- **Convolving with the inverse filter rotates the time-log(f) plane counter clockwise**

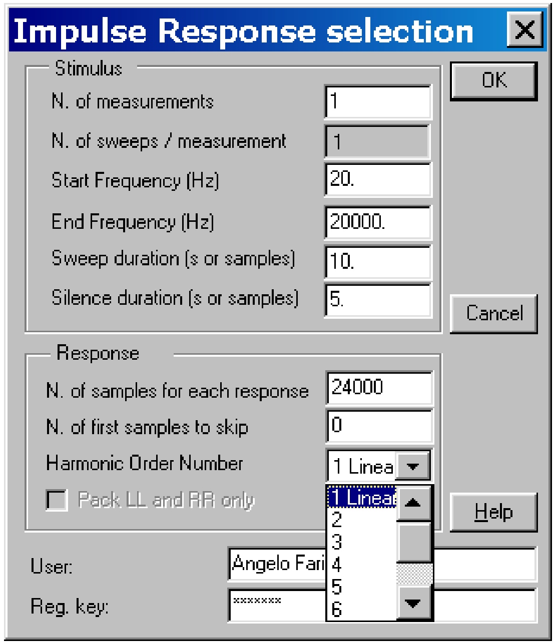
Result of the deconvolution

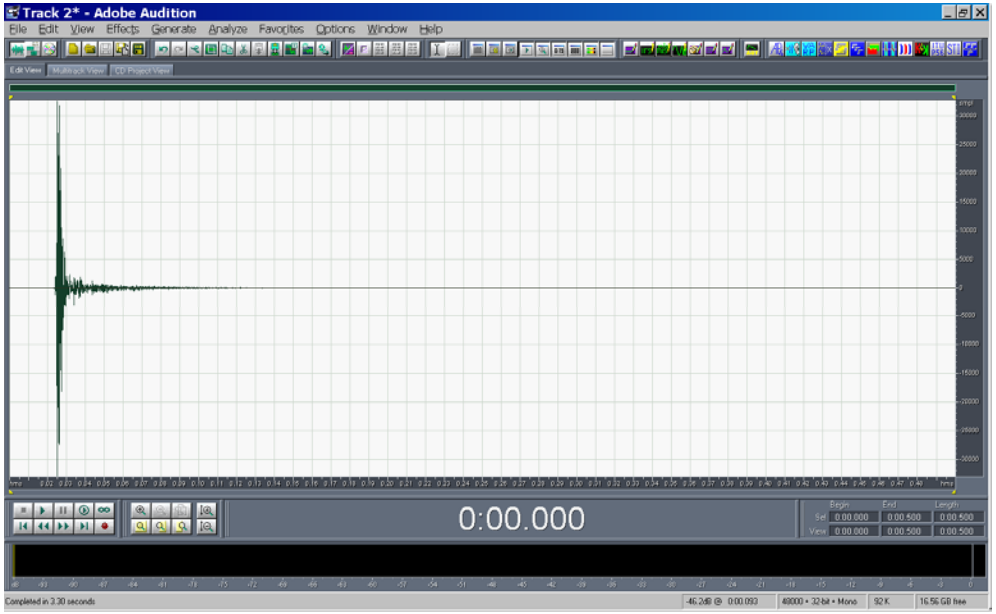


The last impulse response is the linear one, the preceding are the harmonics distortion products of various orders

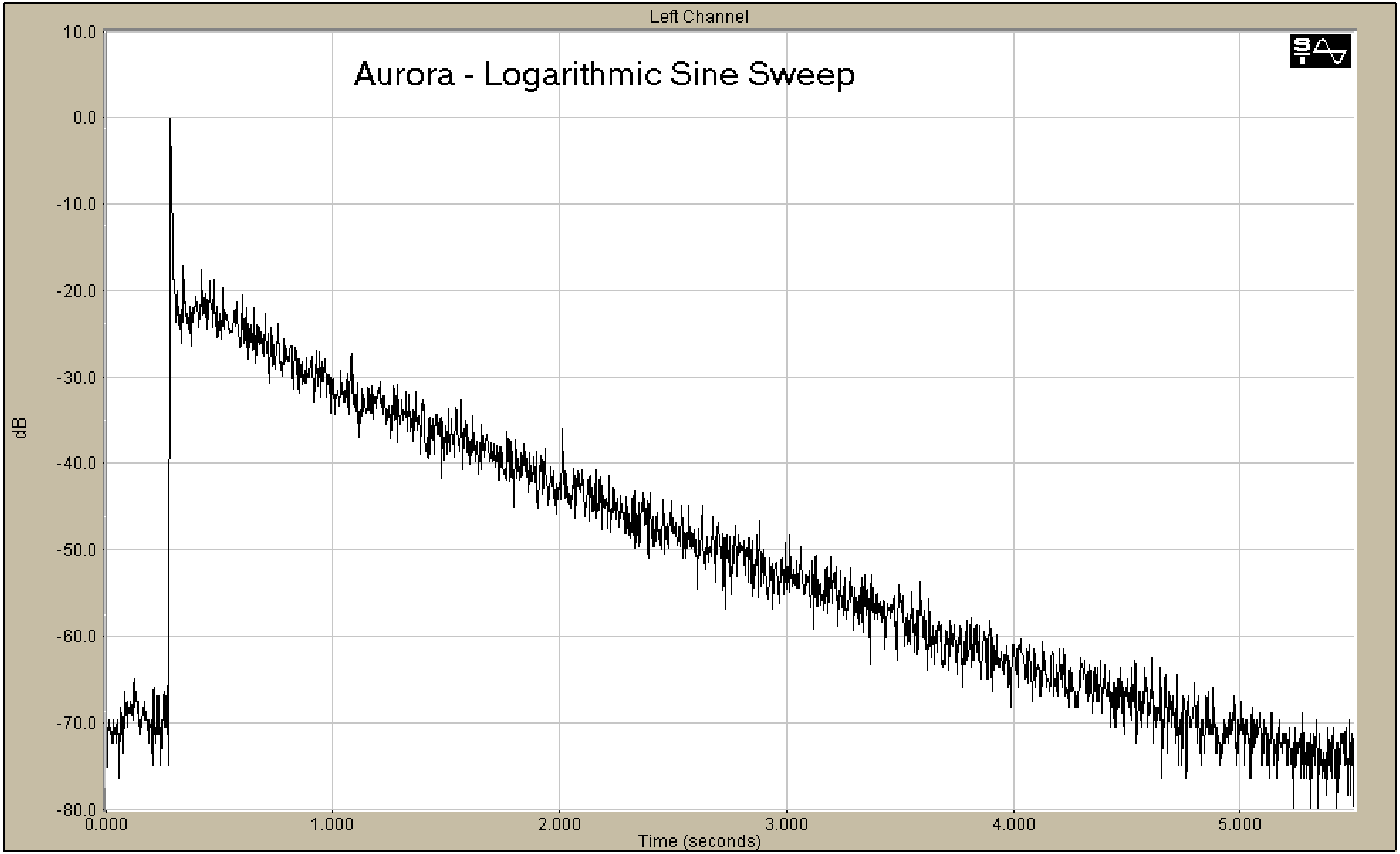
IR Selection

- **After the sequence of impulse responses has been obtained, it is possible to select and insulate just one of them:**

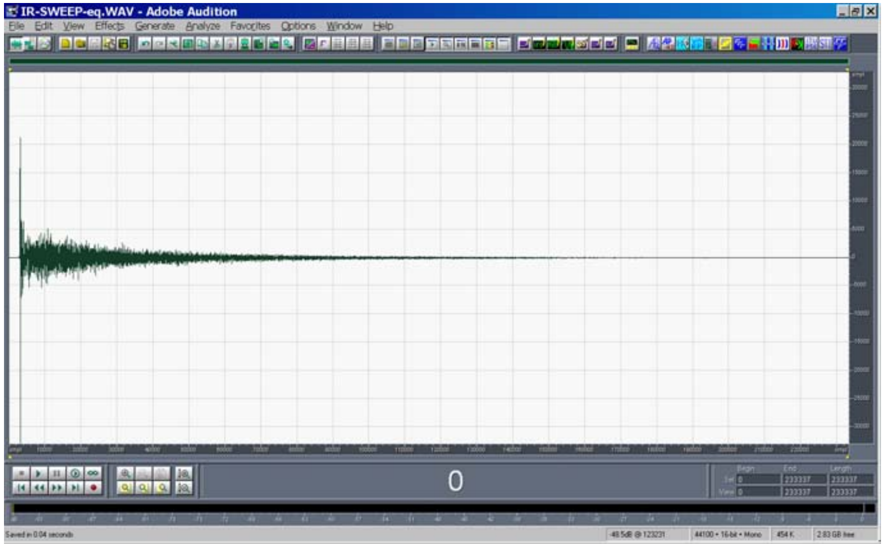


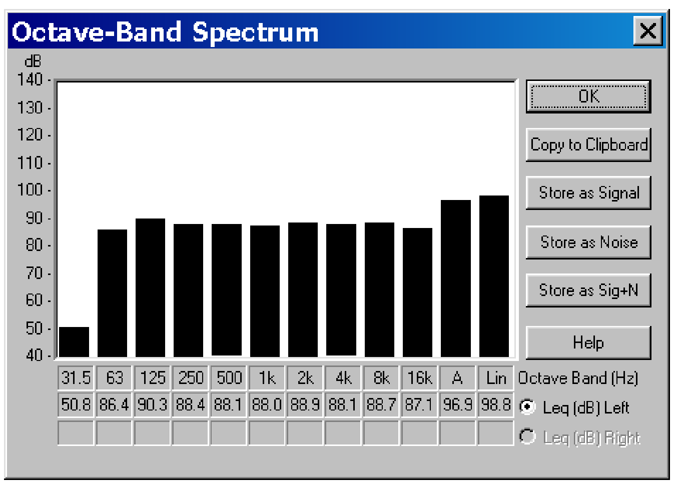


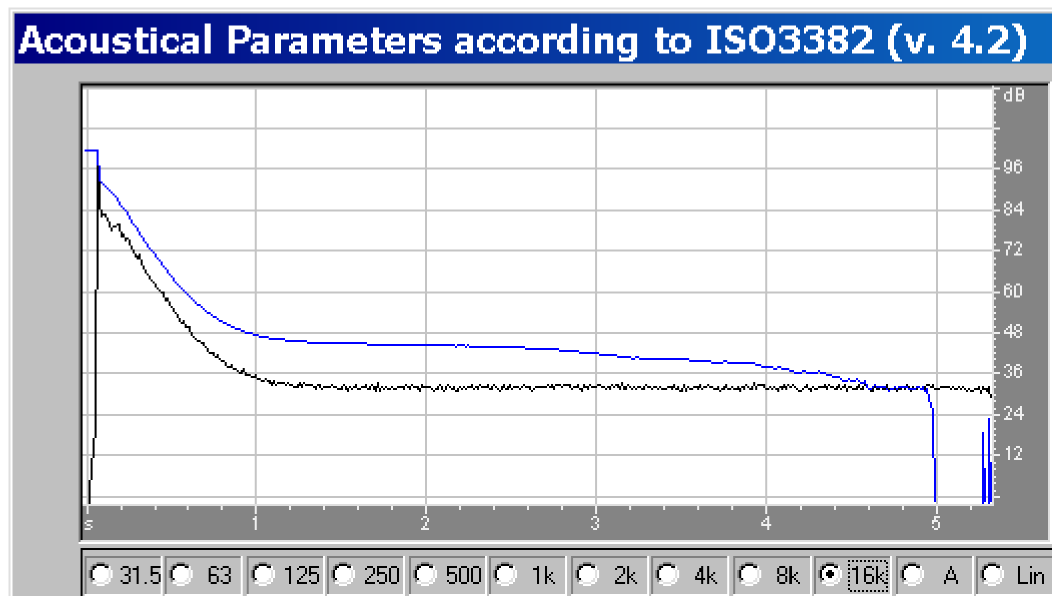
Maximum Length Sequence vs. Exp. Sine Sweep



Example of an ESS impulse response

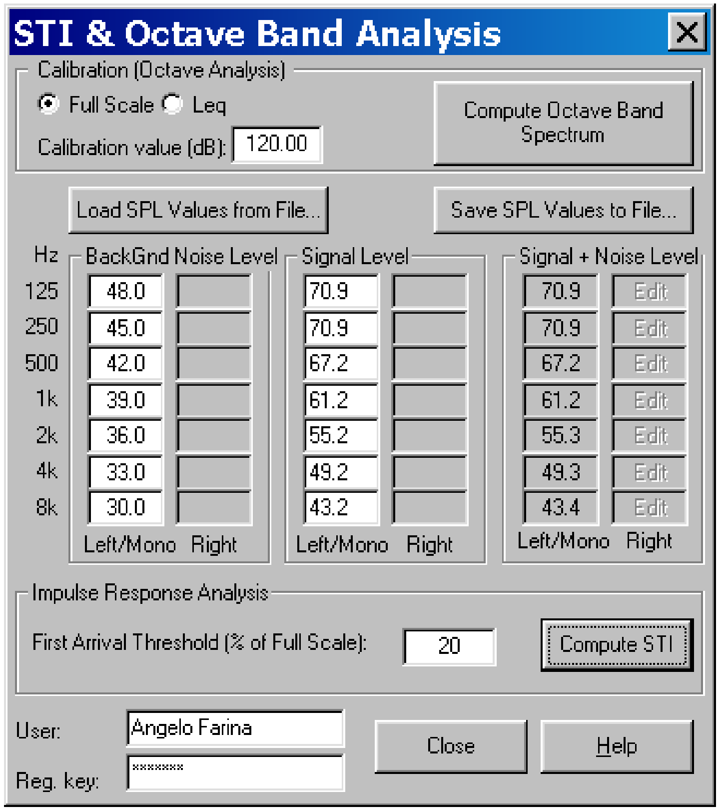


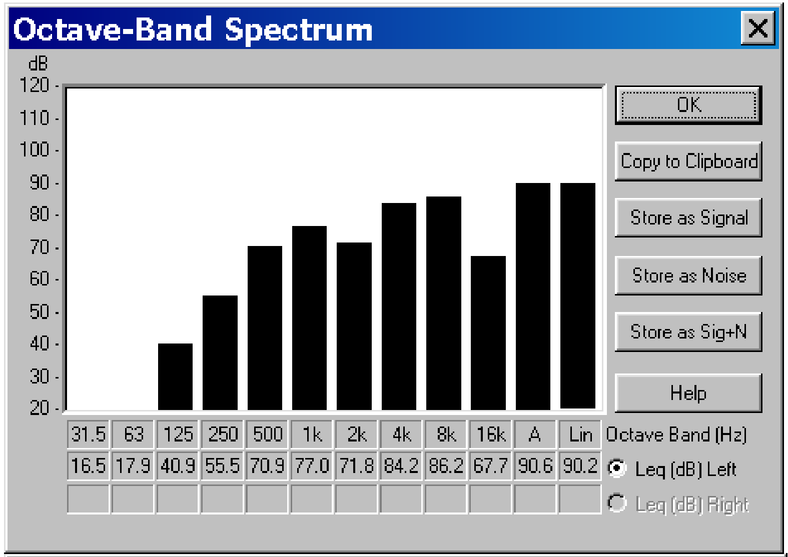


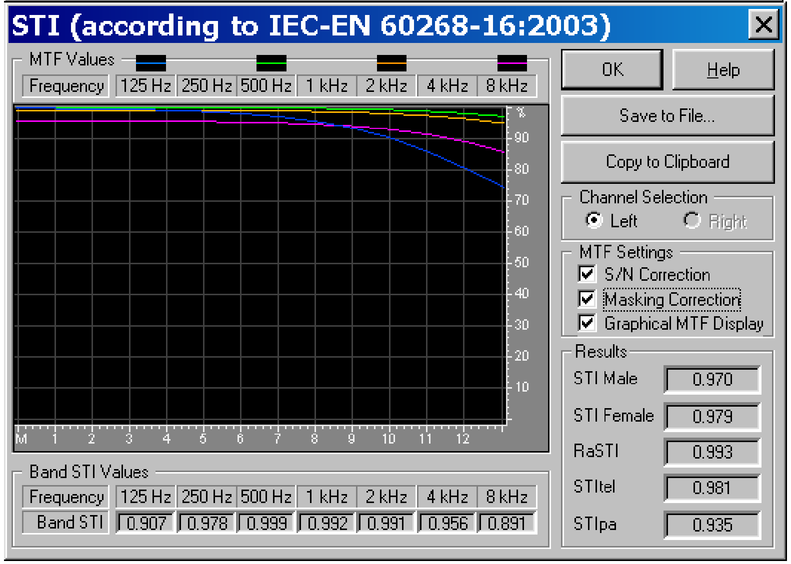


Post processing of impulse responses

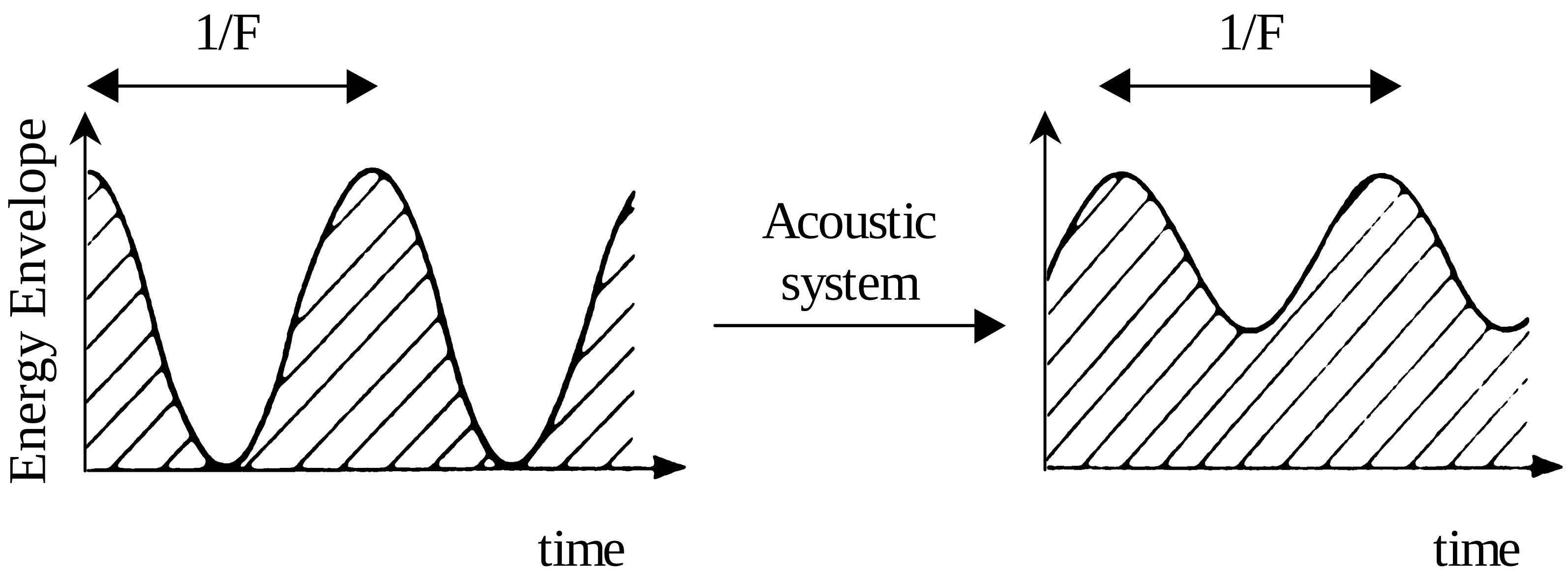
- **A special plugin has been developed for the computation of STI according to IEC-EN 60268-16:2003**







The STI Method



The STI method is based on the MTF concept: a carrier signal (one-octave-band-filtered noise) is amplitude modulated at a given modulation frequency with 100% modulation depth. At the receiver, the modulation depth is reduced, due to noise, reverb, echoes, etc.

MTF from Impulse Response

- **It is possible to derive the MTF values from a single impulse response measurement:**

To compute each value of *m(F)* from the impulse response *h(t)*, an octave-band filter is first applied to the impulse response, in order to select the carrier's frequency band *f*. Then *m(F)* is obtained with the formula

$$m(F) = \frac{\int_0^\infty h_f^{\,2}(\tau) \cdot \exp(-j \cdot 2 \cdot \pi \cdot F \cdot \tau) \cdot d\tau}{\int_0^\infty h_f^{\,2}(\tau) \cdot d\tau}$$

Background noise

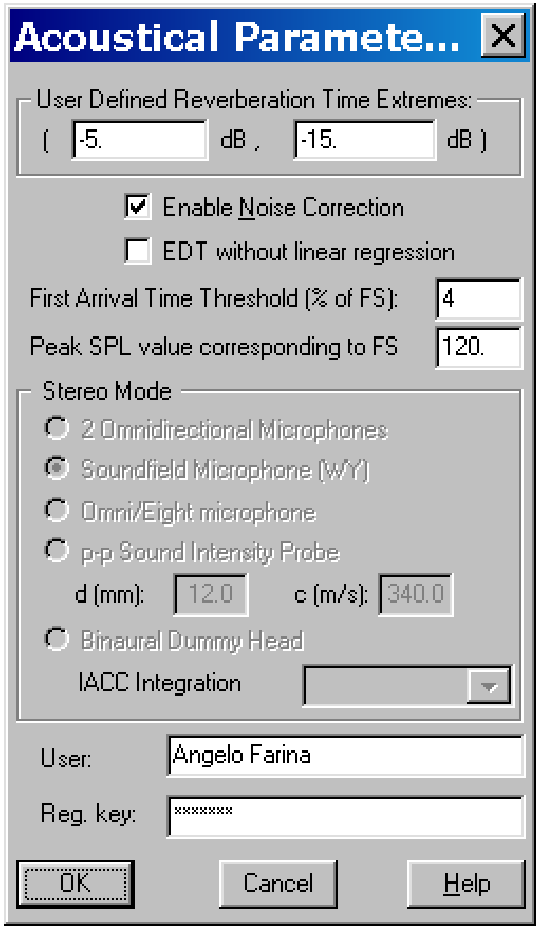
- If the background noise is superposed to the impulse response, the previous method already takes care of it, and the MTF values are measured correctly
- However, in some cases, it is advisable to perform a noise-free measurement of the IR, and then insert the effect of the noise with the following expression:

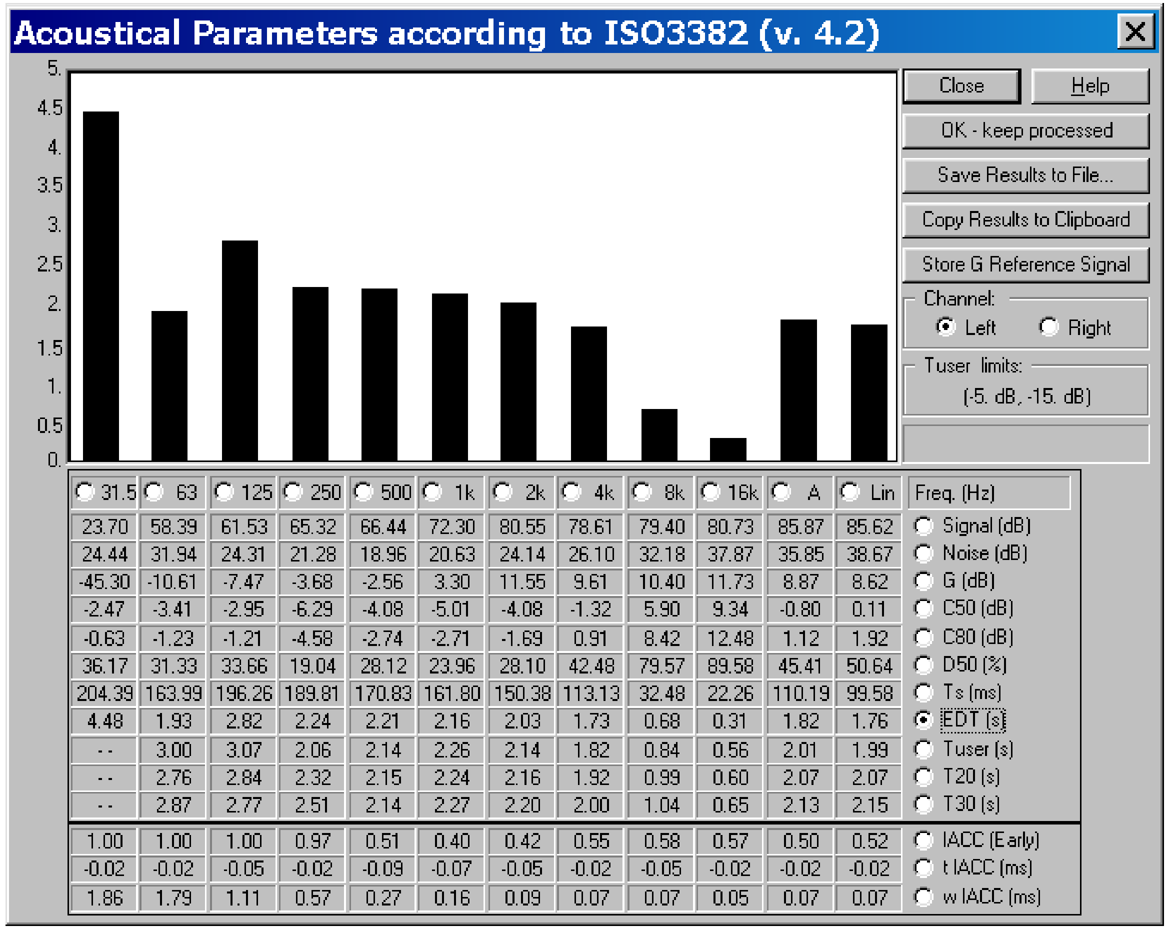
$$m(F) = m'(F) \cdot \frac{1}{1+10^{\left(\frac{L_{noise}-L_{signal}}{10}\right)}}$$

- This makes it possible to measure a “clean” impulse response, and then to perform separately the signl an noise recordigs

Post processing of impulse responses

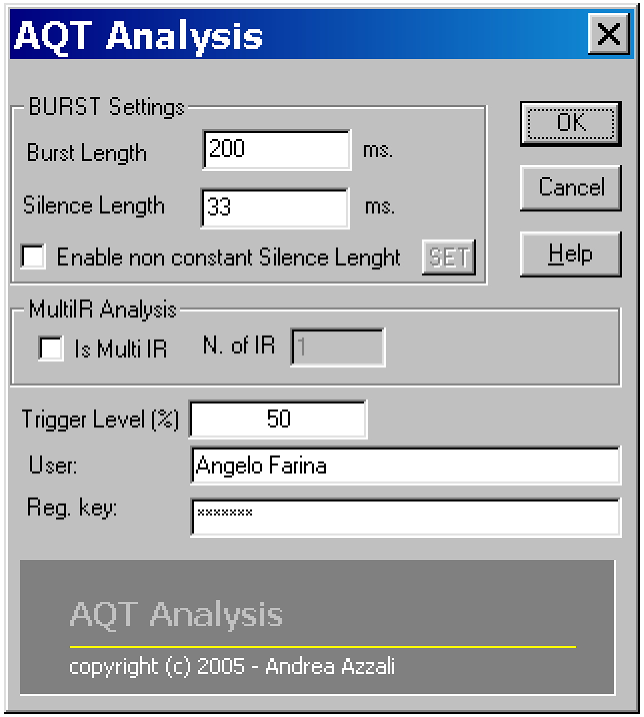
- **A special plugin has been developed for performing analysis of acoustical parameters according to ISO-3382**

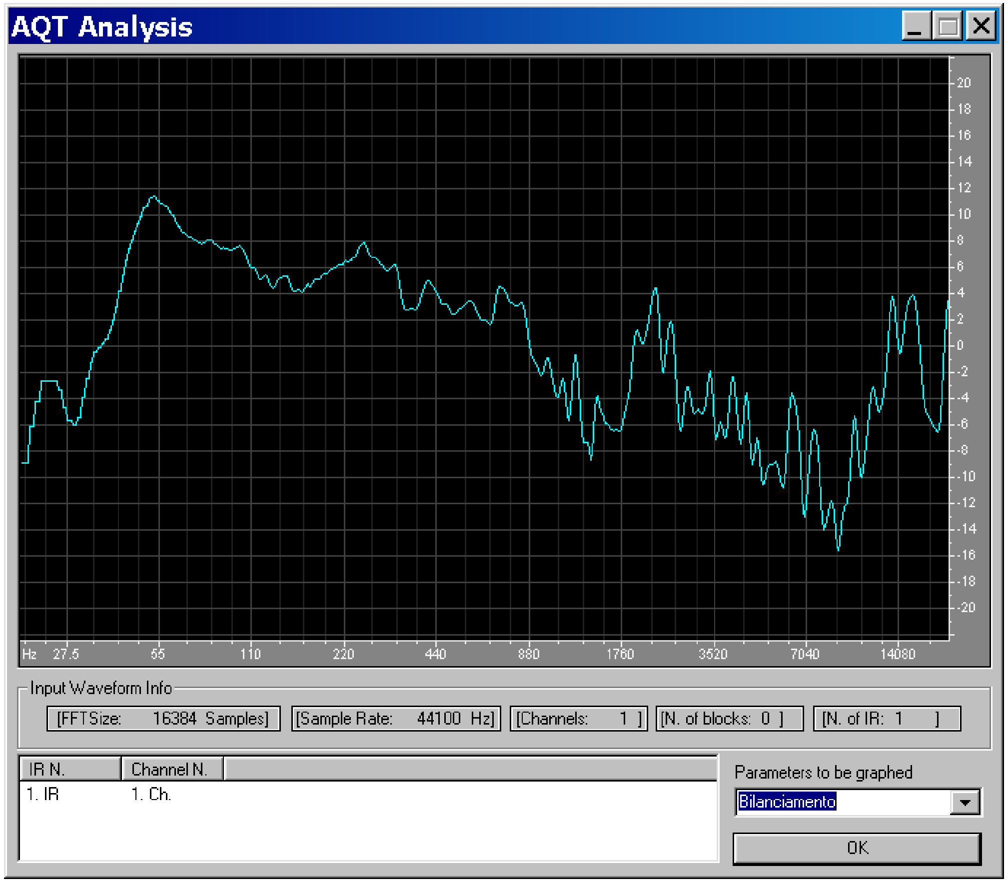




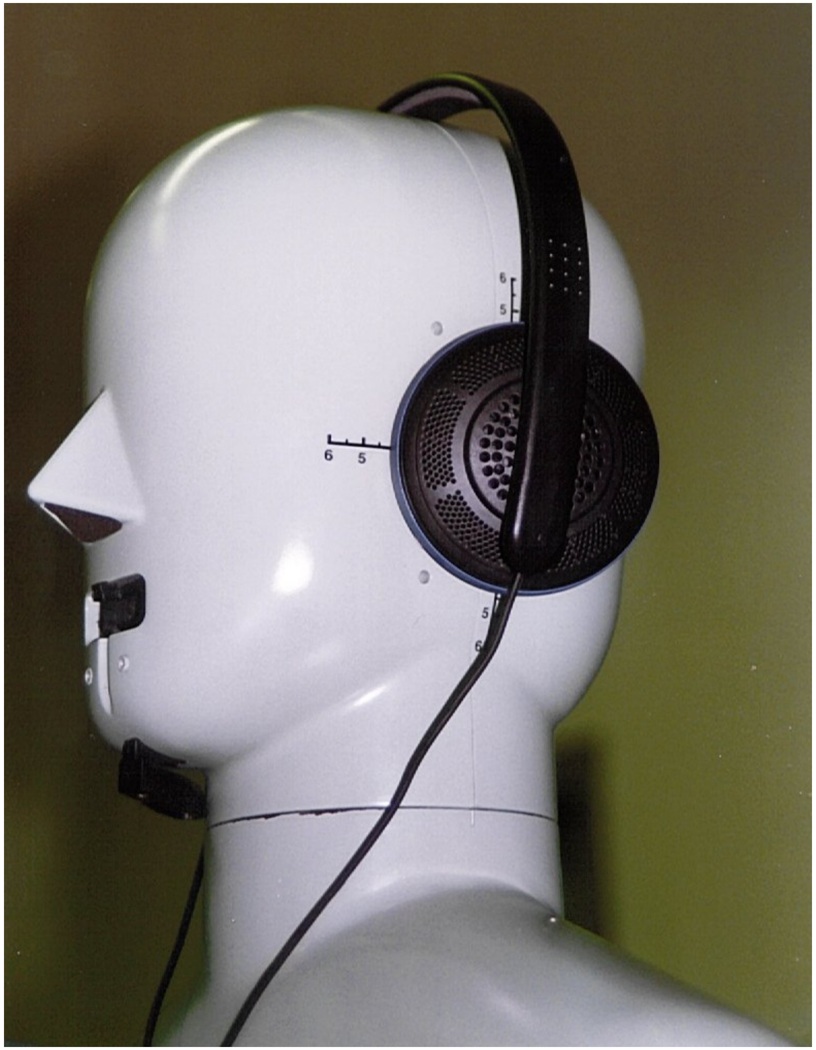
The new AQT plugin for Audition

- **The new module is still under development and will allow for very fast computation of the AQT (Dynamic Frequency Response) curve from within Adobe Audition**



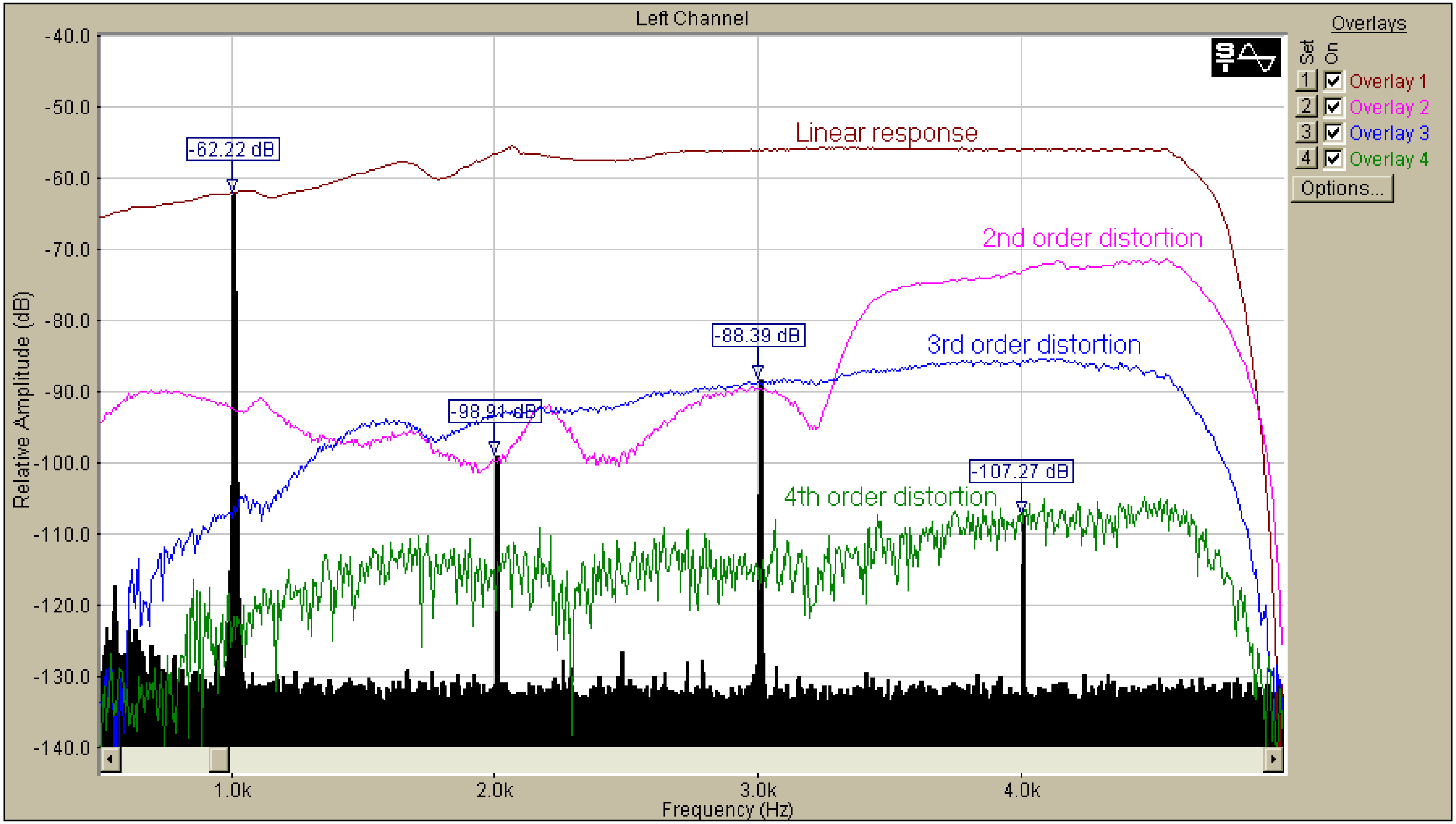


Distortion measurements



- A headphone was driven with a 1 V RMS signal, causing severe distortion in the small loudspeaker.
- The measurement was made placing the headphone on a dummy head.
- Measurements: ESS and traditional sine at 1 kHz

Distortion measurements



- **Comparison between:**
- **traditional distortion measurement with fixed-frequency sine (the black histogram)**
- **the new exponential sweep (the 4 narrow, coloured lines)**

Spatial analysis by directive impulse responses

- The initial approach was to use directive microphones for gathering some information about the spatial properties of the sound field “as perceived by the listener”
- Two apparently different approaches emerged: binaural dummy heads and pressure-velocity microphones:

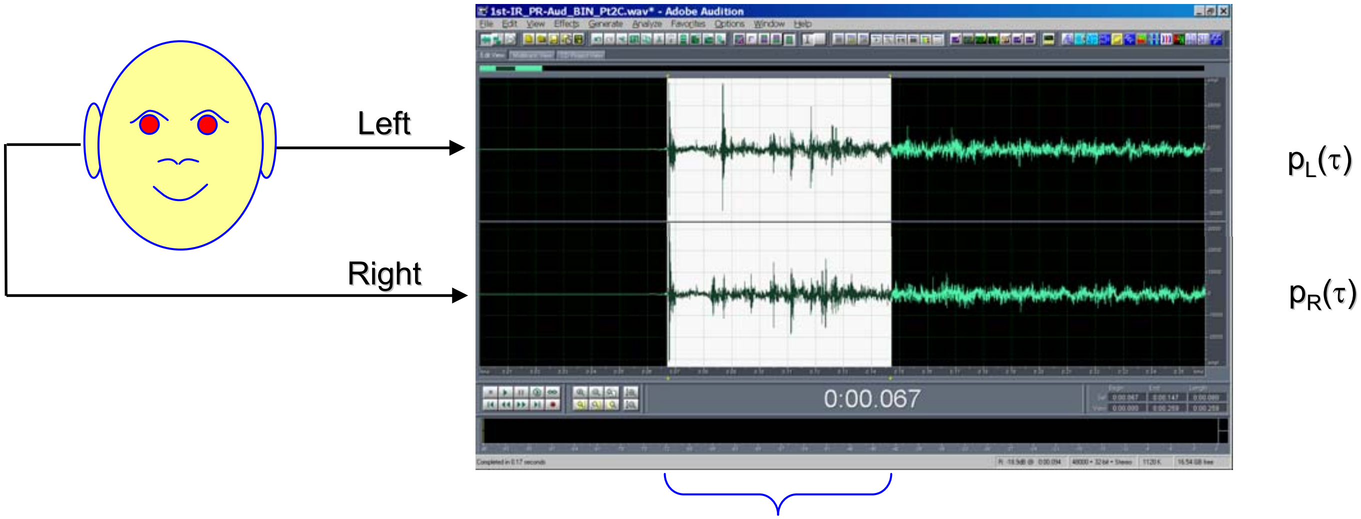
Binaural microphone (left)

and

Pressure-velocity microphone (right)

IACC "objective" spatial parameter

- It was attempted to "quantify" the "spatiality" of a room by means of "objective" parameters, based on 2-channels impulse responses measured with directive microphones
- The most famous "spatial" parameter is IACC (Inter Aural Cross Correlation), based on binaural IR measurements

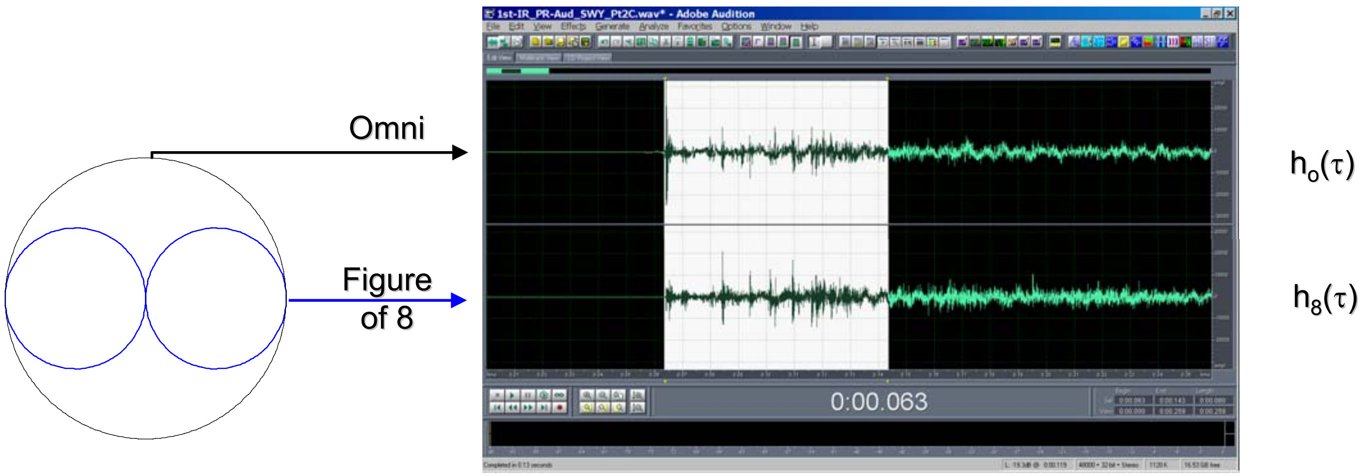


$$\rho(t) = \frac{\int_0^{80ms} p_L(\tau) \cdot p_R(\tau + t) \cdot d\tau}{\sqrt{\int_0^{80ms} p_L^2(\tau) \cdot d\tau \cdot \int_0^{80ms} p_R^2(\tau + t) \cdot d\tau}}$$

$$IACC_E = Max[\rho(t)] \qquad t \in [-1ms \ldots +1ms]$$

Lateral Fraction (LF) spatial parameter

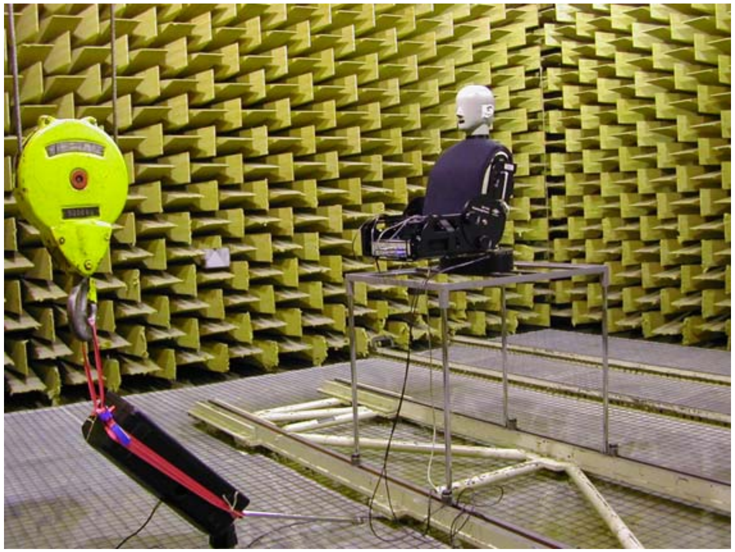
- **Another "spatial" parameter is the Lateral Fraction LF**
- **This is defined from a 2-channels impulse response, the first channel is a standard omni microphone, the second channel is a "figure-of-eight" microphone:**

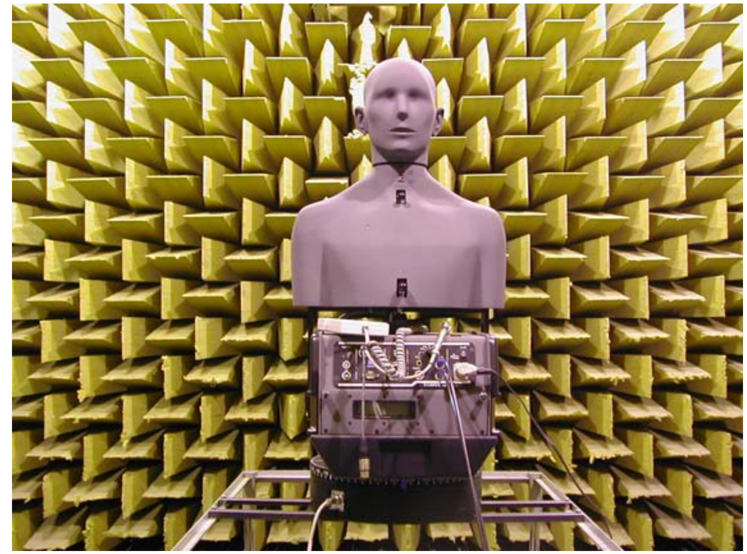


$$\mathrm{LF} = \frac{\int_{5ms}^{80ms} h_8^2(\tau)\cdot d\tau}{\int_{0ms}^{80ms} h_o^2(\tau)\cdot d\tau}$$

Are binaural measurents reproducible?

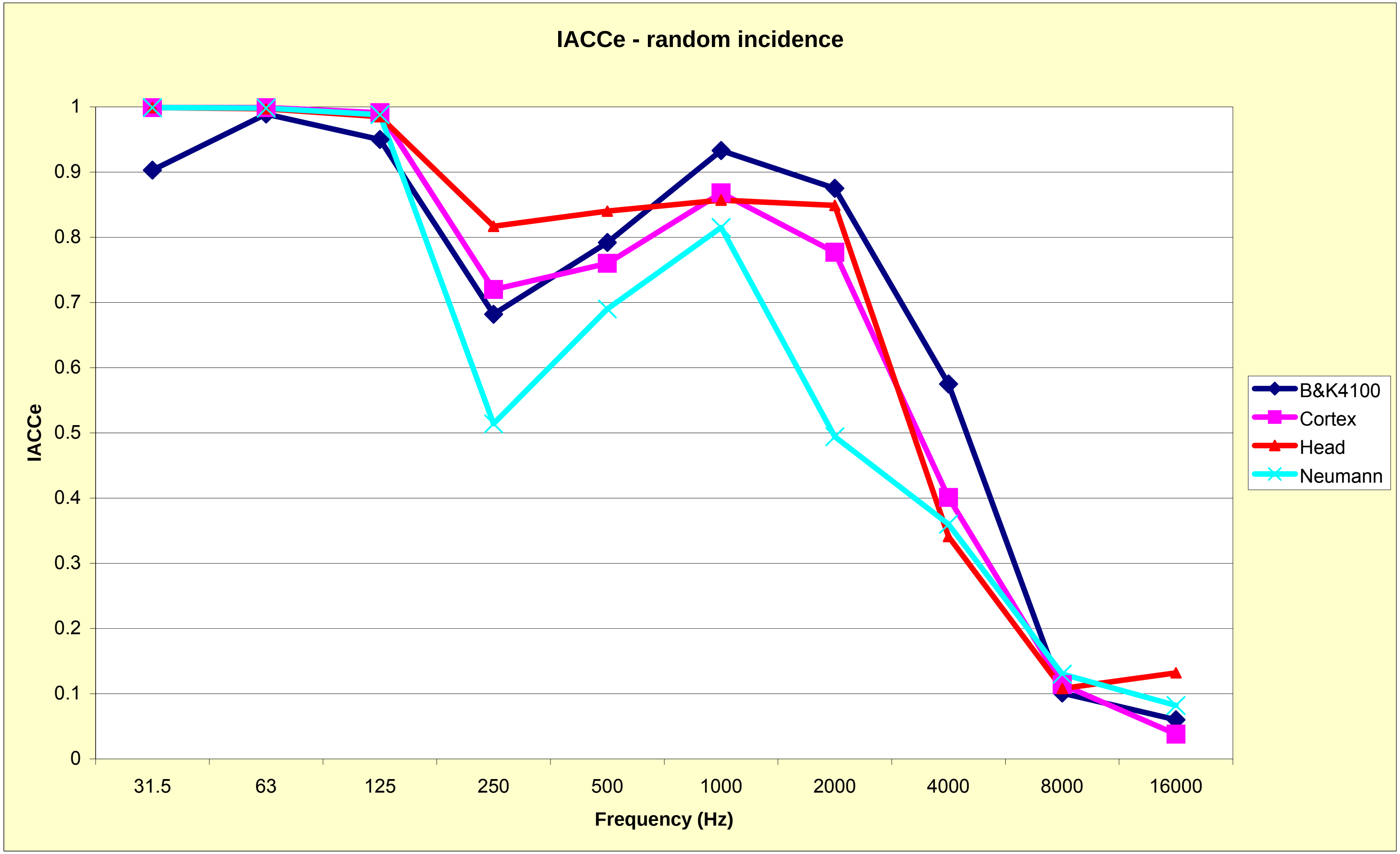
- **Experiment performed in anechoic room - same loudspeaker, same source and receiver positions, 5 binaural dummy heads**





Are IACC measurents reproducible?

- **Diffuse field - huge difference among the 4 dummy heads**

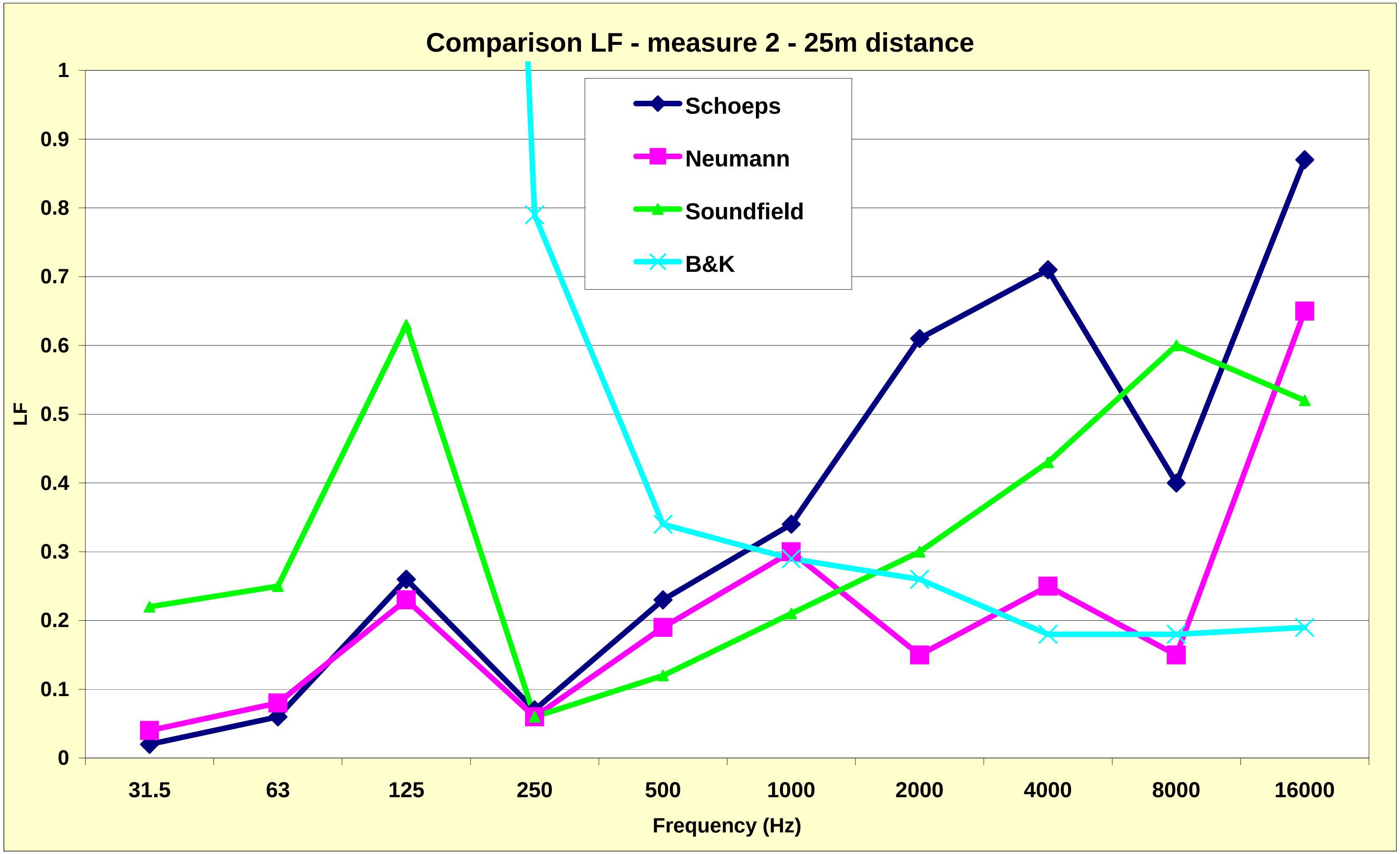


Are LF measurents reproducible?

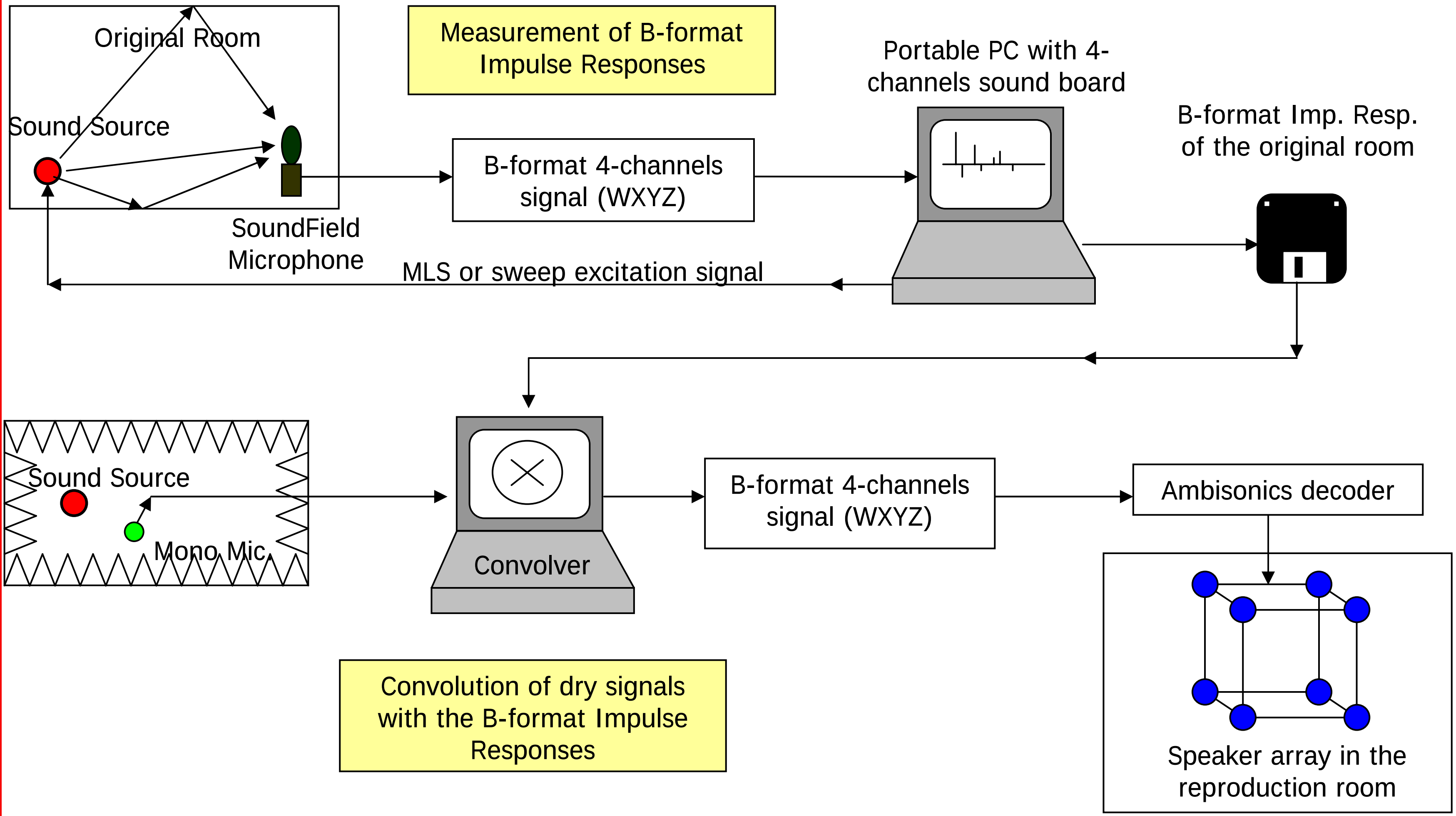
- **Experiment performed in the Auditorium of Parma - same loudspeaker, same source and receiver positions, 4 pressure-velocity microphones**

Are LF measurents reproducible?

- At 25 m distance, the scatter is really big

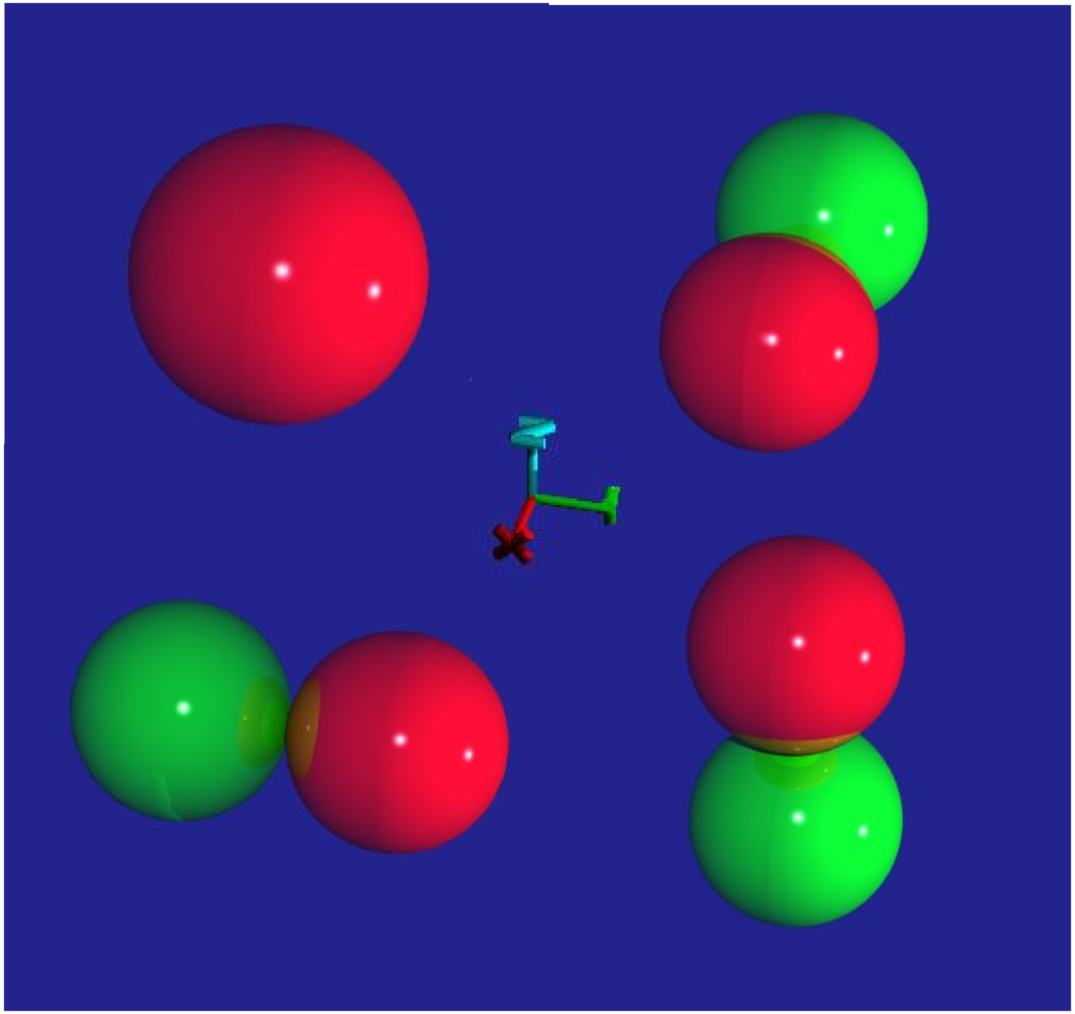


3D Impulse Response (Gerzon, 1975)



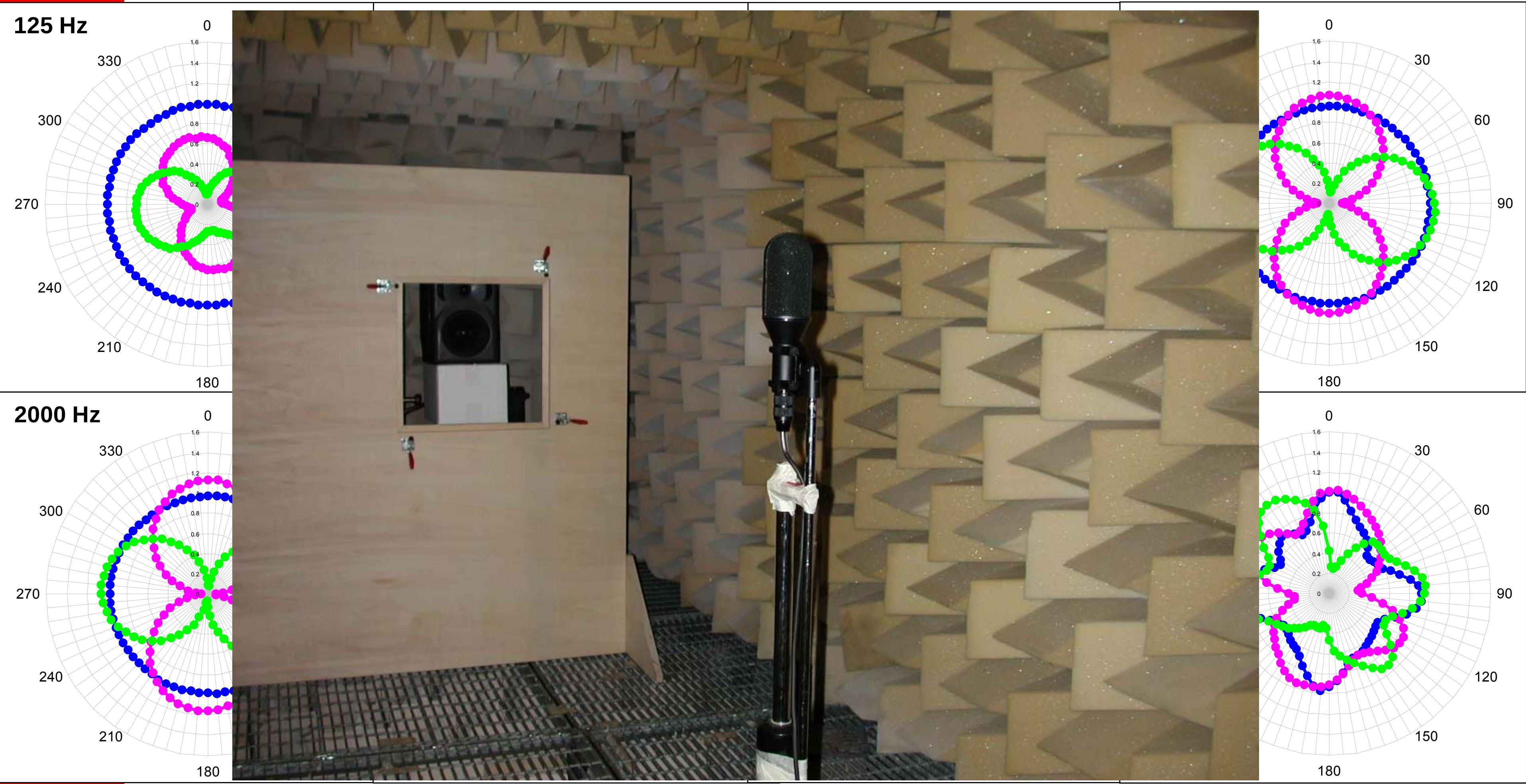
3D extension of the pressure-velocity measurements

- **The Soundfield microphone allows for simultaneous measurements of the omnidirectional pressure and of the three cartesian components of particle velocity (figure-of-8 patterns)**



Directivity of transducers

Soundfield ST-250 microphone



A-format microphone arrays

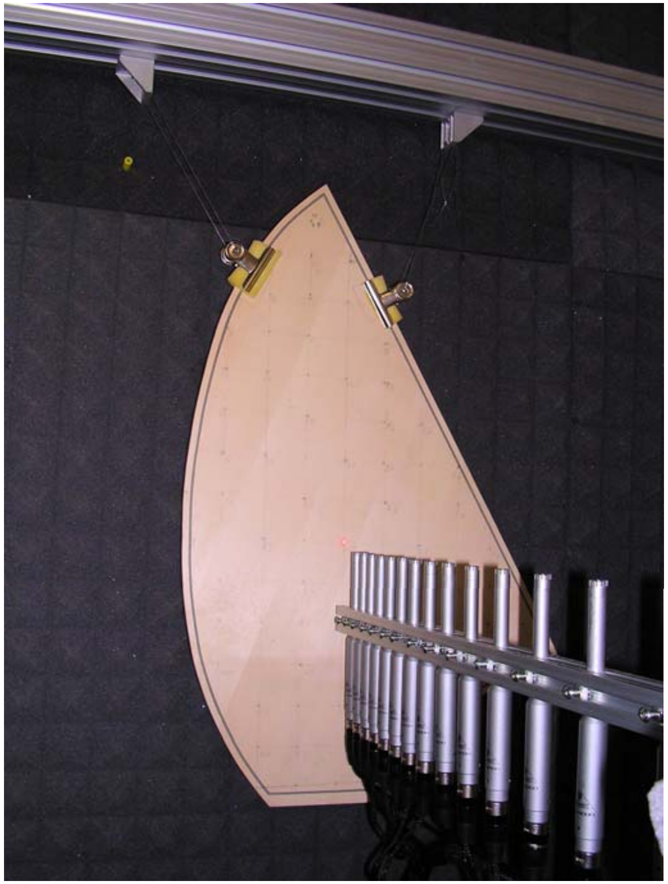
- **Today several alternatives to Soundfield microphones do exists. All of them are providing “raw” signals from the 4 capsules, and the conversion from these signals (A-format) to the standard Ambisonic signals (B-format) is performed digitally by means of software running on the computer**

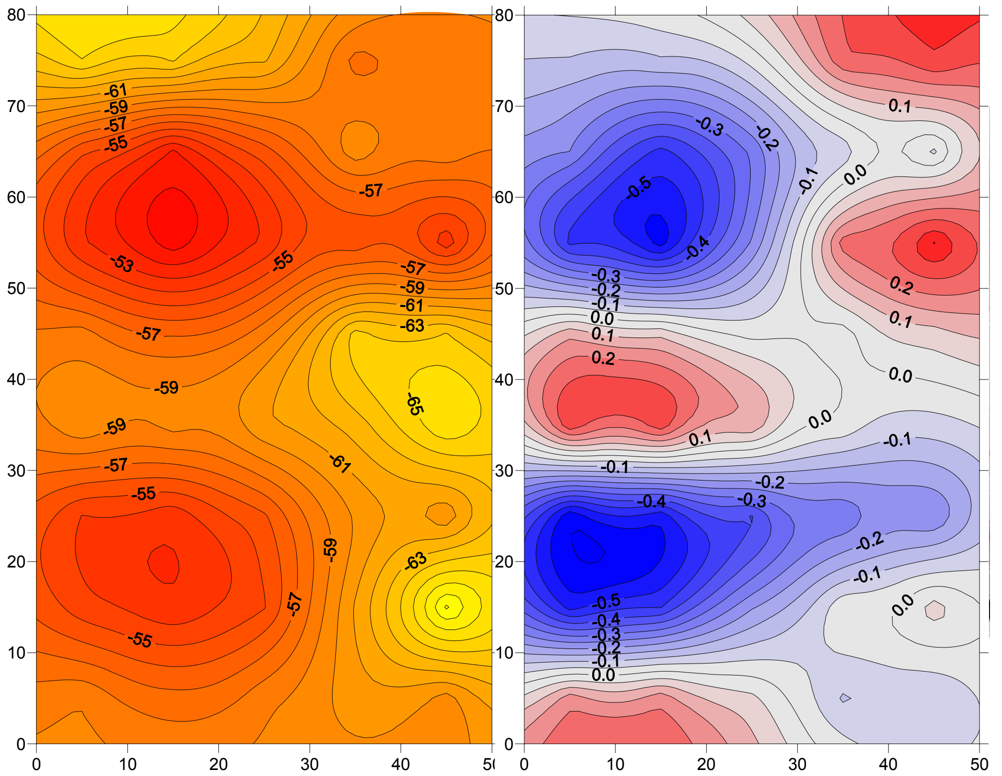
The Waves project (2003)

- The original idea of Michael Gerzon was finally put in practice in 2003, thanks to the Israeli-based company WAVES
- More than 50 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

The Ciresa project (2005)

- Measurements of the vibrations and radiated sound from wood panels
- Mapping of harmonic tables by means on an XY scanner
- Pressure measured by means of a linear microphone array
- Velocity measured by means of a laser vibrometer





Radiated sound pressure level (Left) and vibration velocity (Right)

Conclusions

- The sine sweep method revealed to be systematically superior to the MLS & TDS methods for measuring electroacoustical impulse responses
- The ESS method also allows for measurement of not-linear devices and to assess harmonic distortion
- Current limitation for spatial analysis in room acoustis is due to transducers (loudspeakers and microphones)
- A new generation of loudspeakers and microphones, made of massive arrays, is under development.
- The “harmonic orders” impulse responses obtained by the exponential sine sweep method can be used for not-linear convolution, which yields more realistic auralization