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Advancements in impulse response measurements by sine sweeps

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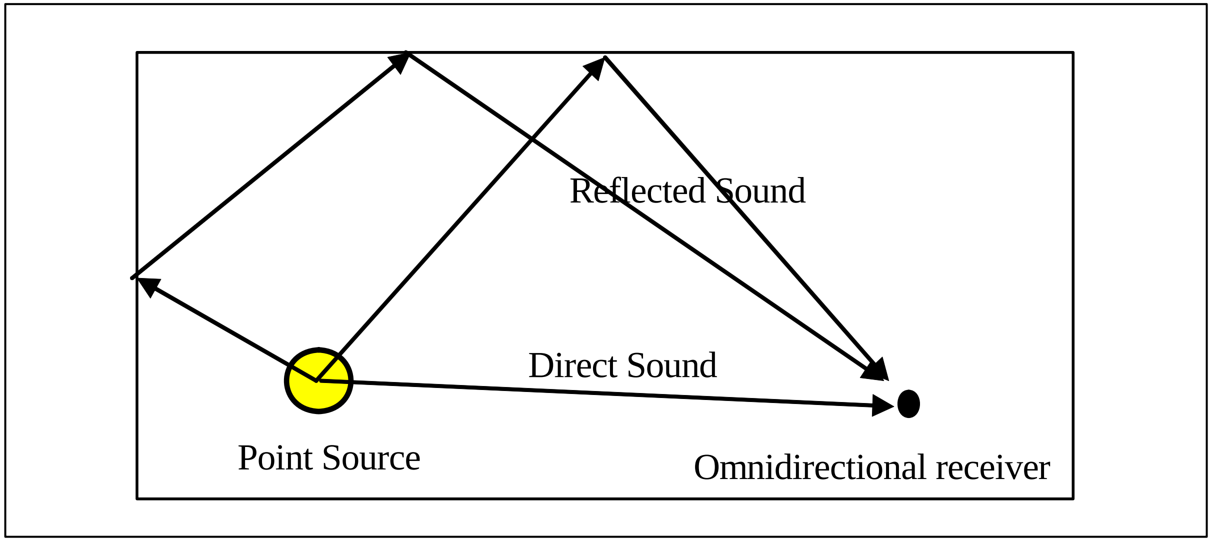
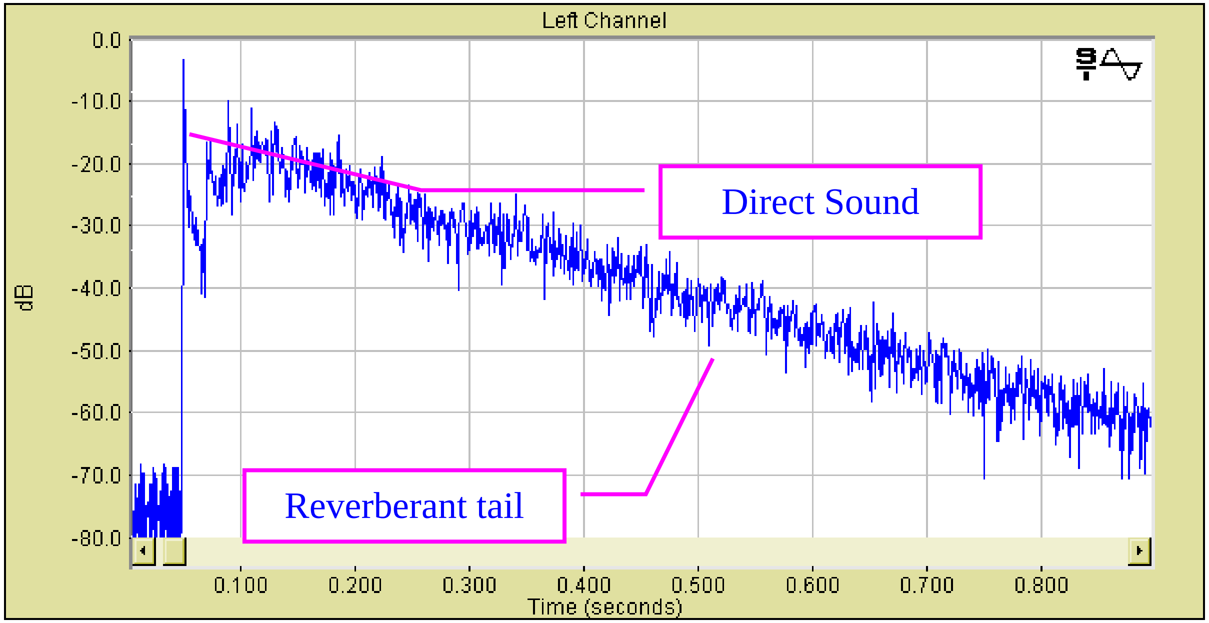
Topics

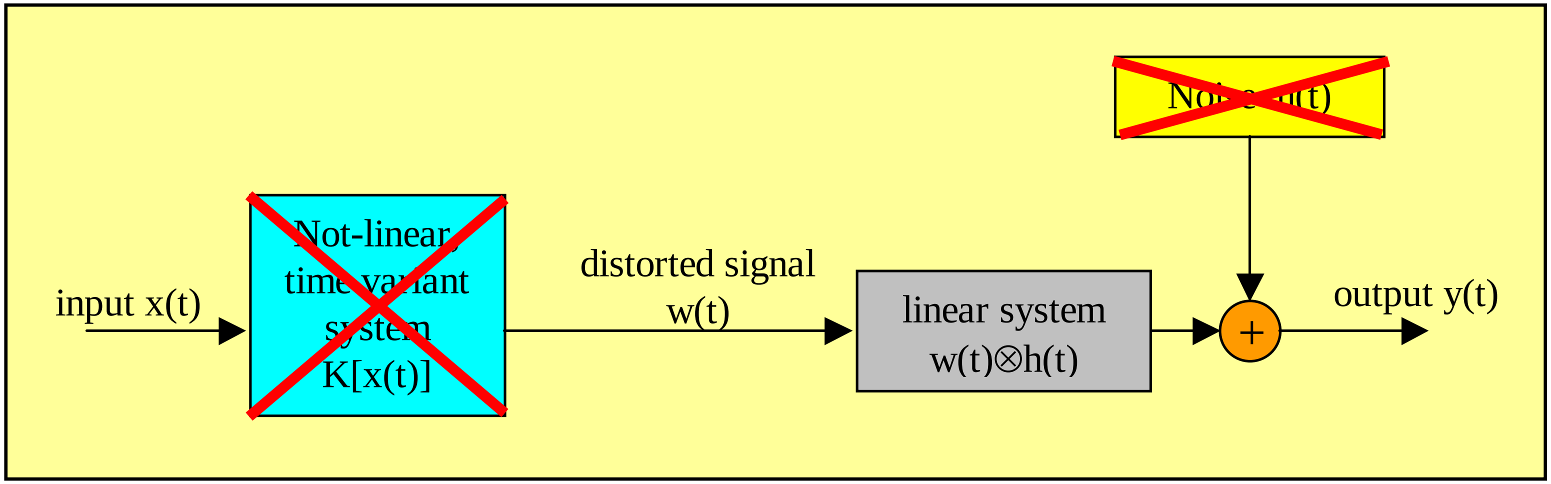
- Basics of Exponential Sine Sweep (ESS) method
- Pre-ringing at high and low frequency before the arrival of the direct sound pulse
- Pre/post equalization of the test signal performed in a way which avoids time-smearing of the impulse response
- Sensitivity to abrupt pulsive noises during the measurement
- Skewing of the measured impulse response when the playback and recording digital clocks are mismatched
- Cancellation of high frequencies in the late part of the tail when performing synchronous averaging
- Directivity of transducers (loudspeakers and microphones)

Basic sound propagation scheme





Measurement process



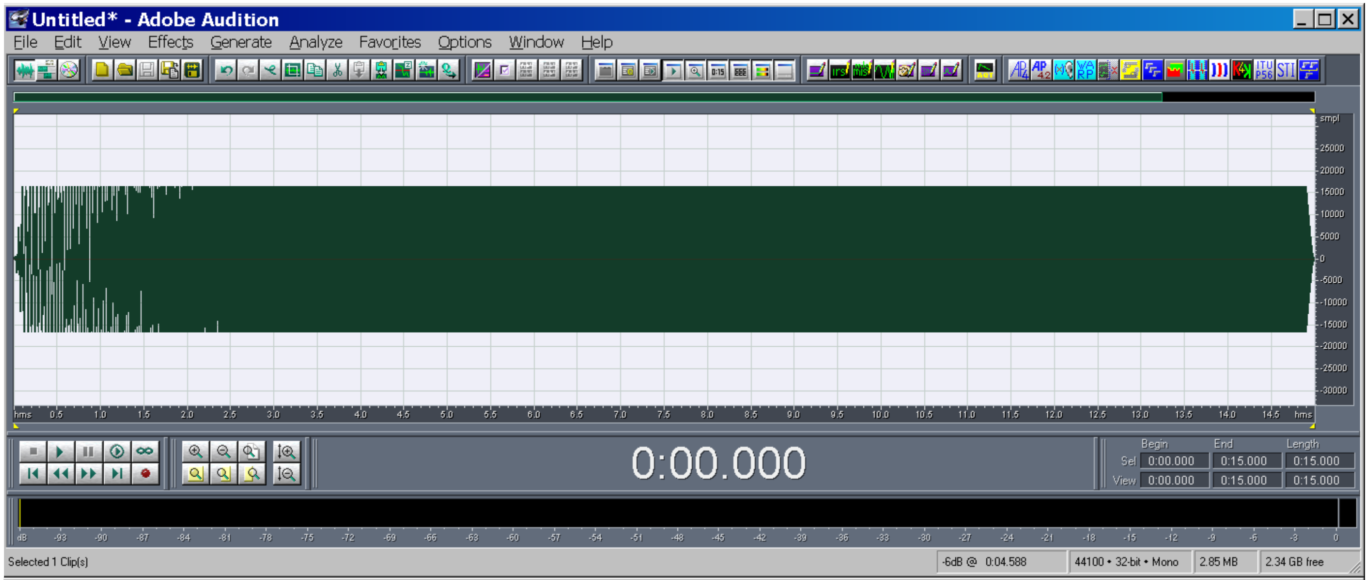
- The desidered result is the linear impulse response of the acoustic propagation h(t). It can be recovered by knowing the test signal x(t) and the measured system output y(t).
- It is necessary to exclude the effect of the not-linear part K and of the background noise n(t).

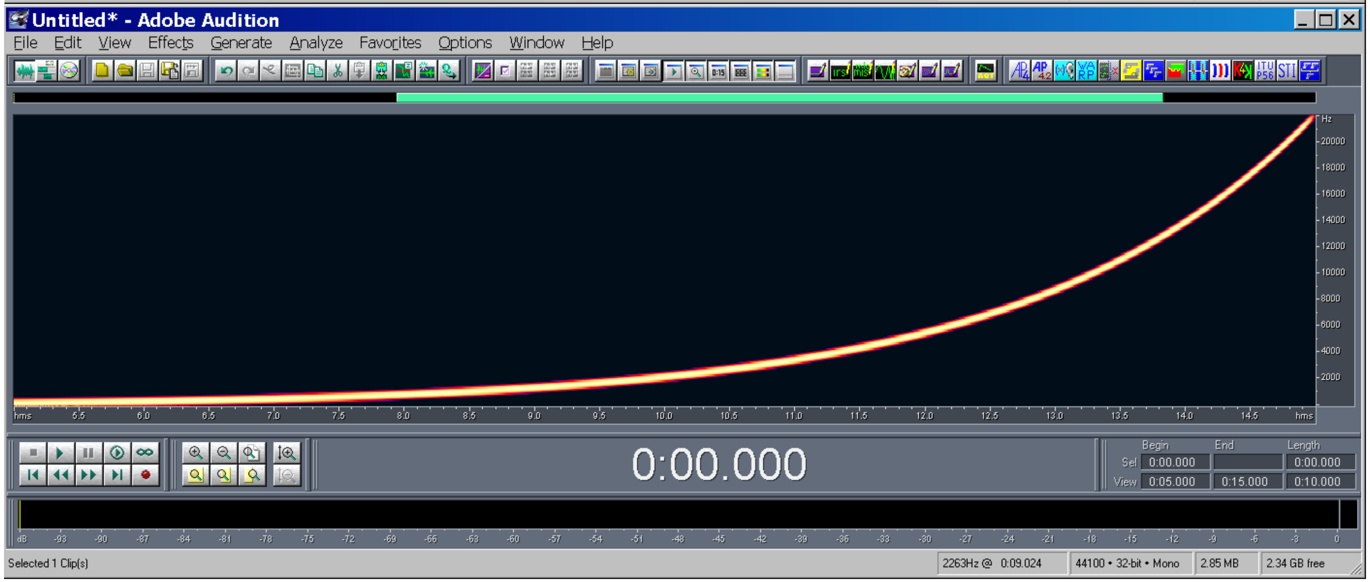
Exponential Sine Sweep method

- **x(t) is a sine signal, which frequency is varied exponentially with time, starting at f_1 and ending at f_2.**

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

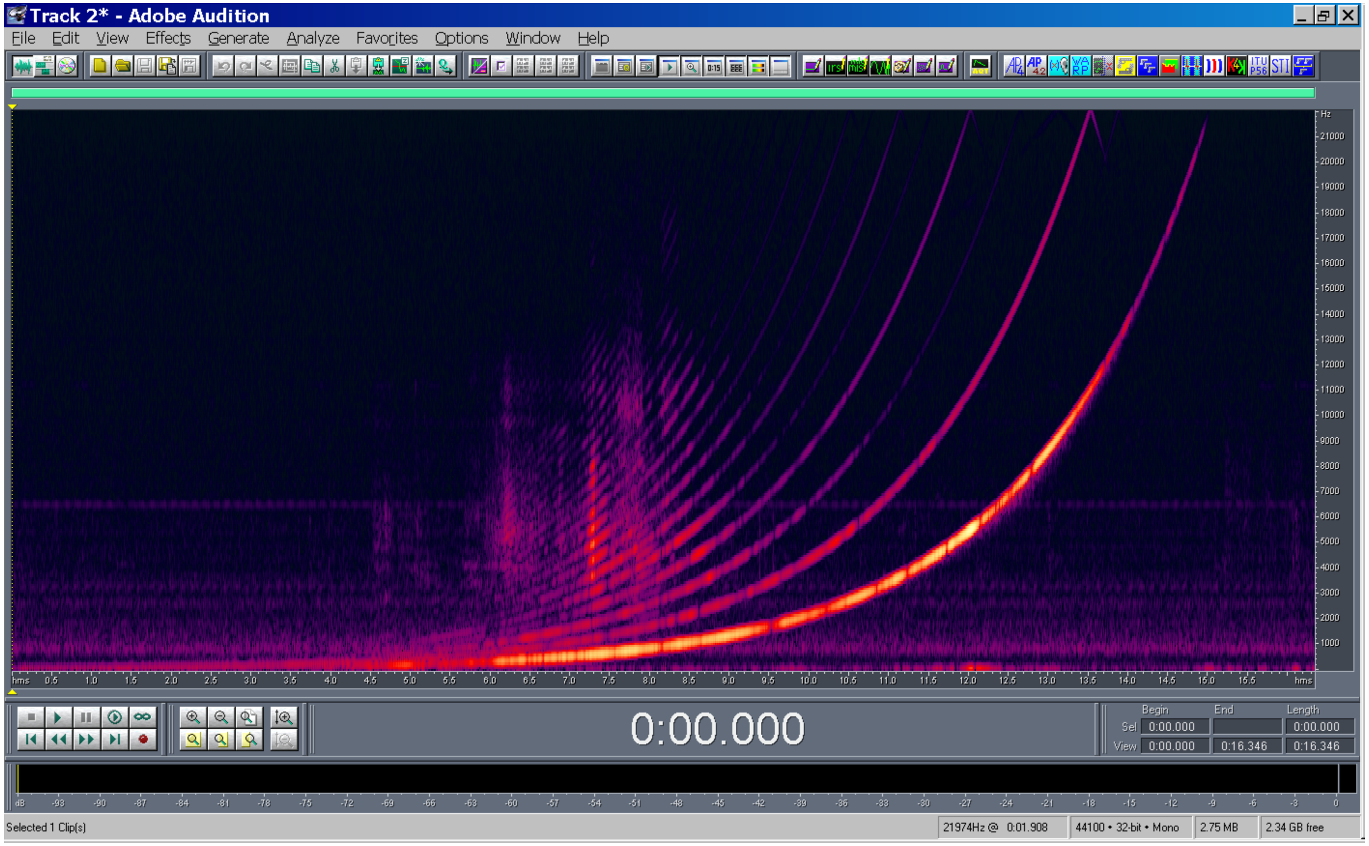
Test Signal – x(t)





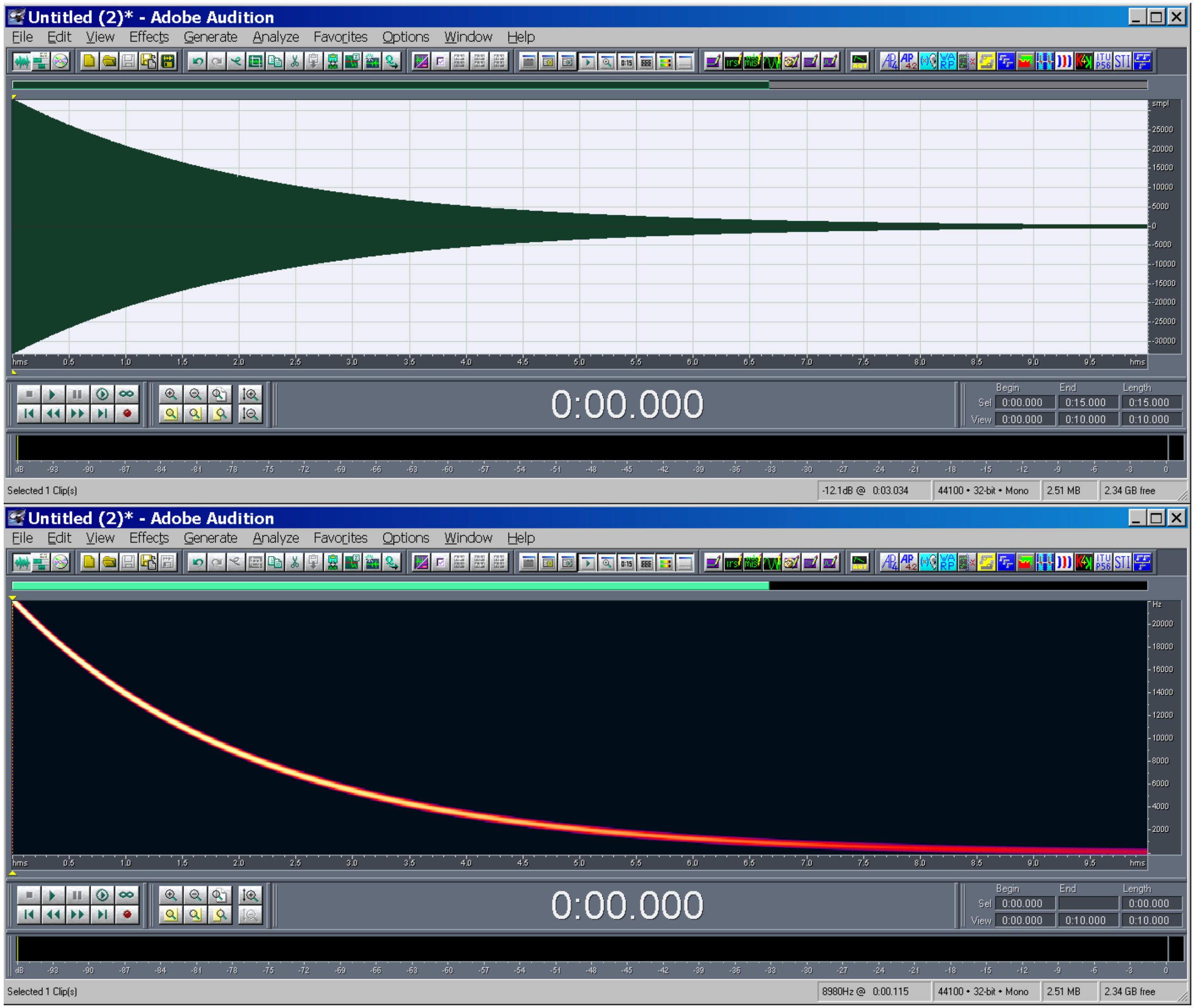
Measured signal - y(t)

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



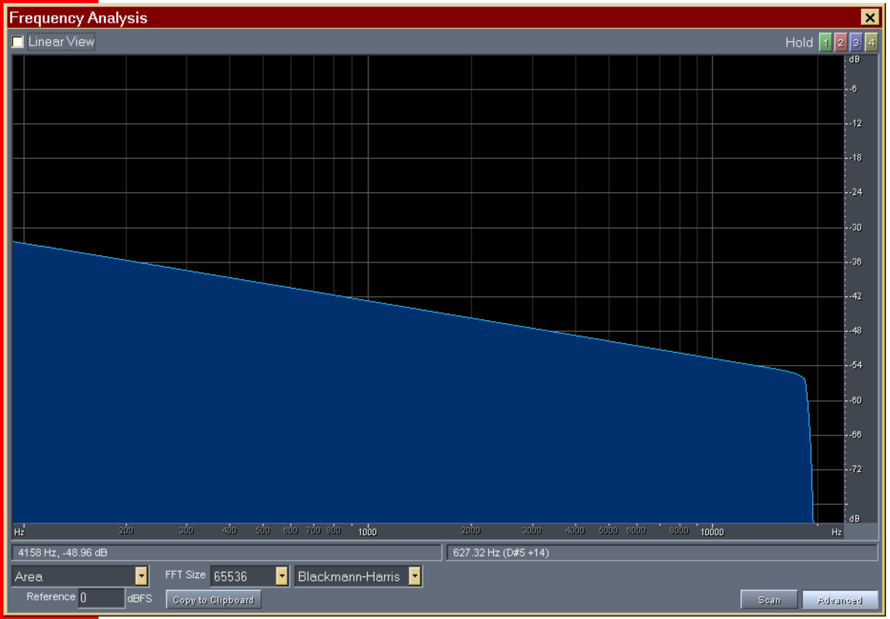
Inverse Filter – z(t)

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

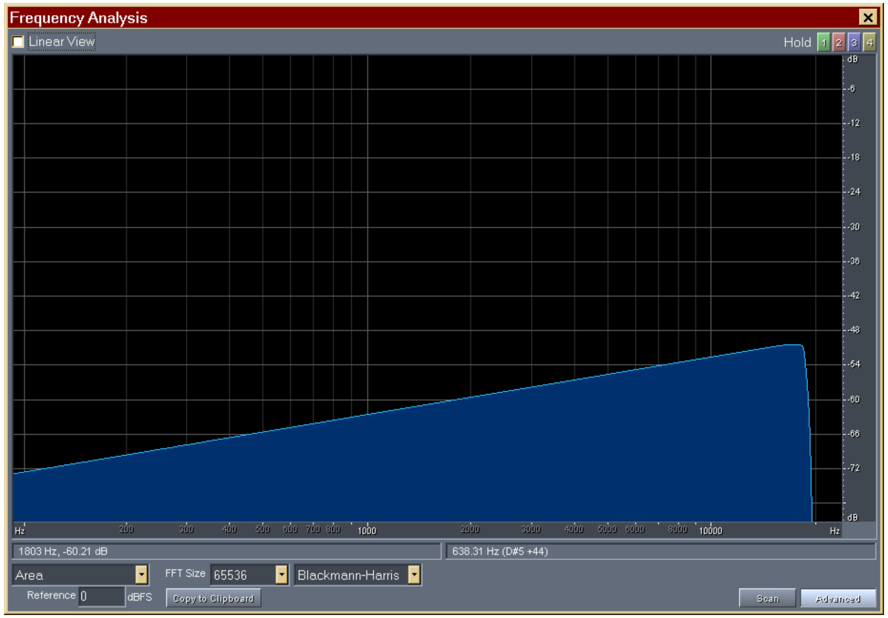


Deconvolution of Log Sine Sweep

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

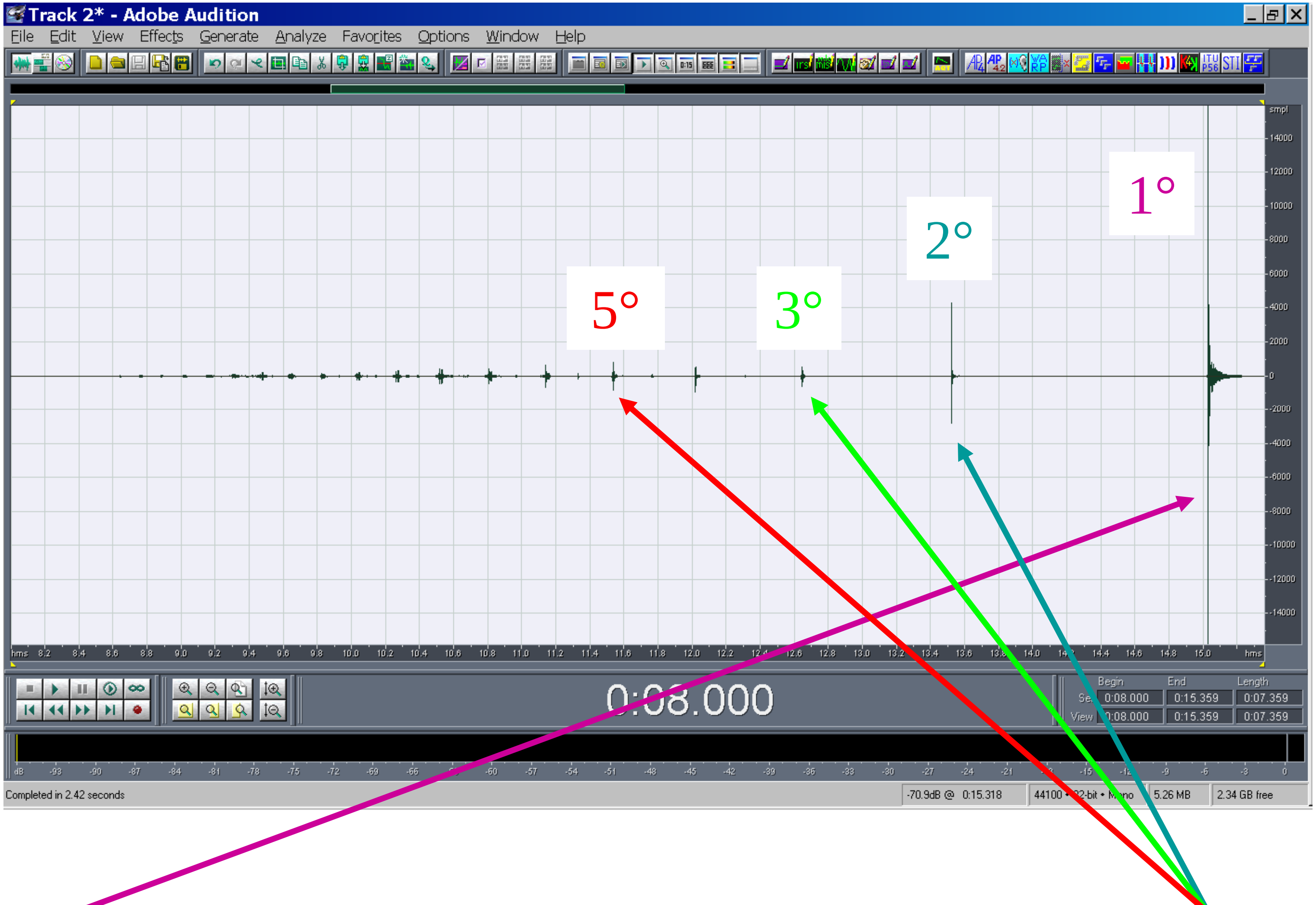


Test Signal x(t)



Inverse Filter z(t)

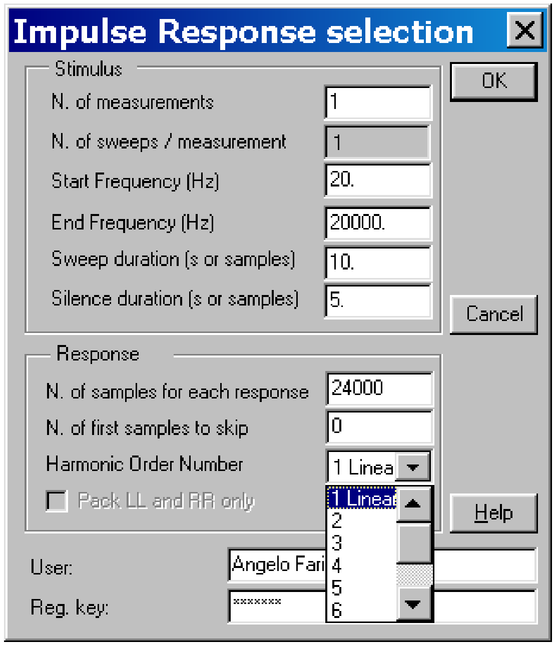
Result of the deconvolution

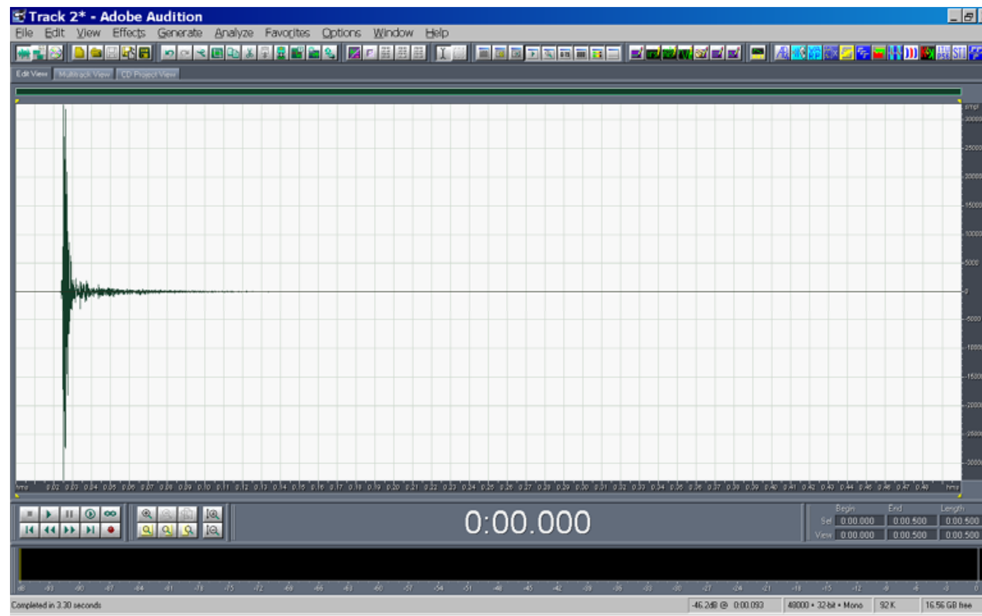


The last impulse response is the linear one, the preceding are the harmonics distortion products of various orders

IR Selection

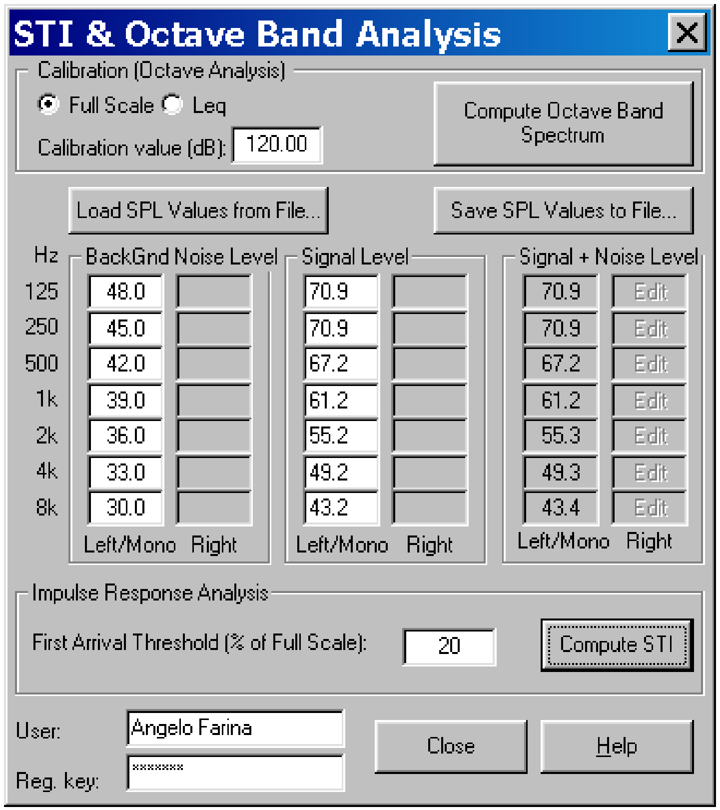
- **After the sequence of impulse responses has been obtained, it is possible to select and extract just one of them:**

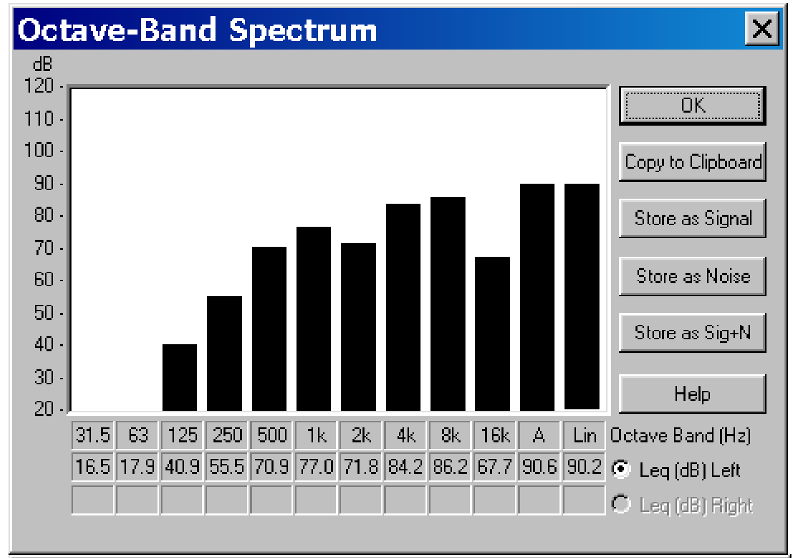


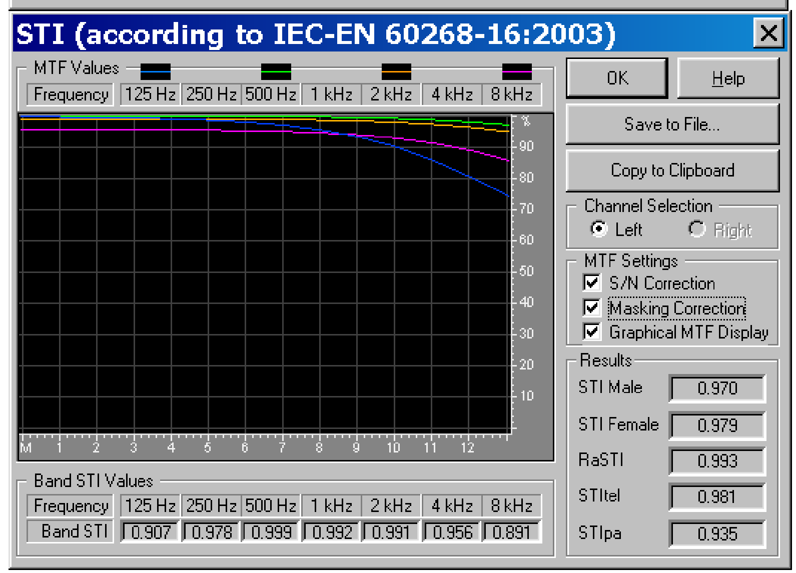


Post processing of impulse responses

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

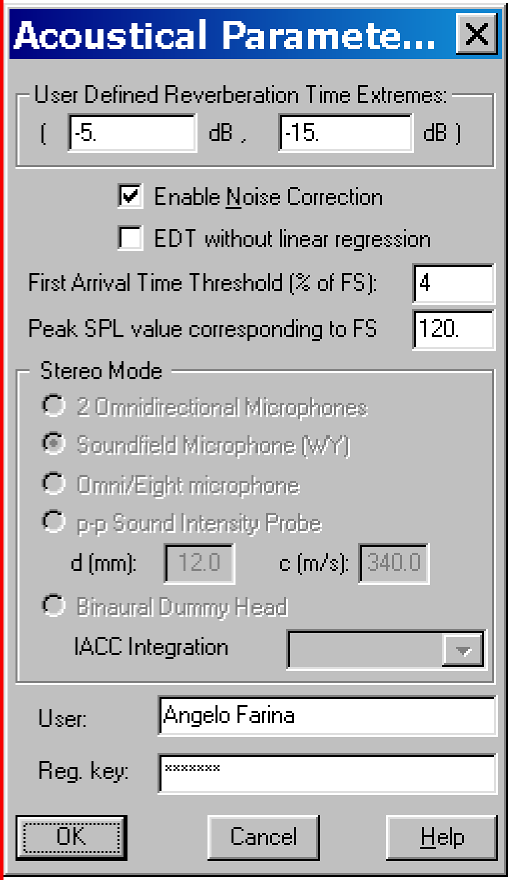


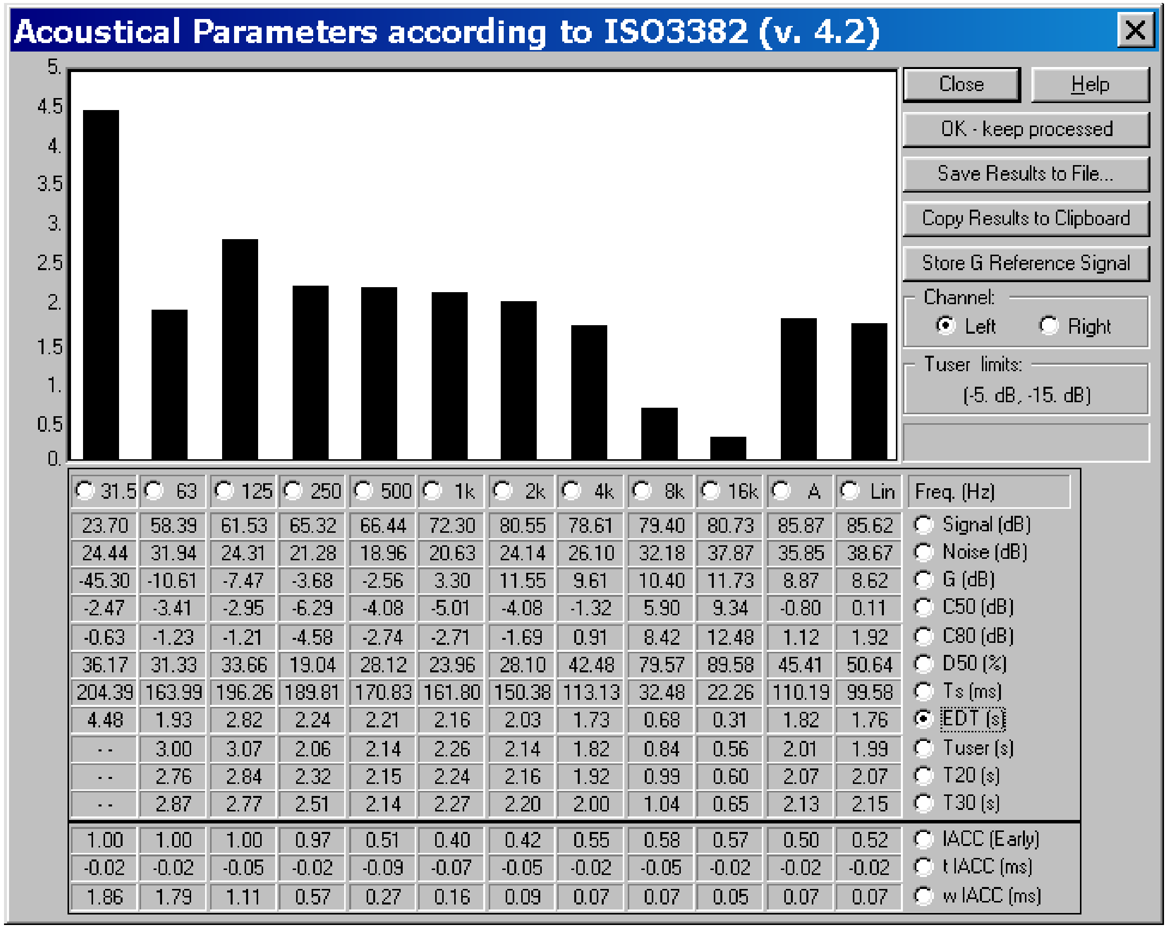




Post processing of impulse responses

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

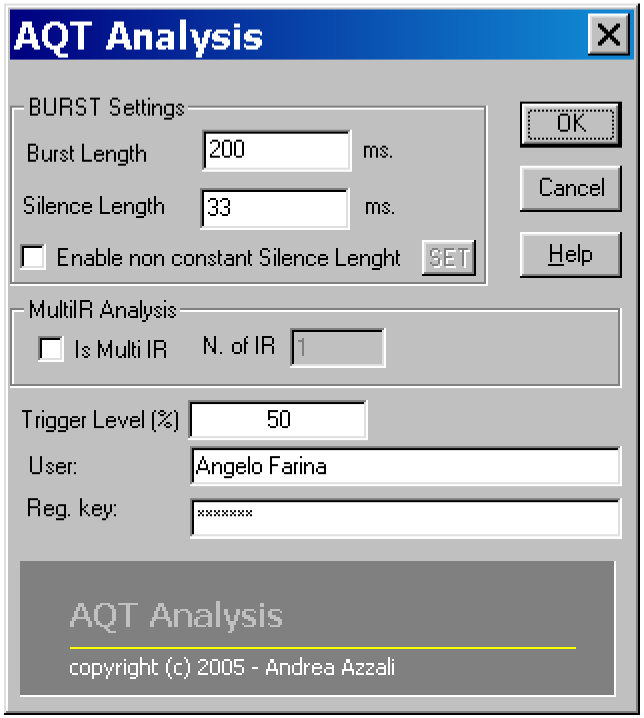


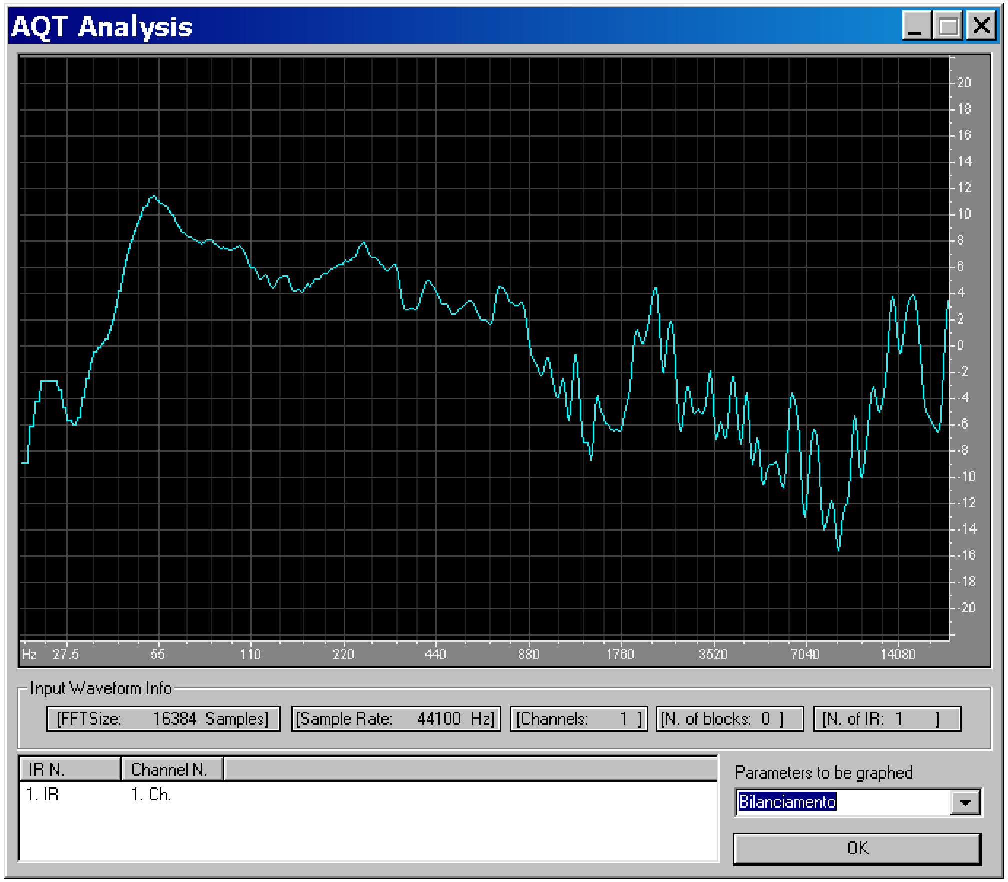


31.5	63	125	250	500	1k	2k	4k	8k	16k	A	Lin	Freq. (Hz)
23.70	58.39	61.53	65.32	66.44	72.30	80.55	78.61	79.40	80.73	85.87	85.62	Signal (dB)
24.44	31.94	24.31	21.28	18.96	20.63	24.14	26.10	32.18	37.87	35.85	38.67	Noise (dB)
-45.30	-10.61	-7.47	-3.68	-2.56	3.30	11.55	9.61	10.40	11.73	8.87	8.62	G (dB)
-2.47	-3.41	-2.95	-6.29	-4.08	-5.01	-4.08	-1.32	5.90	9.34	-0.80	0.11	C50 (dB)
-0.63	-1.23	-1.21	-4.58	-2.74	-2.71	-1.69	0.91	8.42	12.48	1.12	1.92	C80 (dB)
36.17	31.33	33.66	19.04	28.12	23.96	28.10	42.48	79.57	89.58	45.41	50.64	D50 (%)
204.39	163.99	196.26	189.81	170.83	161.80	150.38	113.13	32.48	22.26	110.19	99.58	Ts (ms)
4.48	1.93	2.82	2.24	2.21	2.16	2.03	1.73	0.68	0.31	1.82	1.76	EDT (s)
- -	3.00	3.07	2.06	2.14	2.26	2.14	1.82	0.84	0.56	2.01	1.99	Tuser (s)
- -	2.76	2.84	2.32	2.15	2.24	2.16	1.92	0.99	0.60	2.07	2.07	T20 (s)
- -	2.87	2.77	2.51	2.14	2.27	2.20	2.00	1.04	0.65	2.13	2.15	T30 (s)
1.00	1.00	1.00	0.97	0.51	0.40	0.42	0.55	0.58	0.57	0.50	0.52	IACC (Early)
-0.02	-0.02	-0.05	-0.02	-0.09	-0.07	-0.05	-0.02	-0.05	-0.02	-0.02	-0.02	t IACC (ms)
1.86	1.79	1.11	0.57	0.27	0.16	0.09	0.07	0.07	0.05	0.07	0.07	w IACC (ms)

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**





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Pre-ringing at high and low frequency

- **Pre-ringing at high frequency due to improper fade-out**

This picture shows the preringing obtained deconvolving directly the test signal, without passing through the system under test

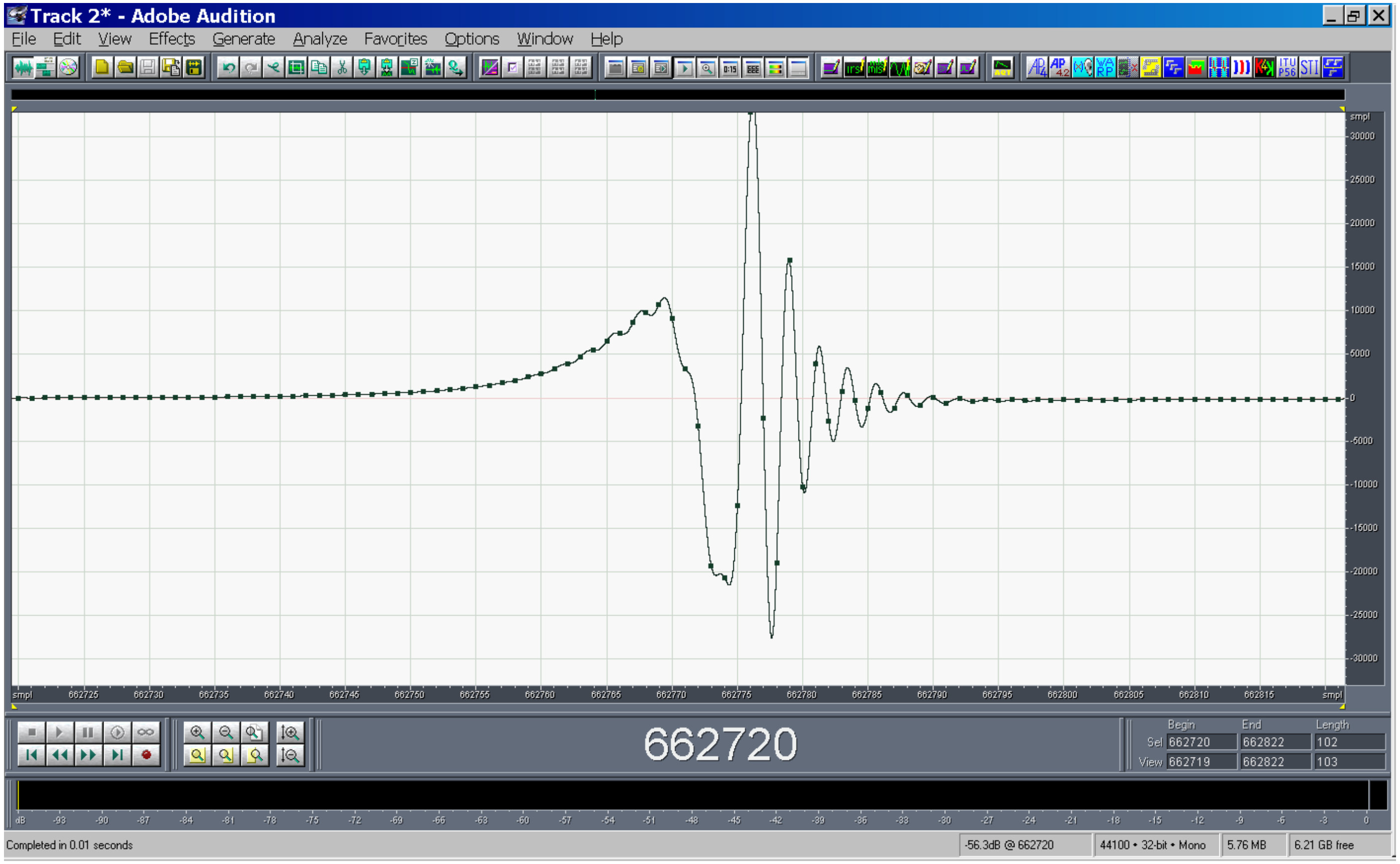
Pre-ringing at high and low frequency

- **Perfect Dirac's delta after removing the fade-out**

This picture shows the result obtained deconvolving directly the test signal, without passing through the system under test, and employing a sine sweep going up to the Nyquist frequency

Pre-ringing at high and low frequency

- **Pre-ringing at low frequency due to a bad sound card featuring frequency-dependent latency**



This artifact can be corrected if the frequency-dependent latency remains the same, by creating a suitable inverse filter with the Kirkeby method

Kirkeby inverse filter

- **The Kirkeby inverse filter is computed inverting the measured IR**

1) The IR to be inverted is FFT transformed to frequency domain:

$$H(f) = FFT\,[h(f)]$$

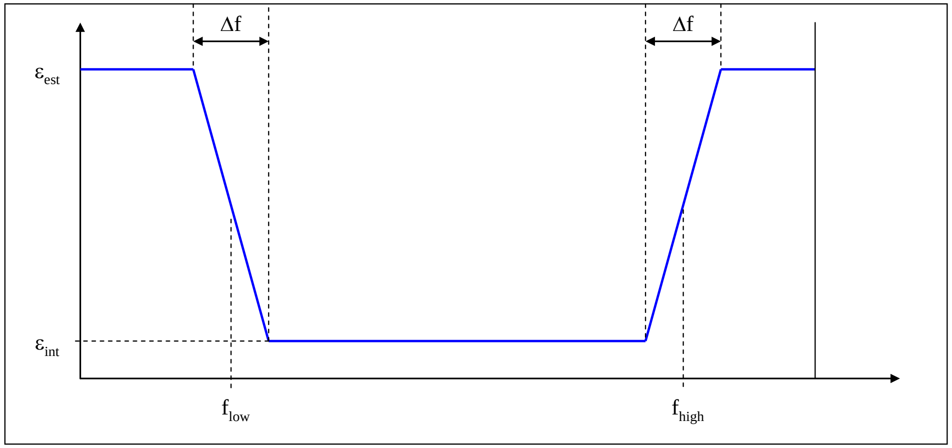
2) The computation of the inverse filter is done in frequency domain:

$$C(f) = \frac{Conj[H(f)]}{Conj[H(f)] \cdot H(f) + \varepsilon(f)}$$

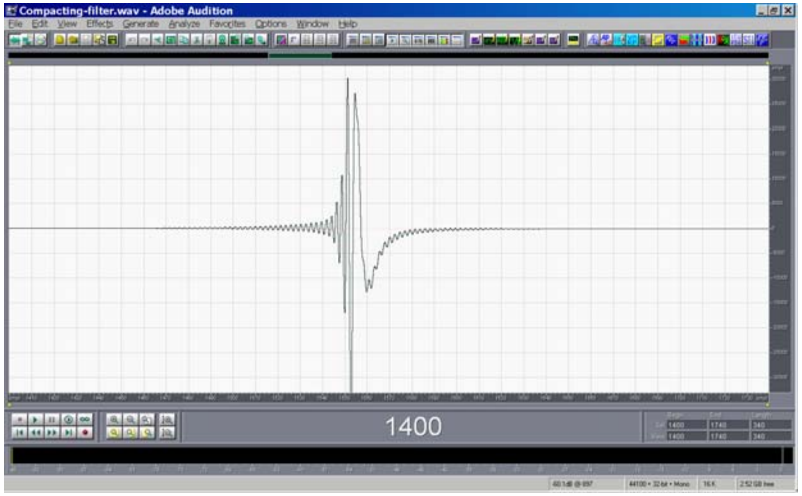
Where $\varepsilon(f)$ is a small, frequency-dependent regularization parameter

3) Finally, an IFFT brings back the inverse filter to time domain:

$$c(t) = IFFT\,[C(f)]$$



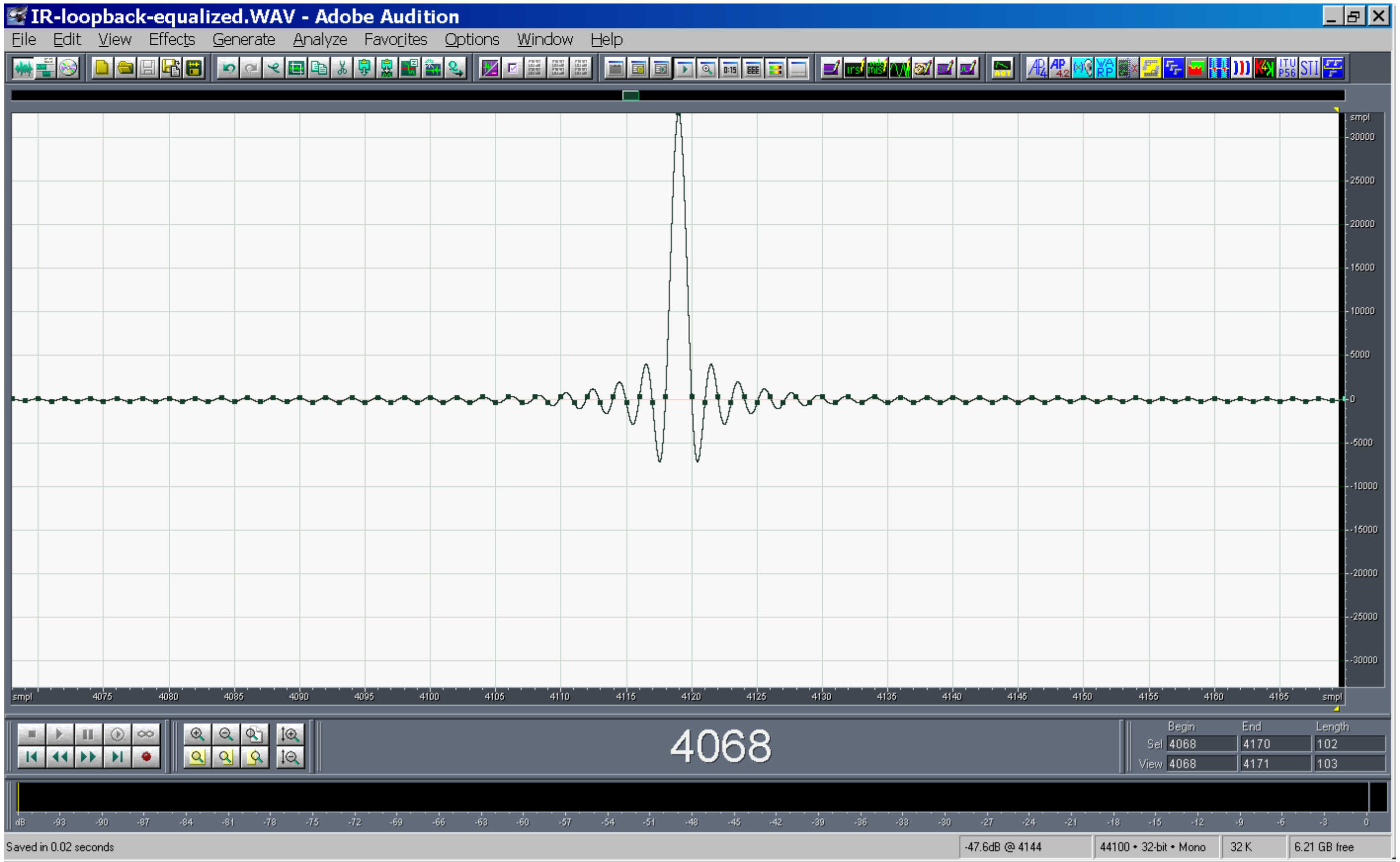
Frequency-dependent regularization parameter



Inverse filter

Pre-ringing at high and low frequency

- **Convolving the time-smeared IR with the Kirkeby compacting filter, a very sharp IR is obtained**



The same method can also be applied for correcting the response of the loudspeaker/microphone system, if an anechoic preliminary test is done

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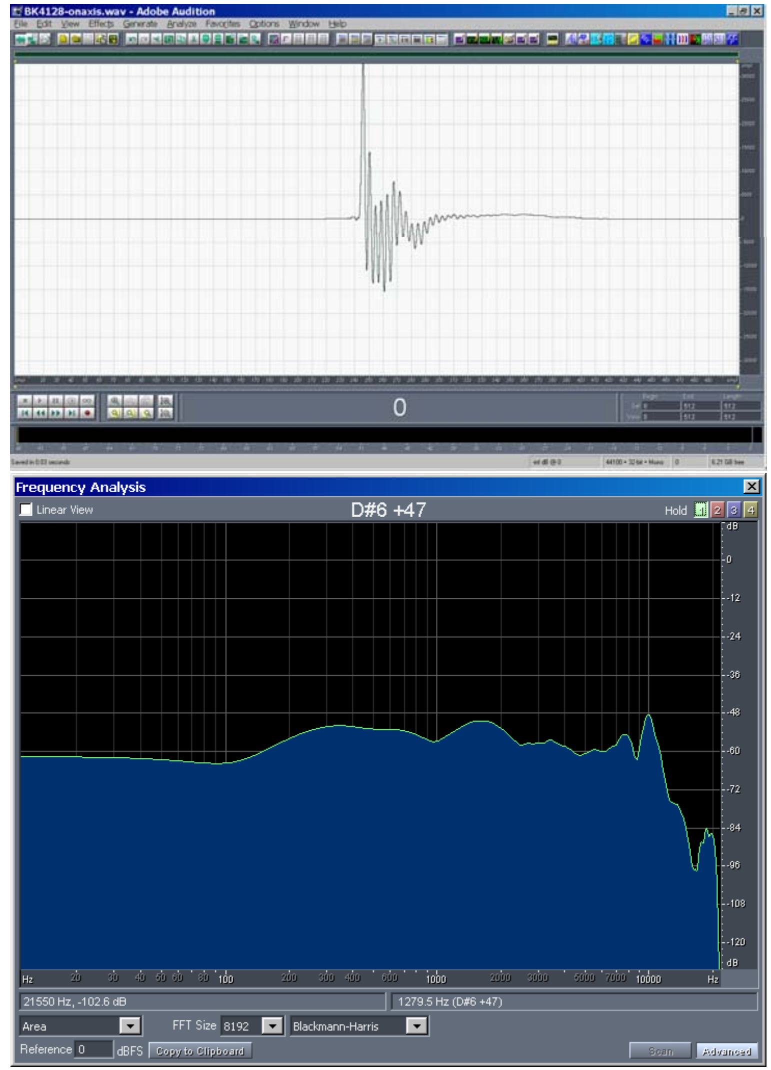
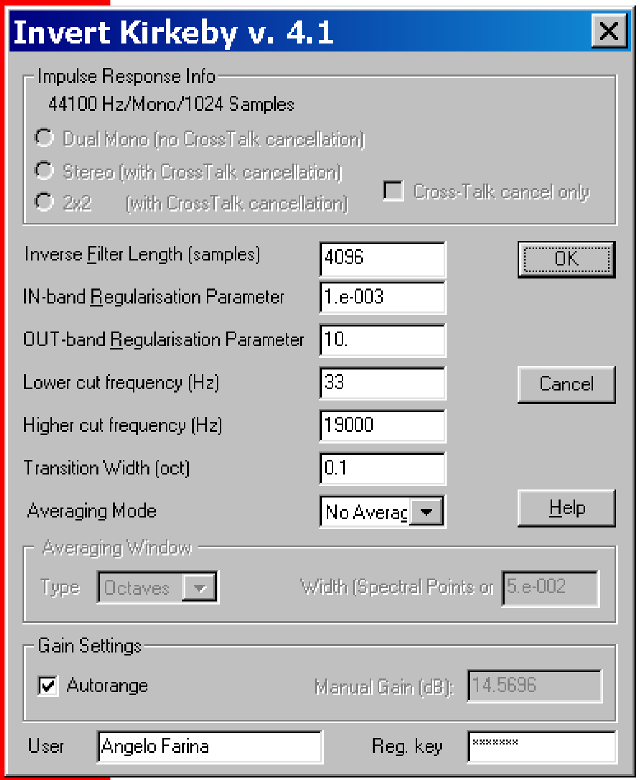
Equalization of the whole system

- **An anechoic measurement is first performed**



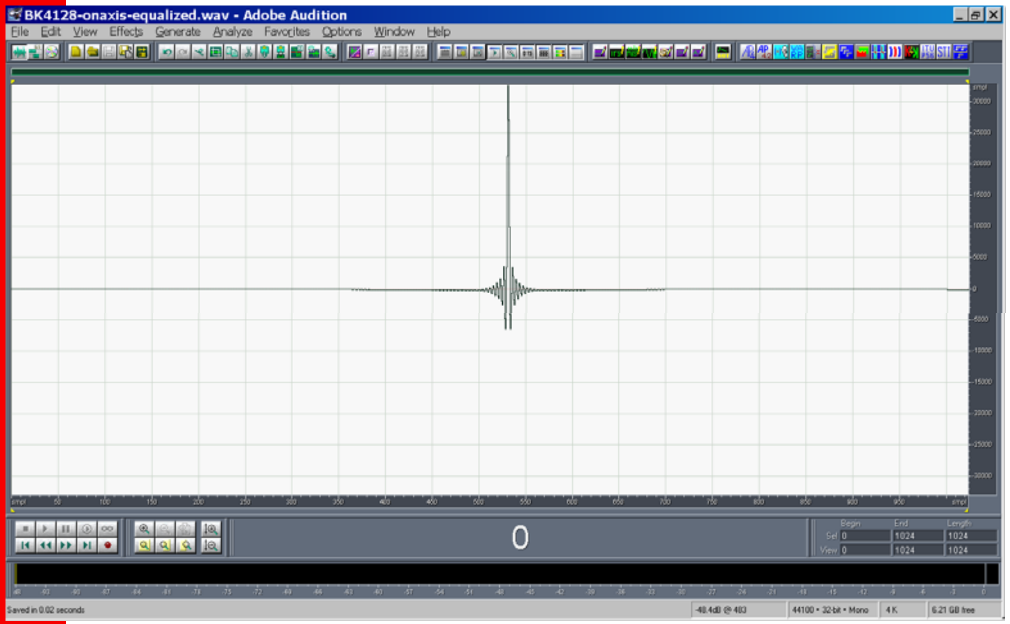
Equalization of the whole system

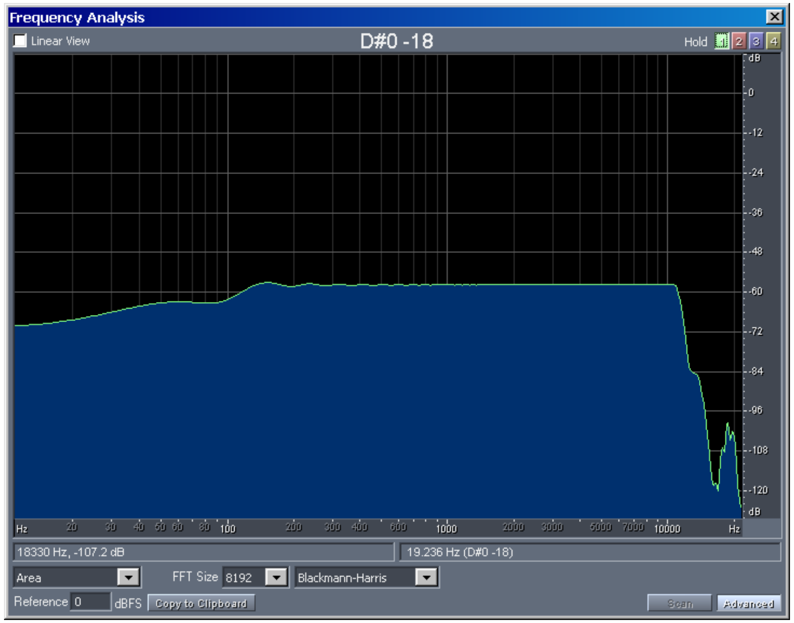
- **A suitable inverse filter is generated with the Kirkeby method by inverting the anechoic measurement**



Equalization of the whole system

- The inverse filter can be either pre-convolved with the test signal or post-convolved with the result of the measurement
- Pre-convolution usually reduces the SPL being generated by the loudspeaker, resulting in worst S/N ratio
- On the other hand, post-convolution can make the background noise to become “coloured”, and hence more perciptible
- The resulting anechoic IR becomes almost perfectly a Dirac’s Delta function:





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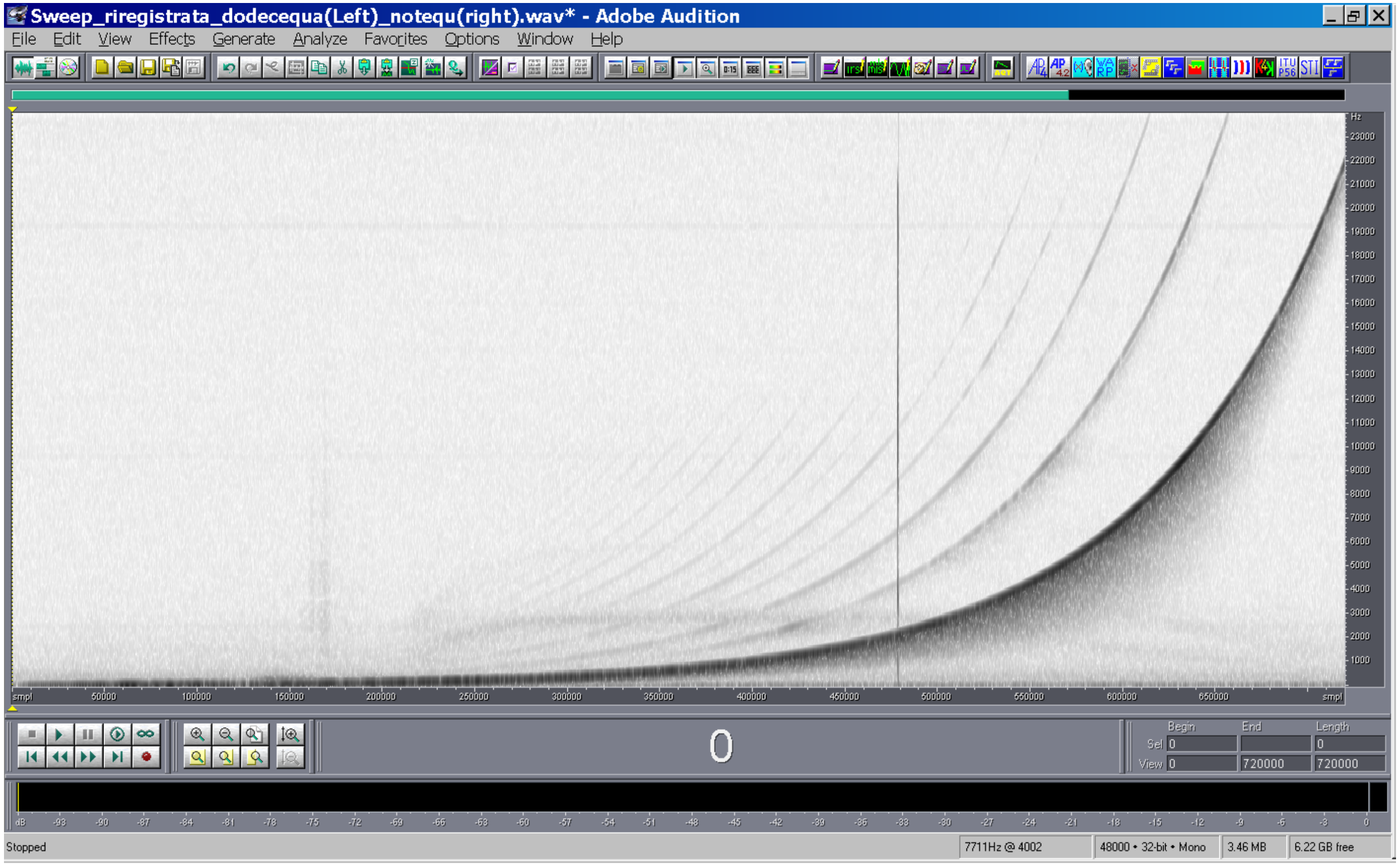
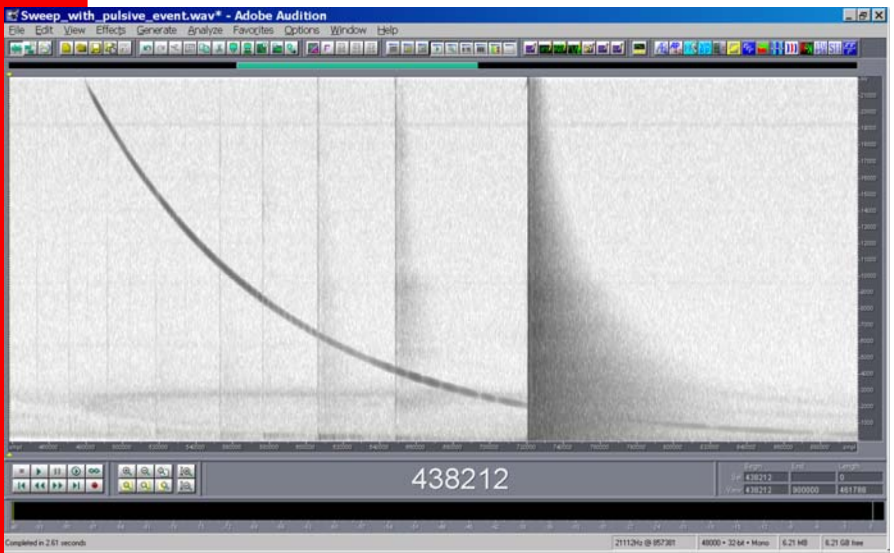
Sensitivity to abrupt pulsive noises

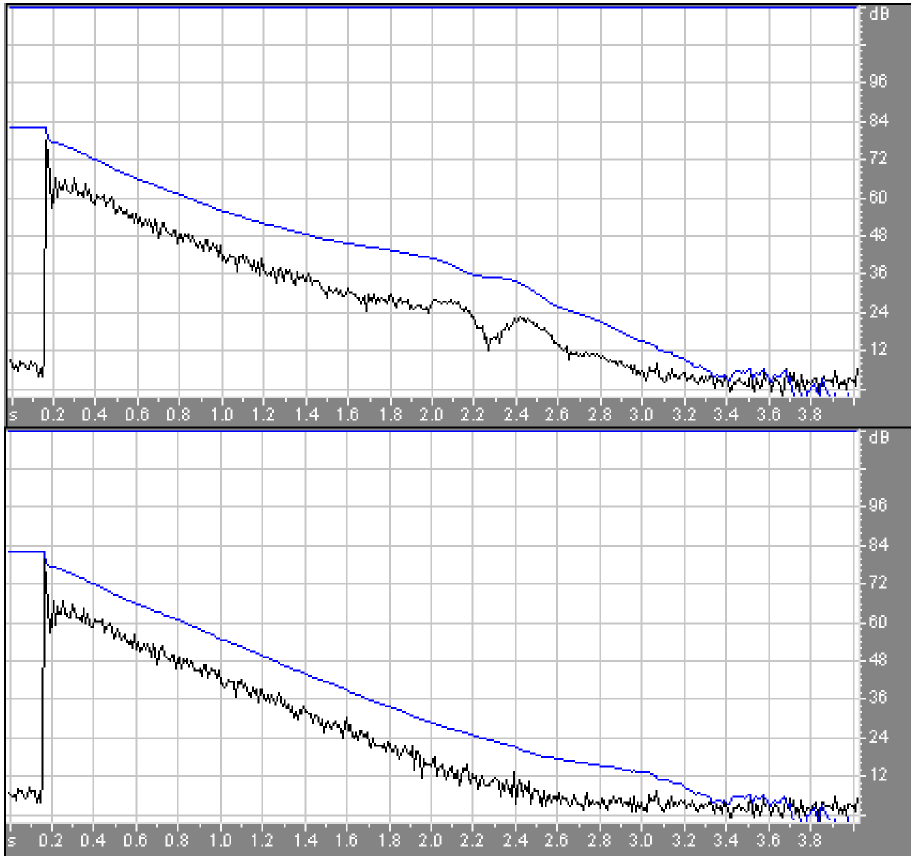
- **Often a pulsive noise occurs during a sine sweep measurement**



Sensitivity to abrupt pulsive noises

- **After deconvolution, the pulsive sound causes untolerable artifacts in the impulse response**

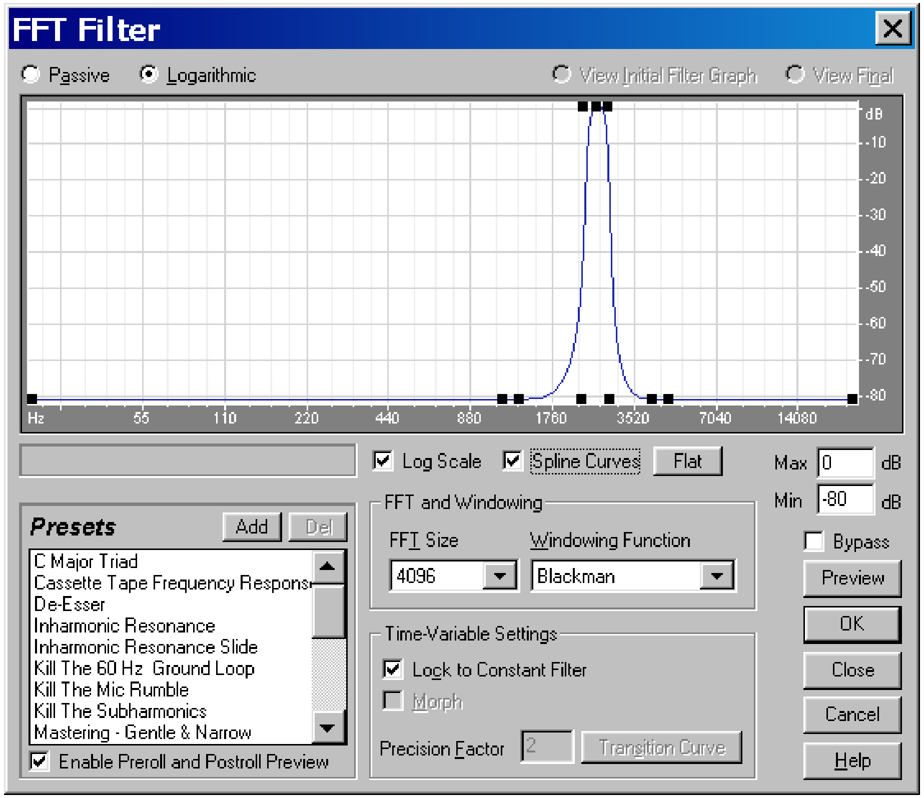


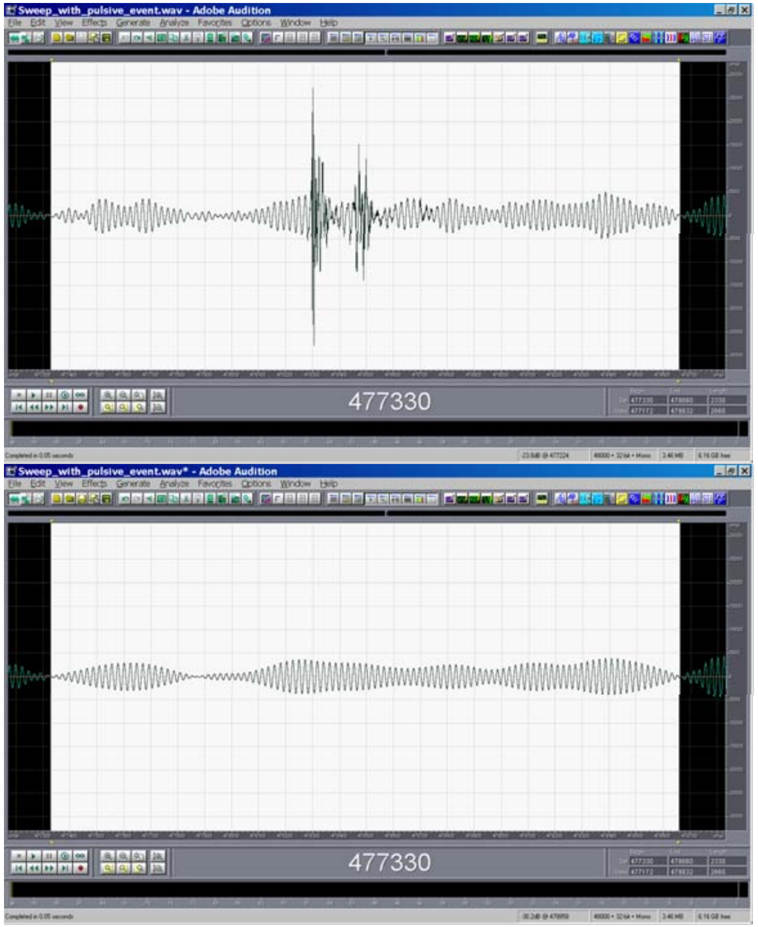


The artifact appears as a down-sloping sweep on the impulse response. At the 2 kHz octave band the decay is distorted, and the reverb. time is artificially increased from 2.13 to 2.48 s

Sensitivity to abrupt pulsive noises

- **Several denoising techniques can be employed:**
  - Brutely silencing the transient noise
  - Employing the specific “click-pop eliminator” plugin of Adobe Audition
  - Applying a narrow-passband filter around the frequency which was being generated in the moment in which the pulsive noise occurred
- **The third approach provides the better results:**





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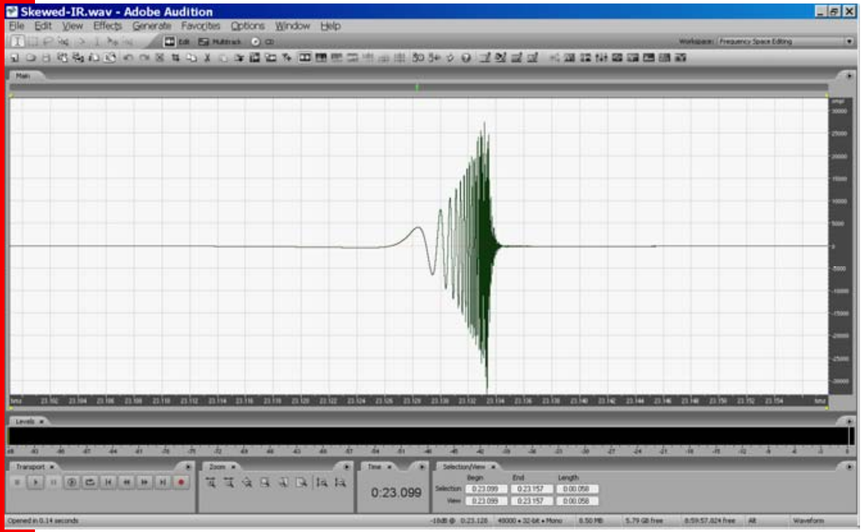
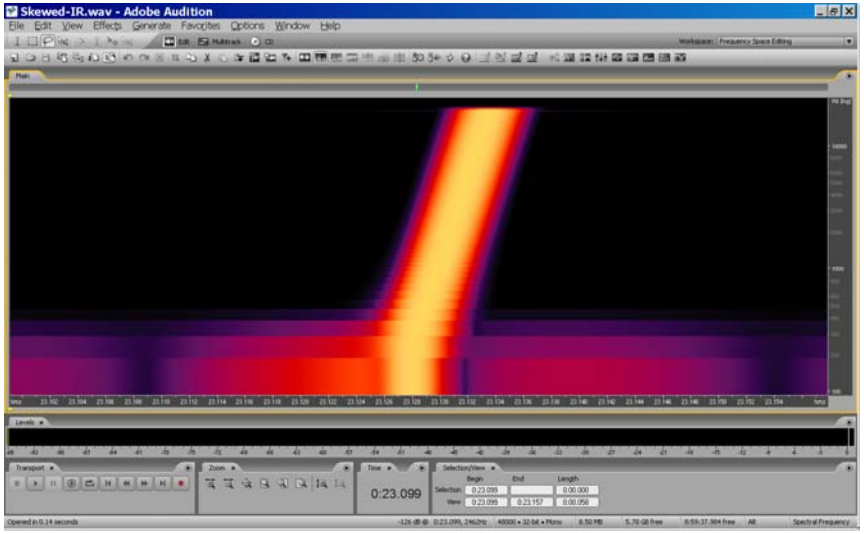
Clock mismatch

- **When the measurement is performed employing devices which exhibit signifcant clock mismatch between playback and recording, the resulting impulse response is "skewed" (stretched in time):**

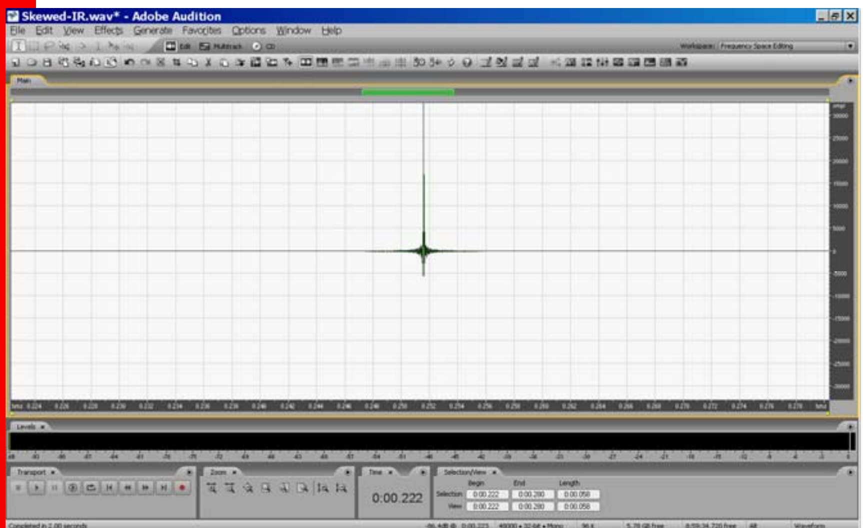


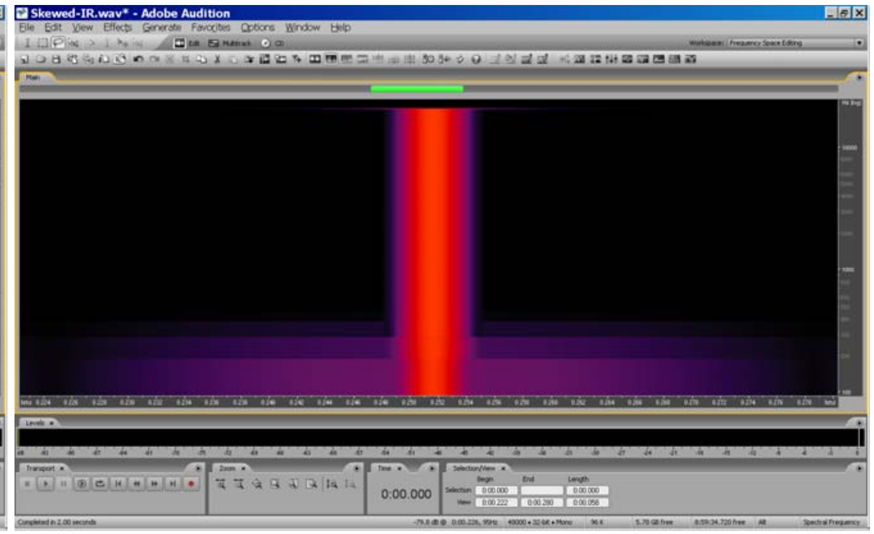


The pictures show the results of an electrical measuremnt performed connecting directly a CD-player with a DAT recorder

Clock mismatch

- **It is possible to re-pack the impulse response employing the already-described approach based on the usage of a Kirkeby inverse filter:**

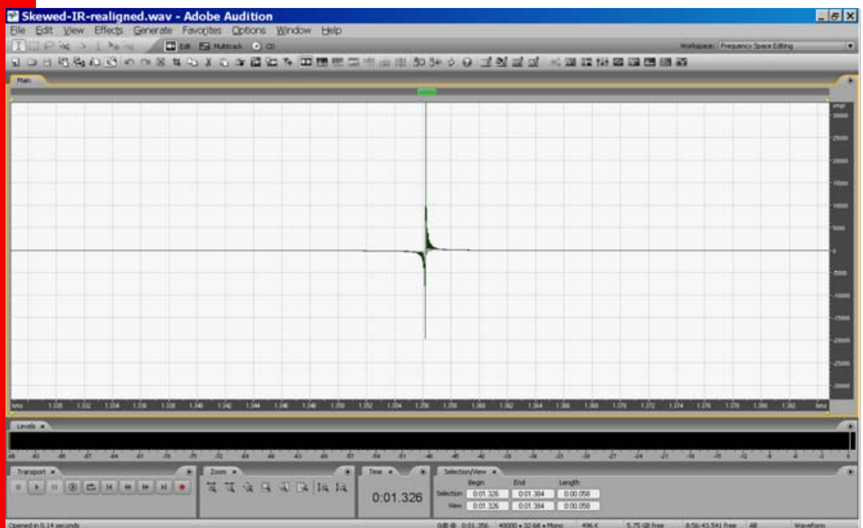


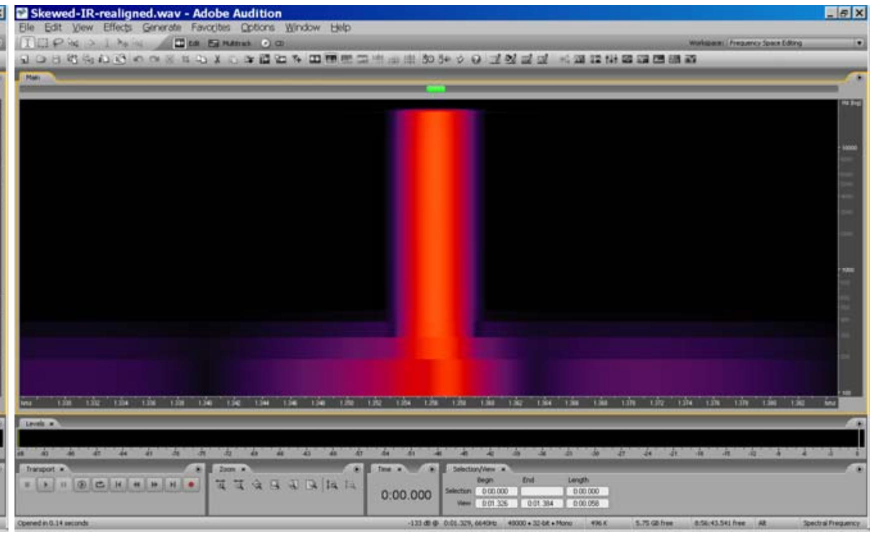


However, this is possible only if a “reference” electrical (or anechoic) measurement has been performed. But, in many cases, one only gets the re-recorded signals, and no reference measurement is available, so the Kirkeby inverse filter cannot be computed.

Clock mismatch

- **However, it is always possible to generate a pre-stretched inverse filter, which is longer or shorter than the “theoretical” one - by proper selection of the lenght of the inverse filter, it is possible to deconvolve impulse responses which are almost perfectly “unskewed”:**





The pictures show the result of the deconvolvution of a clock-mismatched measurement, in which a pre-strecthed inverse filter is employed, 8.5 ms longer than the theoretical one.

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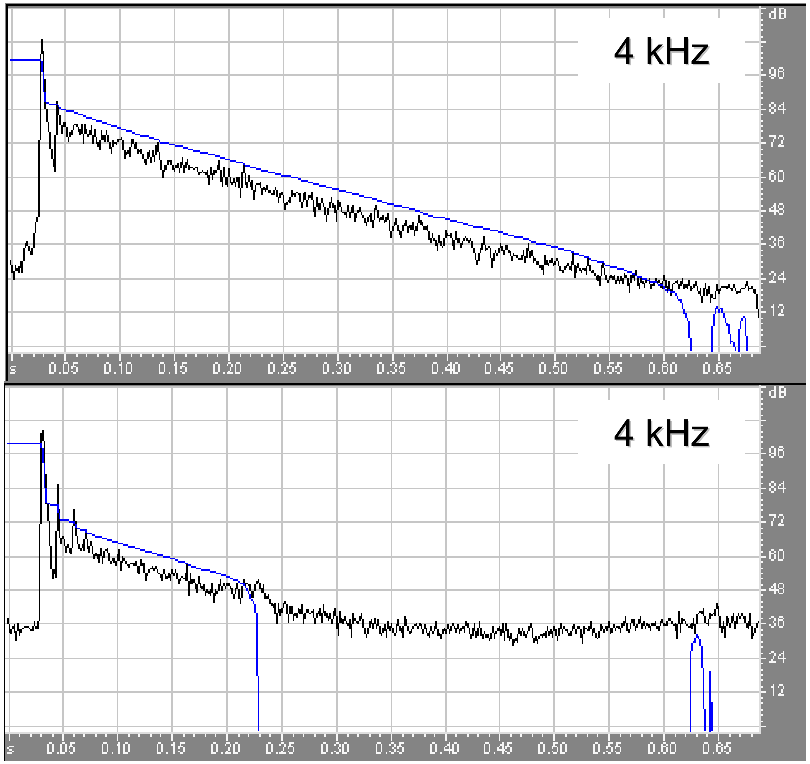
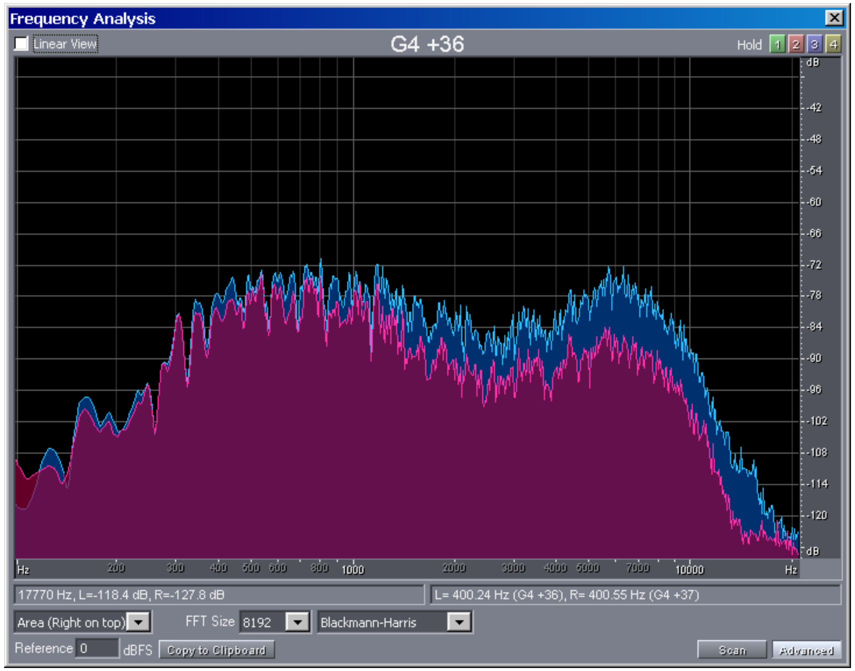
High-frequency cancellation due to averaging

- When several impulse response measurements are synchronously-averaged for improving the S/N ratio, the late part of the tail cancels out, particularly at high frequency, due to slight time variance of the system



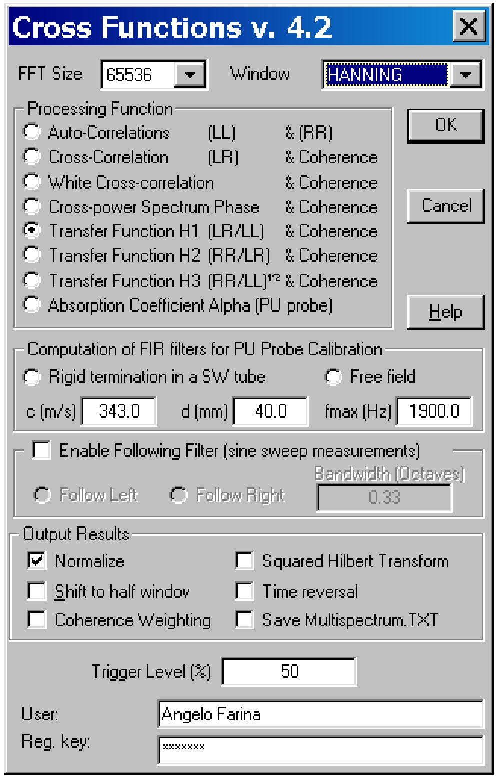


Spectrum of a single sweep of 50s (above) versus 50 sweeps of 1s (below)
short-FFT spectrum at 200 ms after direct sound

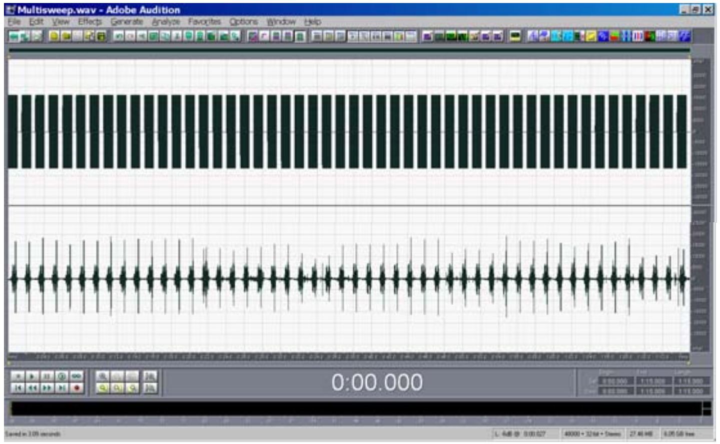
Comparison of a single sweep 50 s long with the synchronous average of 50 sweeps, 1 s long each.

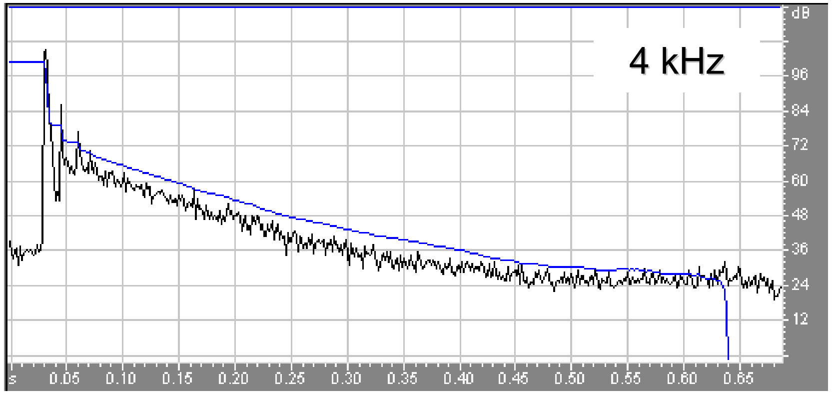
High-frequency cancellation due to averaging

- **However, if averagaing is performed properly in spectral domain, and a single conversion to time domain is performed after averaging, this artifact is significantly reduced**
- **The new “cross Functions” plugin can be used for computing H1:**



$$H_1(f) = \frac{\overline{G_{LR}}}{\overline{\overline{G_{LL}}}}$$





Result of transfer function H1, processing a sequence of 50 sine sweeps (above)

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Directivity of transducers

- Analysis of performances of binaural dummy heads
- Analysis of performances of omni / figure-of-8 microphone assemblies
- Polar patterns of dodechaedron loudspeakers

Spatial analysis by directive microphones

- **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:**

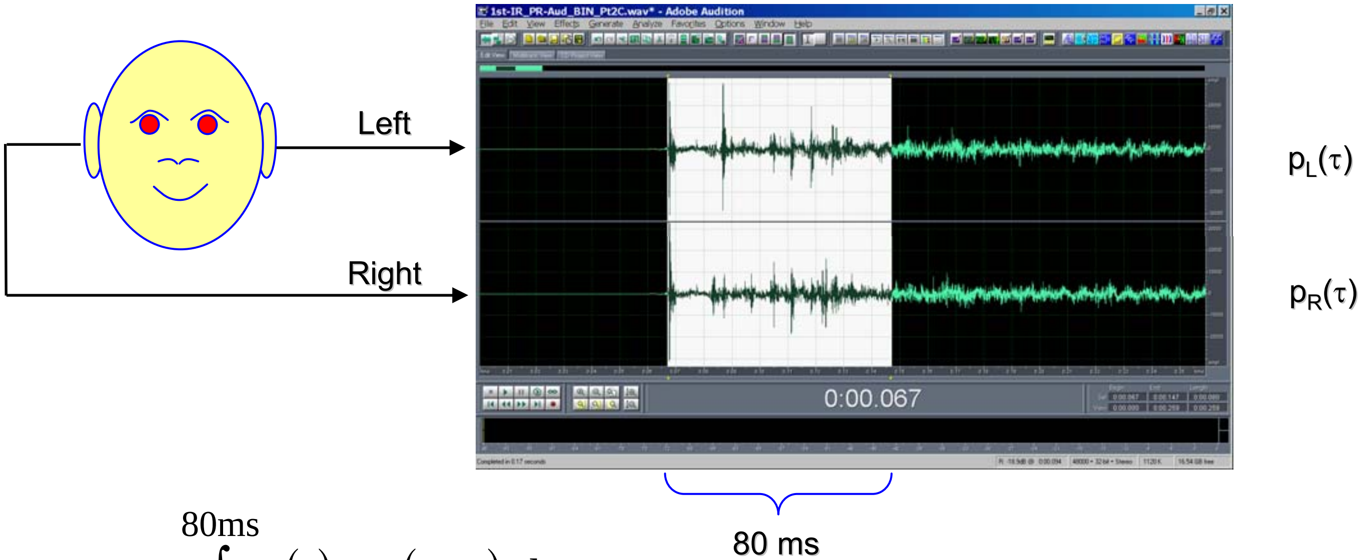
Binaural microphone (left)

and

variable-directivity microphone (right)

“objective” spatial parameters

- 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

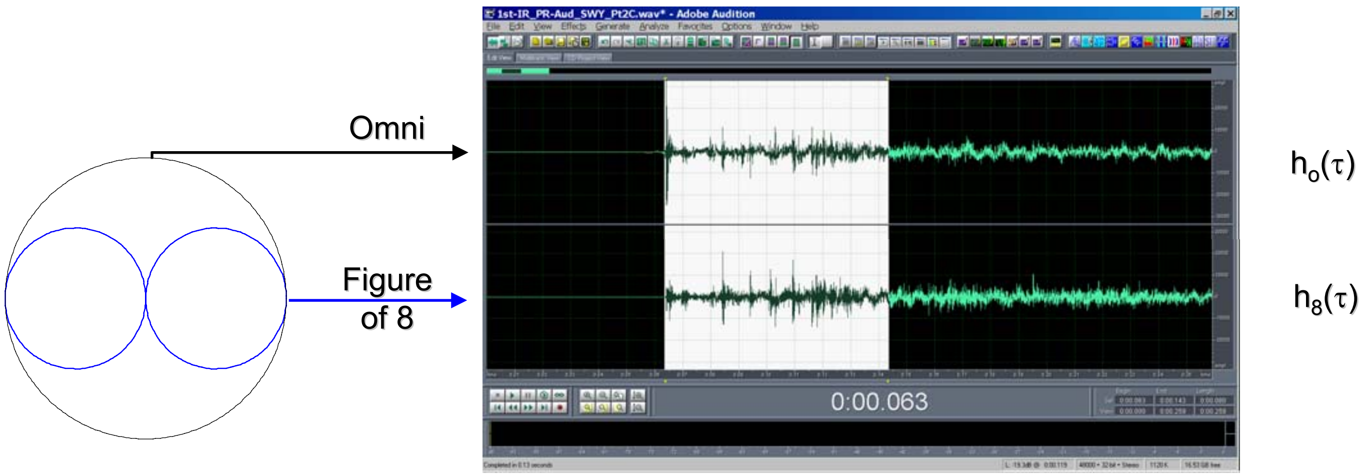


$$\rho(t) = \frac{\int_0^{80ms} p_L(\tau) \cdot p_R(\tau + t) \cdot d\tau}{\sqrt{\int_0^{80ms} p_L^2(\tau) \cdot d\tau \cdot \int_0^{80ms} p_R^2(\tau + t) \cdot d\tau}}$$

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

“objective” spatial parameters

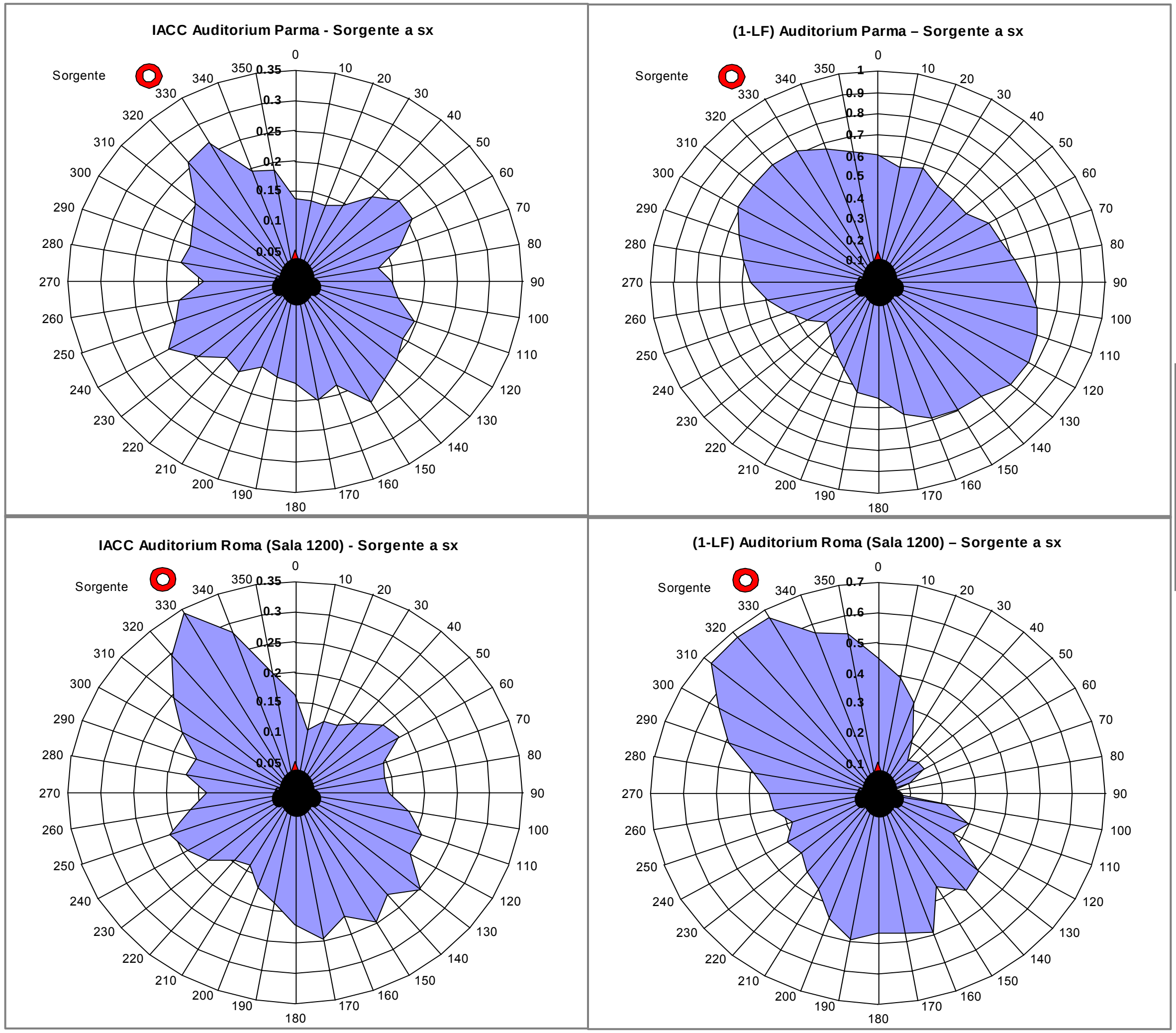
- Other “spatial” parameters are the Lateral Energy ratios: LE, LF, LFC
- These are defined from a 2-channels impulse response, the first channel is a standard omni microphone, the second channel is a “figure-of-eight” microphone:



$$LE = \frac{\int_{25ms}^{80ms} h_8^2(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_o^2(\tau) \cdot d\tau} \qquad LF = \frac{\int_{5ms}^{80ms} h_8^2(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_o^2(\tau) \cdot d\tau} \qquad LFC = \frac{\int_{5ms}^{80ms} h_8(\tau) \cdot h_o(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_o^2(\tau) \cdot d\tau}$$

Robustness of spatial parameters

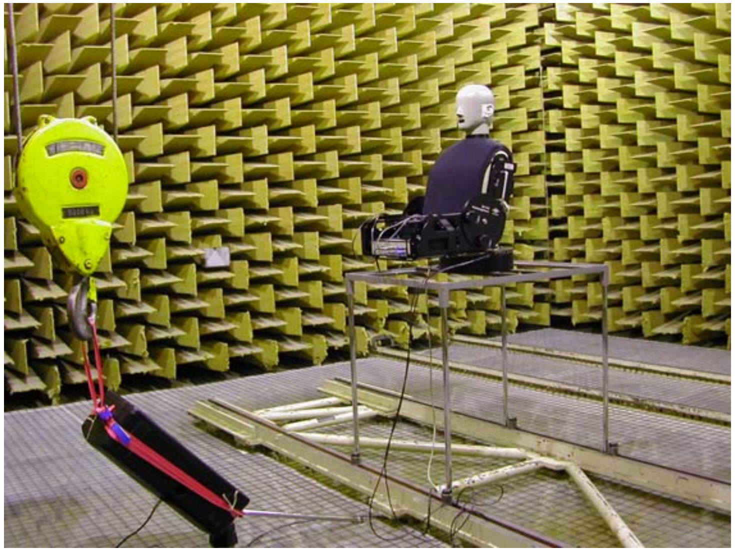
- **Both IACC and LF depend strongly on the orientation of the microphones**
- **Binaural and pressure-velocity measurements were performed in 2 theatres employing a rotating table for turning the microphones**

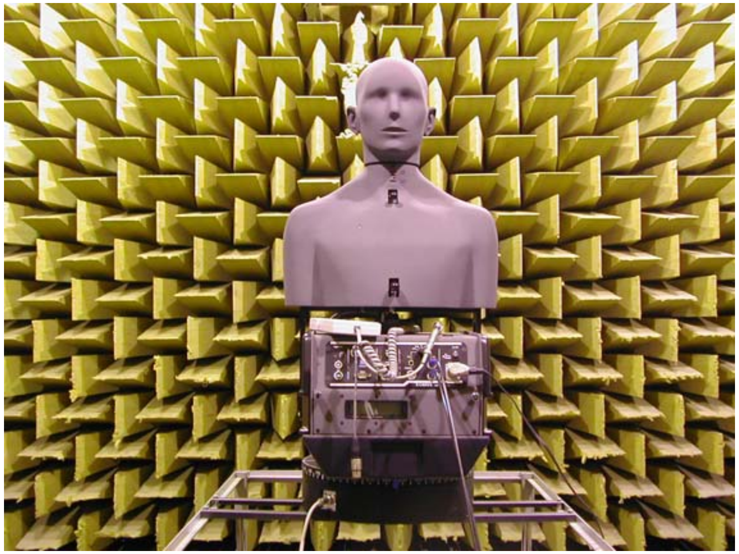


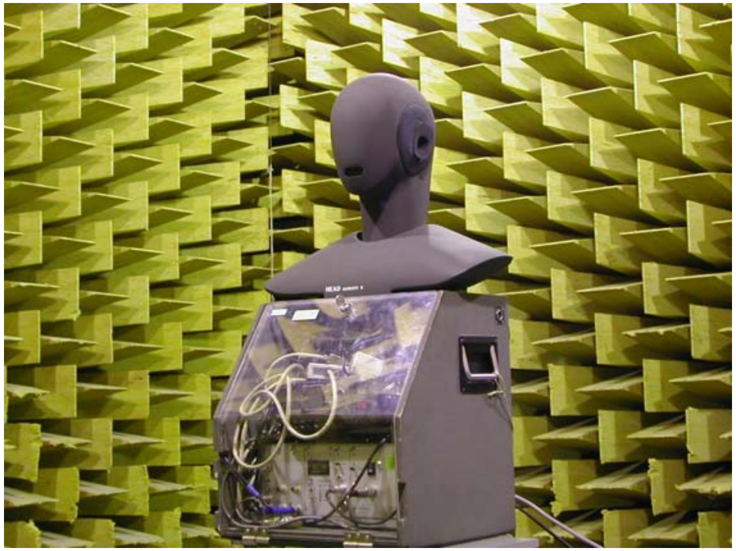
Theatre	1-LF	IACC
Parma	0.725	0.266
Roma	0.676	0.344

Are binaural measurents reproducible?

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

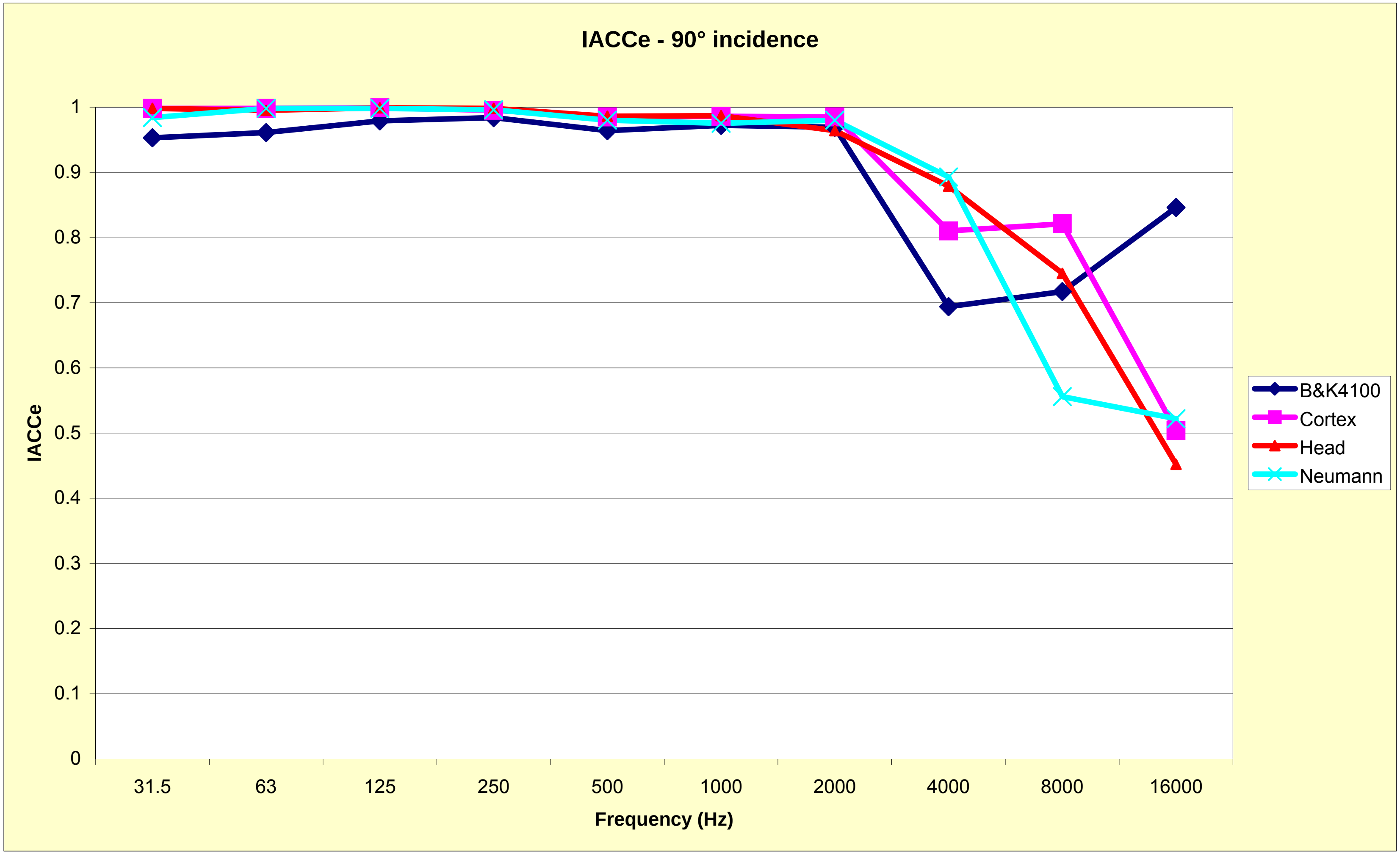






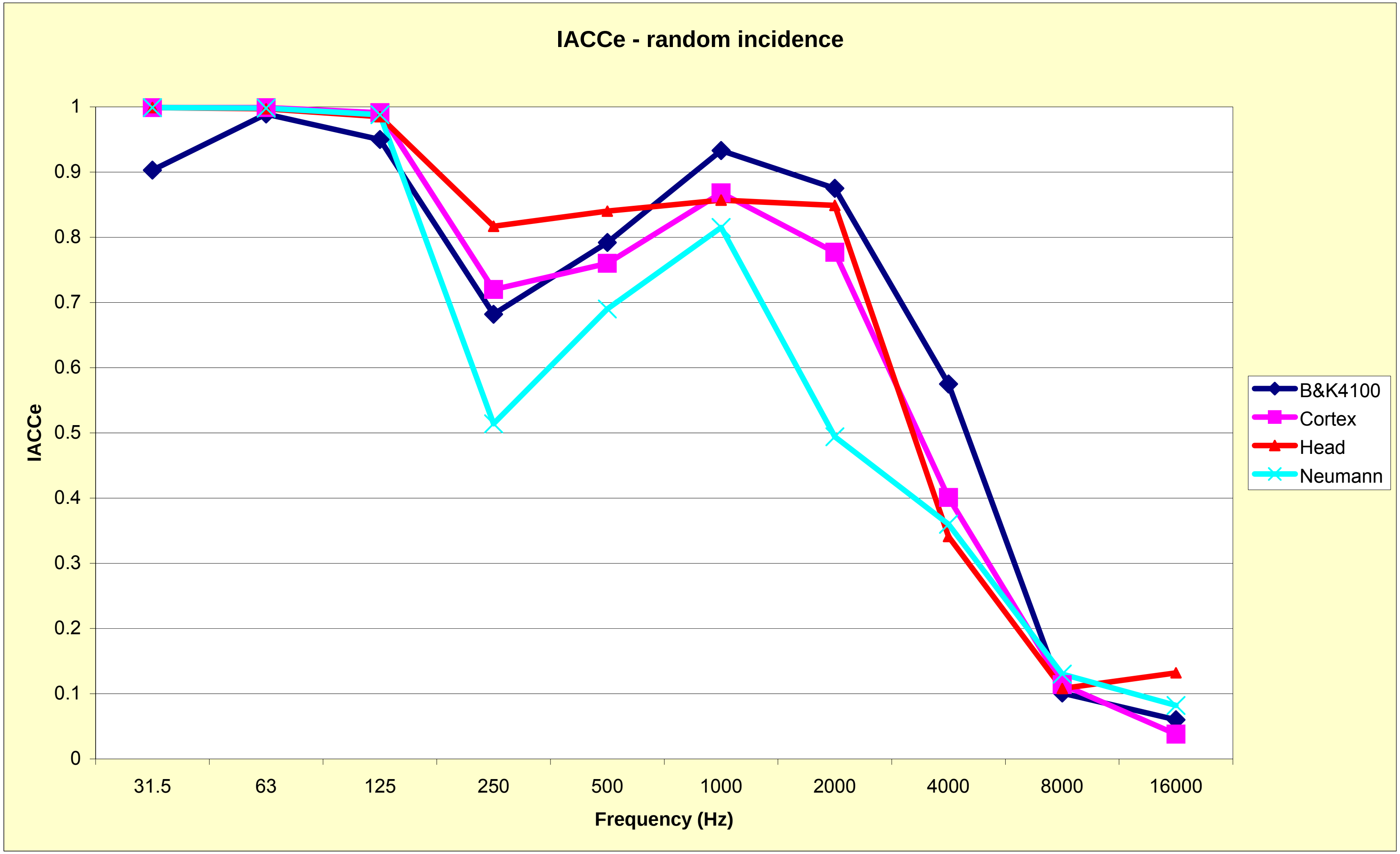
Are binaural measurents reproducible?

- **90° incidence - at low frequency IACC is almost 1, at high frequency the difference between the heads becomes evident**



Are binaural measurents reproducible?

- Diffuse field - the difference between the heads is now dramatic

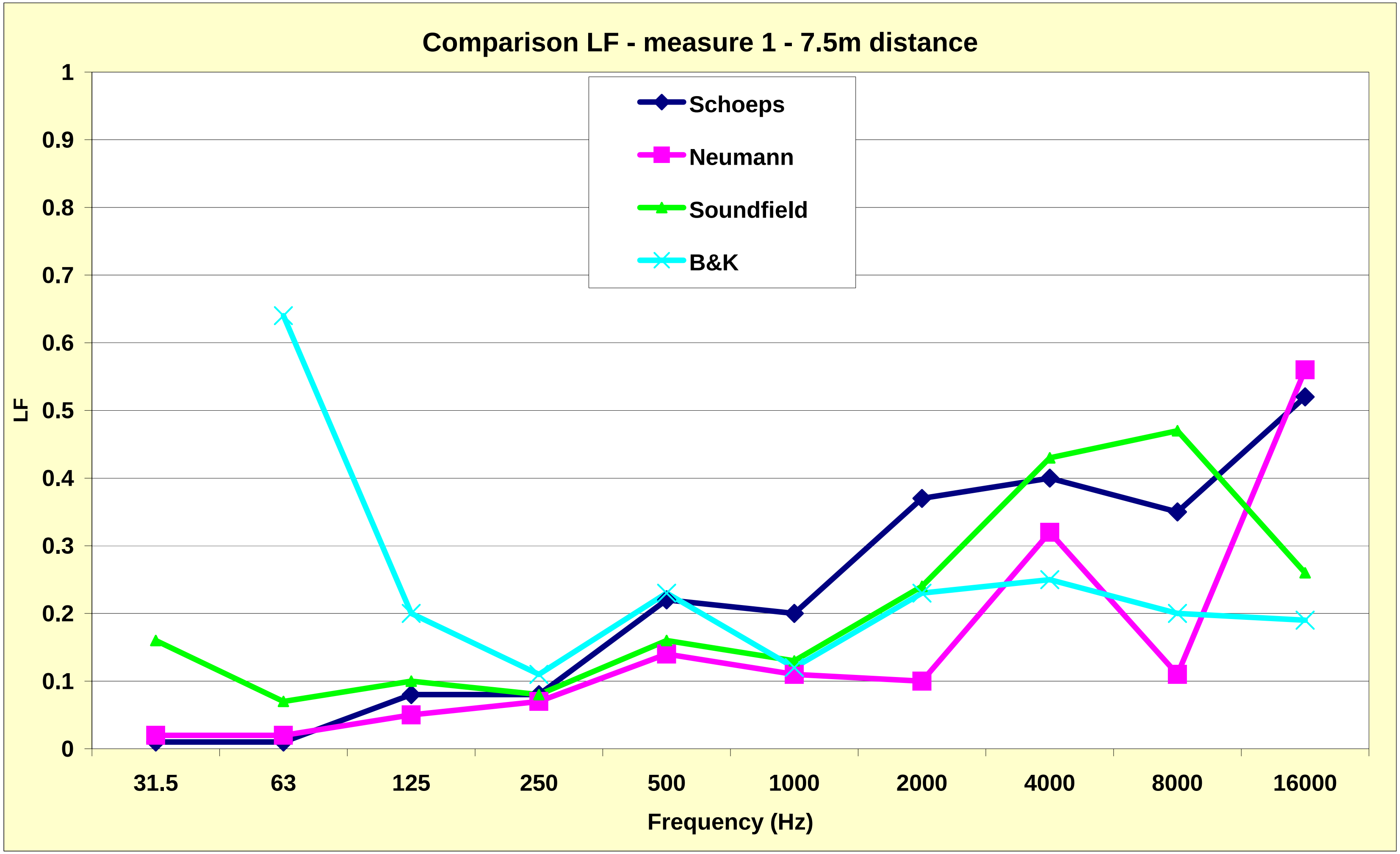


Are LF measurents reproducible?

- **Experiment performed in the Auditorium of Parma - same loudspeaker, same source and receiver positions, 5 pressure-velocity microphones**

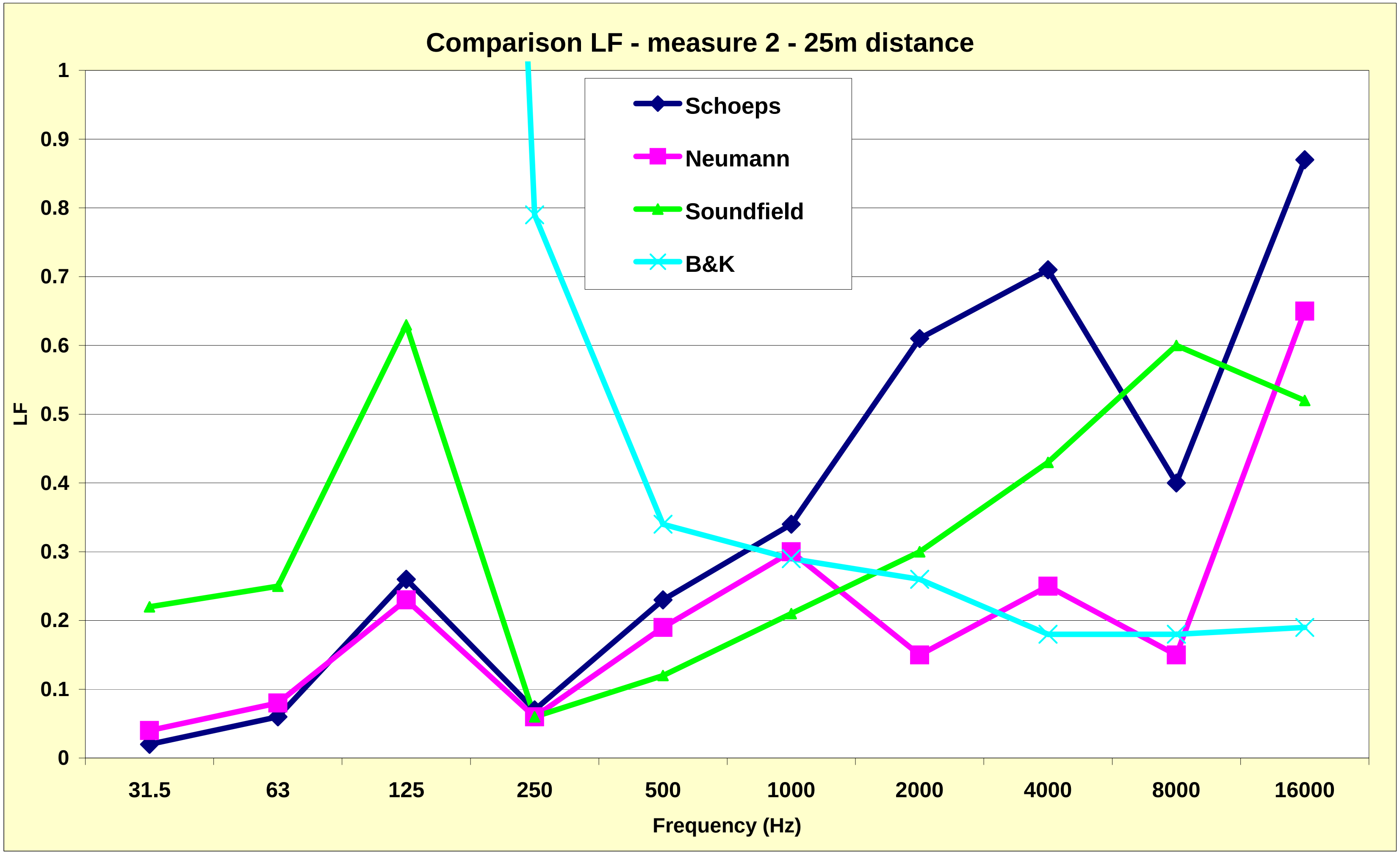
Are LF measurents reproducible?

- **At 7.5 m distance, the results already exhibit significant scatter**



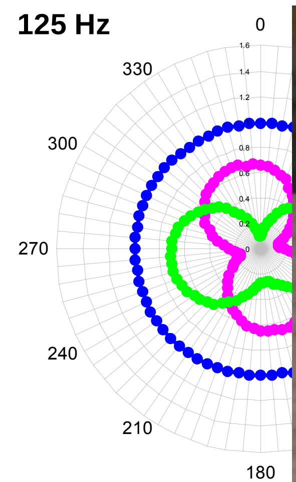
Are LF measurents reproducible?

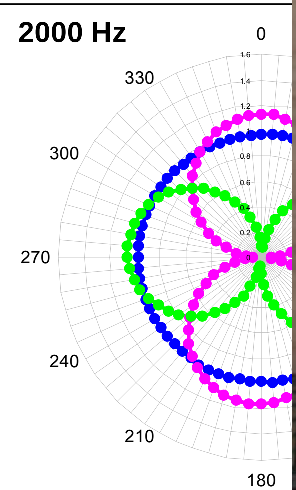
- **At 25 m distance, the scatter is even larger....**

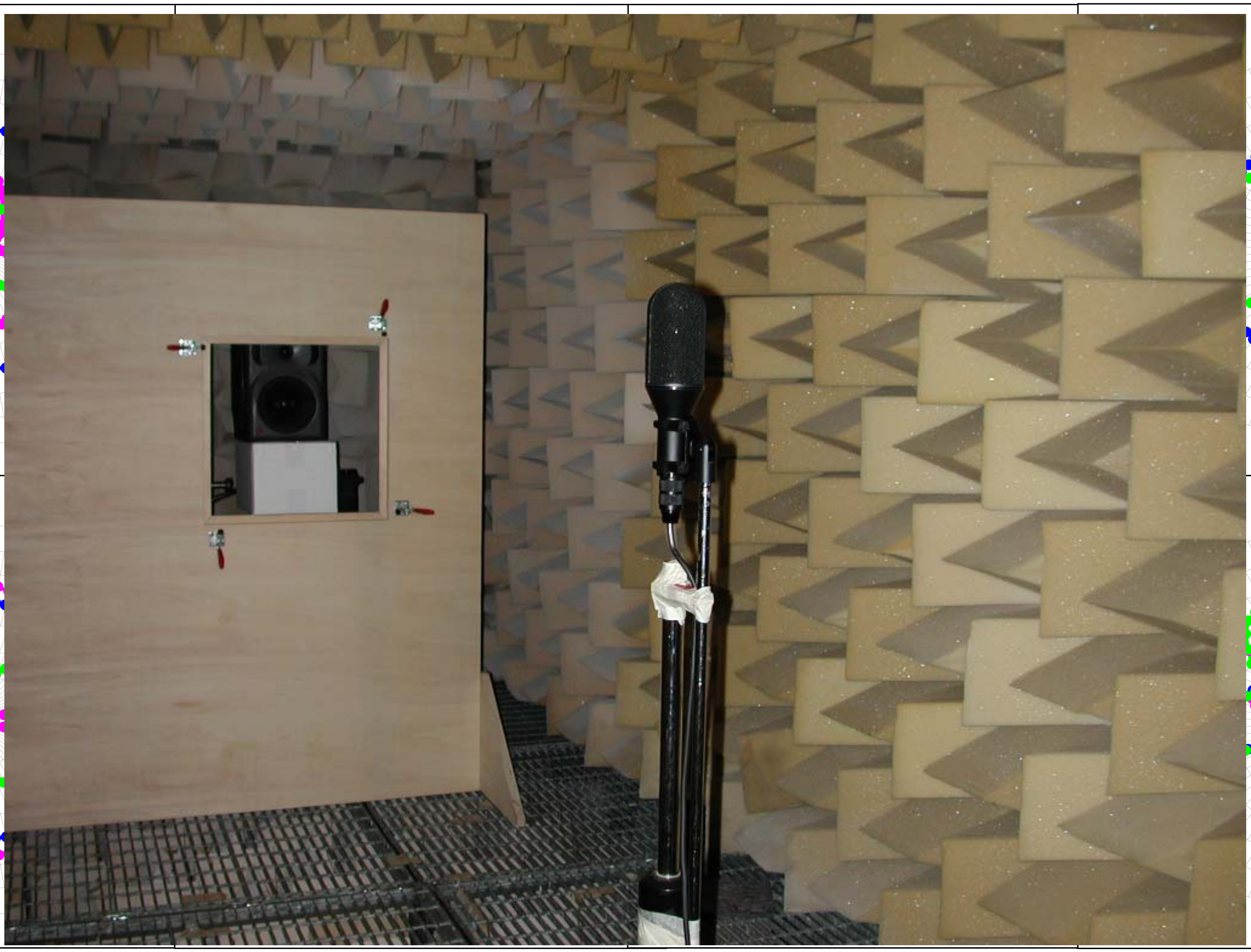


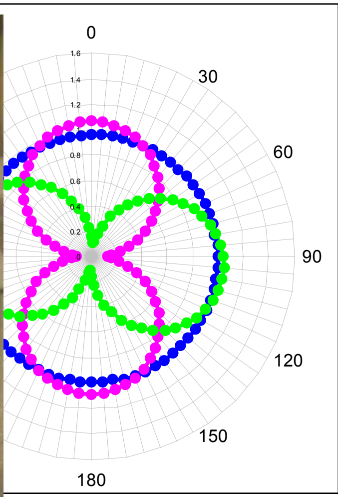
Directivity of transducers

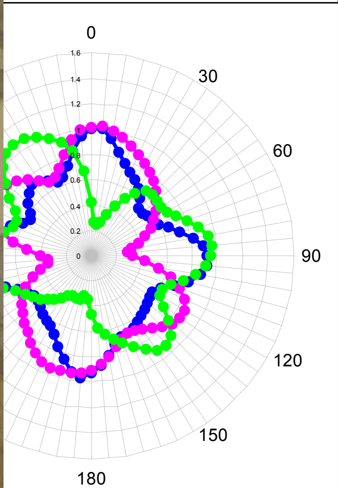
Soundfield ST-250 microphone





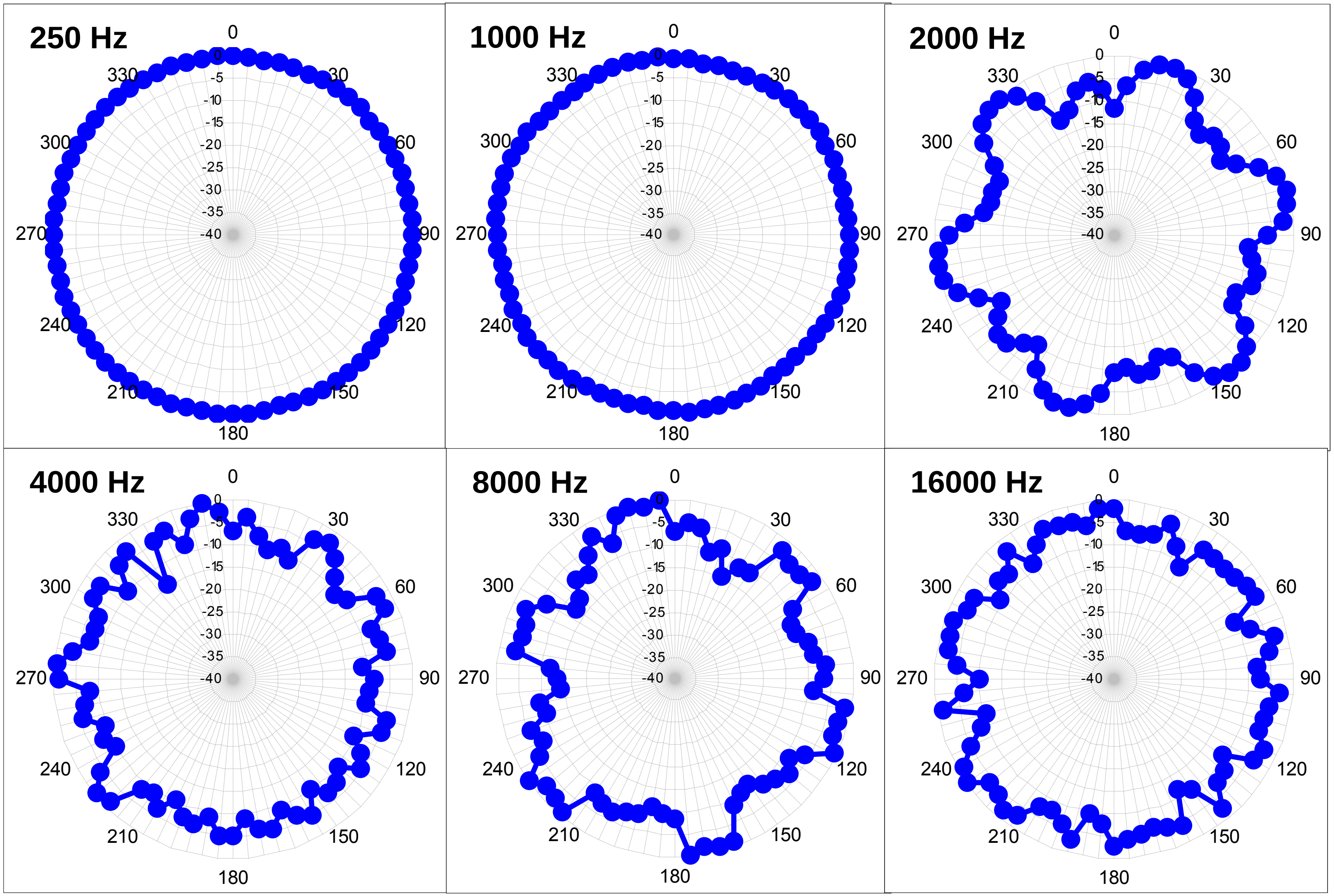






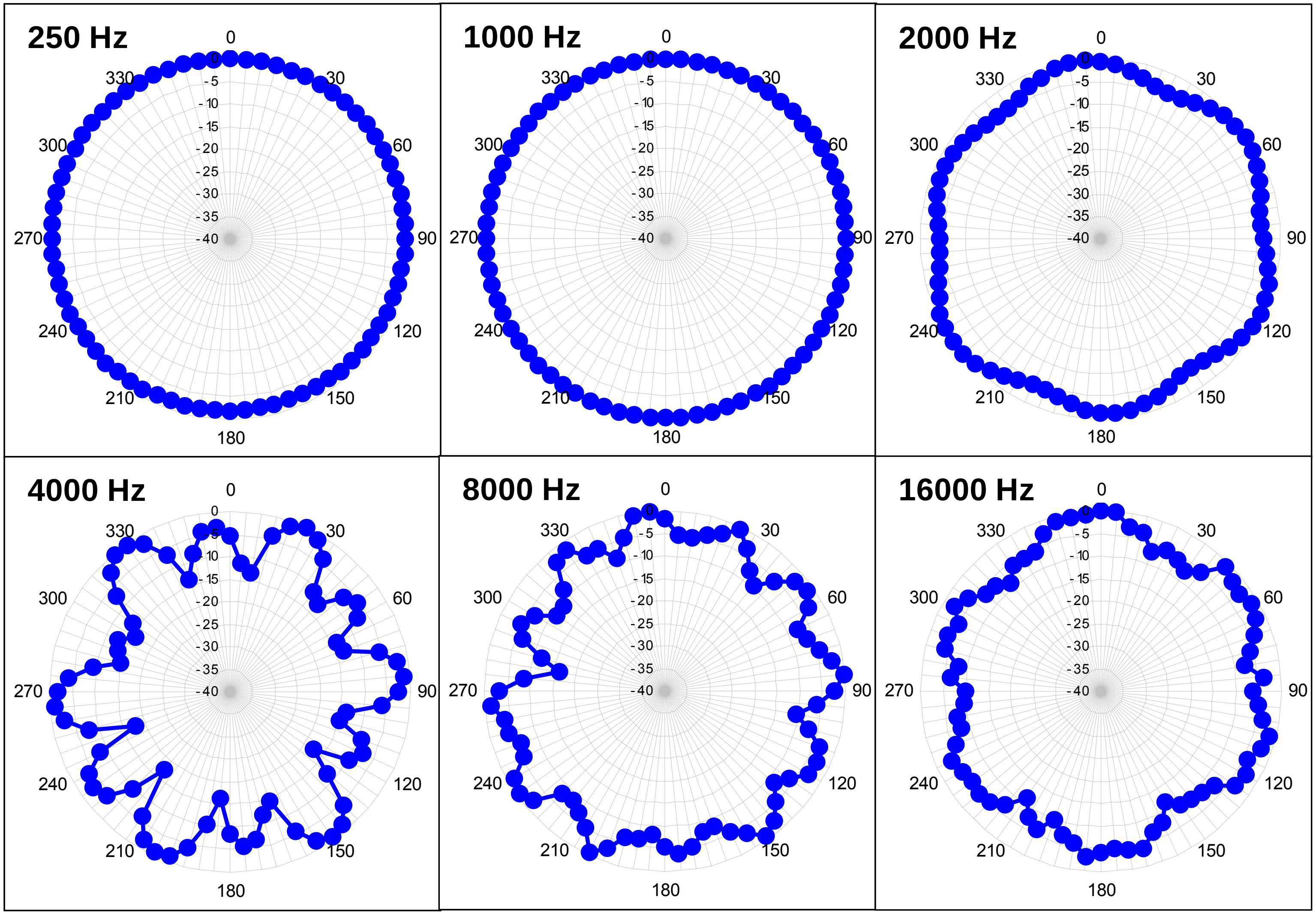
Directivity of transducers

LookLine D-300 dodechaedron



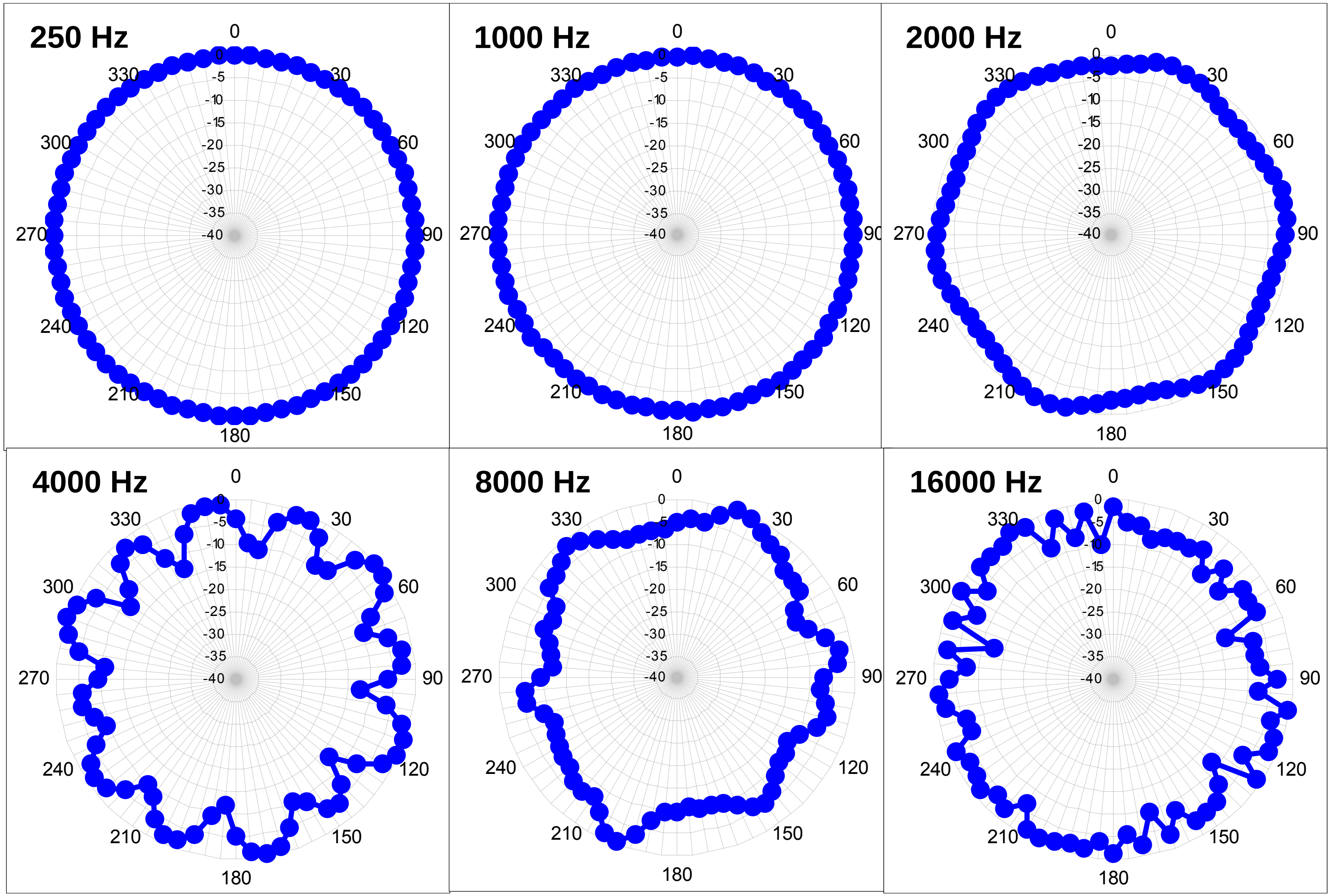
Directivity of transducers

LookLine D-200 dodechaedron



Directivity of transducers

Omnisonic 1000 dodechaedron



Conclusions

- ESS is now employed in top-grade measurement systems, including Audio Precision (TM), Rhode-Schwartz and Bruel & Kjaer's DIRAC software
- However, these completely-packaged measurement systems often do not allow to play "tricks" and to adjust the signals for solving problems, which have been shown here
- Workarounds have been found for almost all the problems occurring when performing ESS measurements
- These workarounds are easily applied by working with a general purpose sound editor (Adobe Audition)
- A number of additional plugins have been developed, making easy to generate the test signal, to deconvolve and process impulse responses, to compute inverse filters and to perform advanced processing (STI, AQT, etc.)
- These plugins are freely downloadable at the AURORA web site:

www.aurora-plugins.com

- The only remaining problems are related to existing transducers (microphones and loudspeakers), as their directivity is far from the theoretical one