

# RECORDING CONCERT HALL ACOUSTICS FOR POSTERITY



Angelo Farina <sup>(1)</sup> – Regev Ayalon <sup>(2)</sup>

<sup>(1)</sup> Dipartimento di Ingegneria Industriale, Università di Parma, Via delle Scienze 181/A  
Parma, 43100 ITALIA

[HTTP://pcfarina.eng.unipr.it](http://pcfarina.eng.unipr.it) - mail: [farina@unipr.it](mailto:farina@unipr.it)

<sup>(2)</sup> K.S. Waves Inc., Azrieli Center, Tel Aviv, ISRAEL

[HTTP://www.waves.com](http://www.waves.com) - mail: [regev@waves.com](mailto:regev@waves.com)



*Multichannel Audio - The New Reality*

*24th AES International Conference*

*June 26 - 28, 2003*

# Background

- The title of this paper is exactly the same employed by Michael Gerzon in its JAES paper (Vol. 23, Number 7, 1975)
- He first proposed to collect impulse responses measured in famous theatres, with a microphone capable of capturing the complete spatial information
- This paper is consequently basically a tribute to M.Gerzon, who had foreseen most of the modern multichannel audio applications, including impulse response measurements and auralization obtained by convolution.

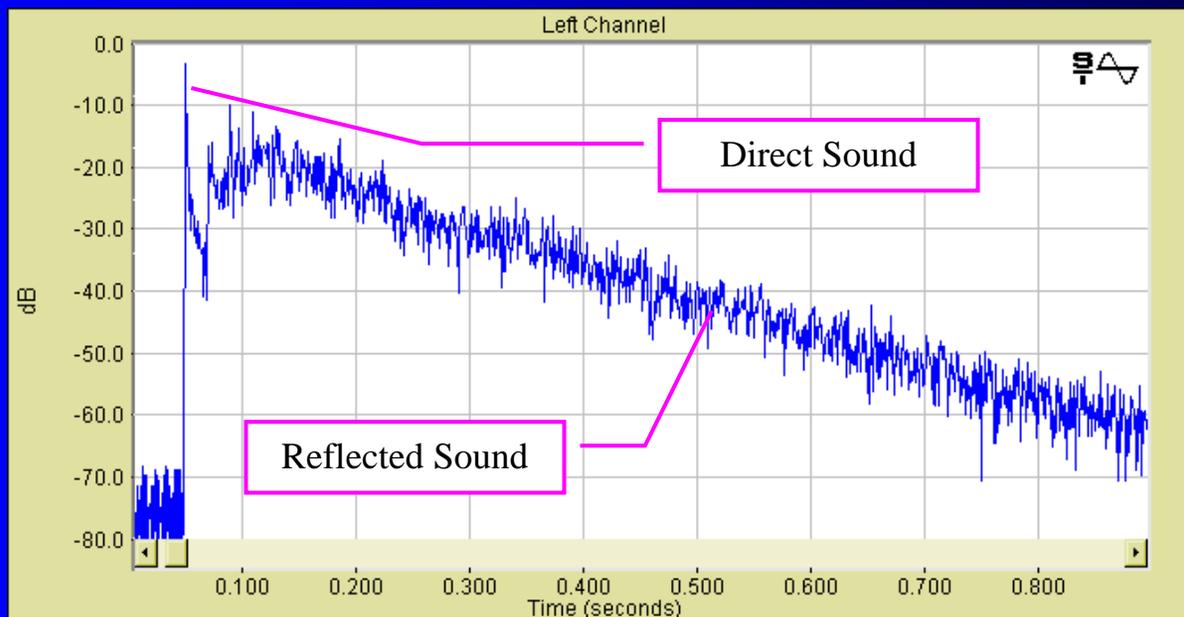
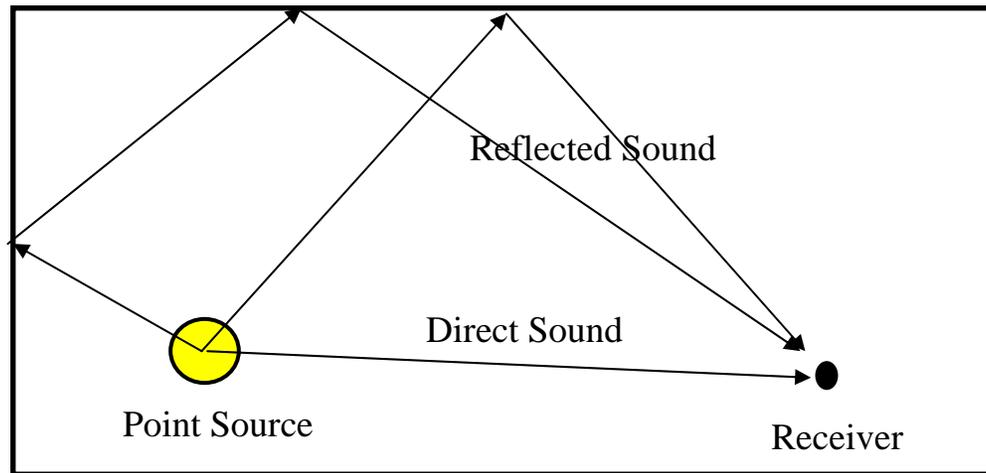
# Goals

- The main goal is to measure an huge collection of impulse response in famous theatres, concert halls, cathedrals, etc.
- These impulse responses have two main uses:
  1. In case something happens to the original space (remember the case of La Fenice theater) they contain a detailed “acoustical photography” which is preserved for the posterity
  2. They can be used for studio sound processing, as artificial reverb and surround filters for today’s and tomorrow’s musical productions

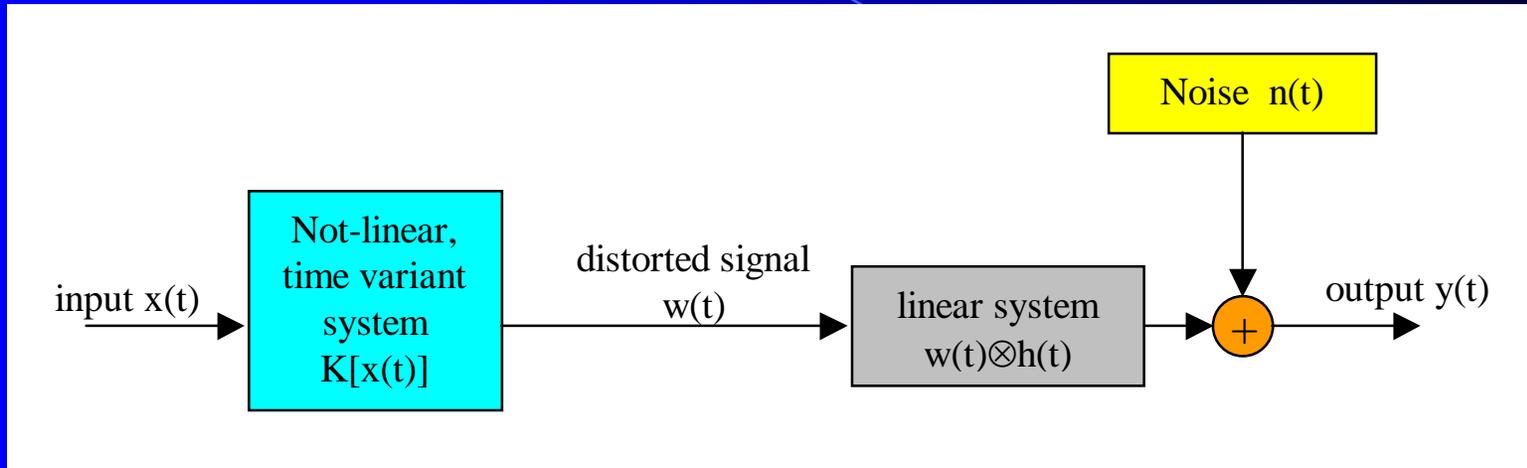
# Topics

- Description of the measurement technique
- Analysis of some acoustical parameters of the first theaters already measured
- Description of the processing methods to be employed for transforming the measured data in audible reconstructions of the original spaces
- Description of the usage of the measured data for studio processing and production

# Sound propagation in rooms



# Measurement process



- The desired 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  $n(t)$ .

# Test signal: Log Sine Sweep

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

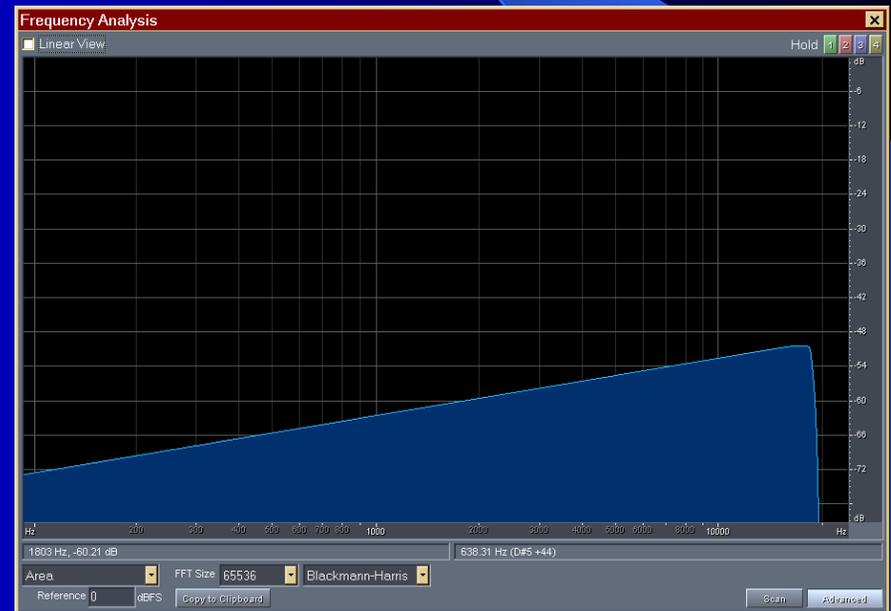
$$x(t) = \sin \left[ \frac{2 \cdot \pi \cdot f_1 \cdot T}{\ln \left( \frac{f_2}{f_1} \right)} \cdot \left( e^{\frac{t}{T} \cdot \ln \left( \frac{f_2}{f_1} \right)} - 1 \right) \right]$$

# 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

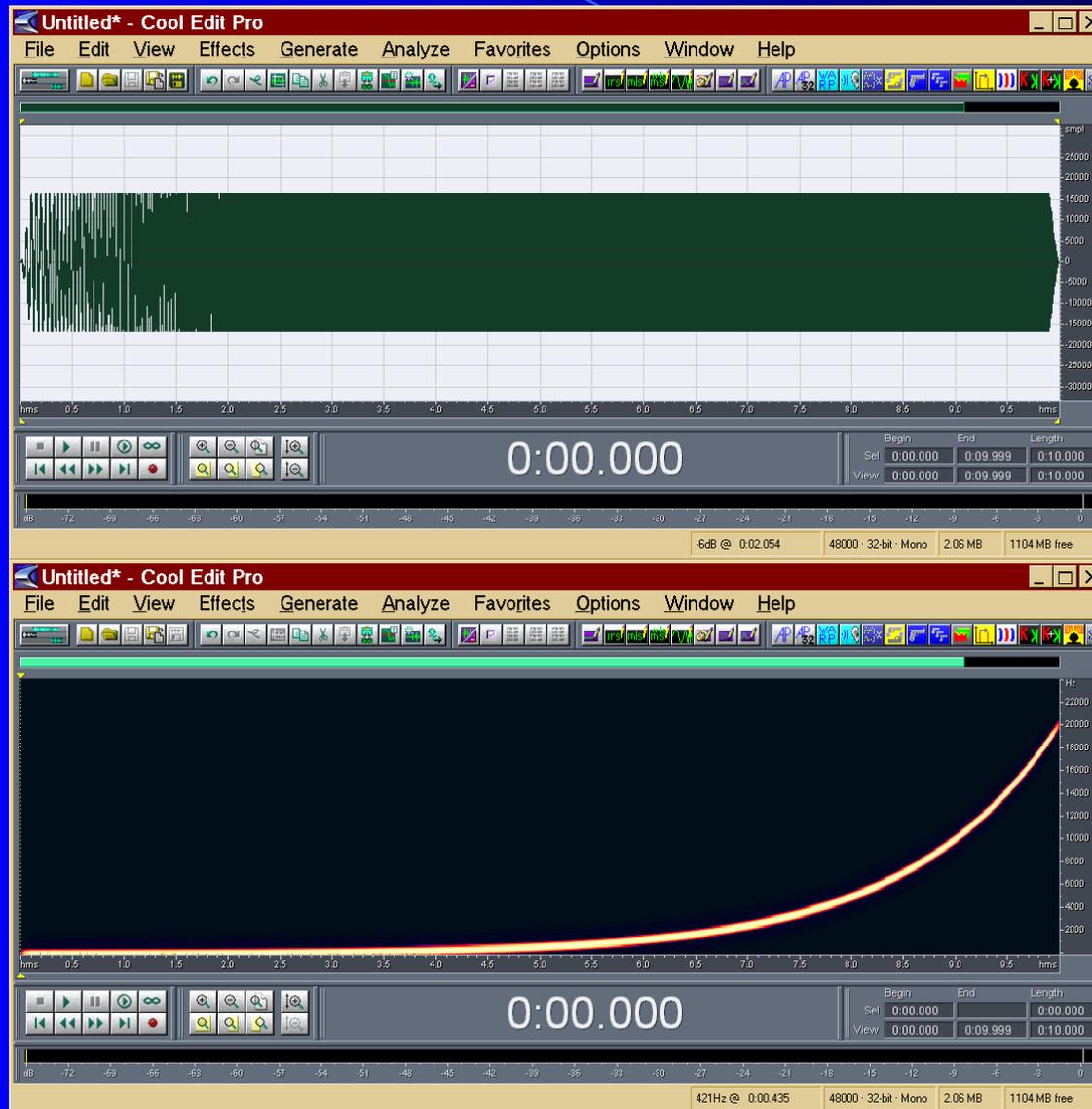


Test Signal  $x(t)$

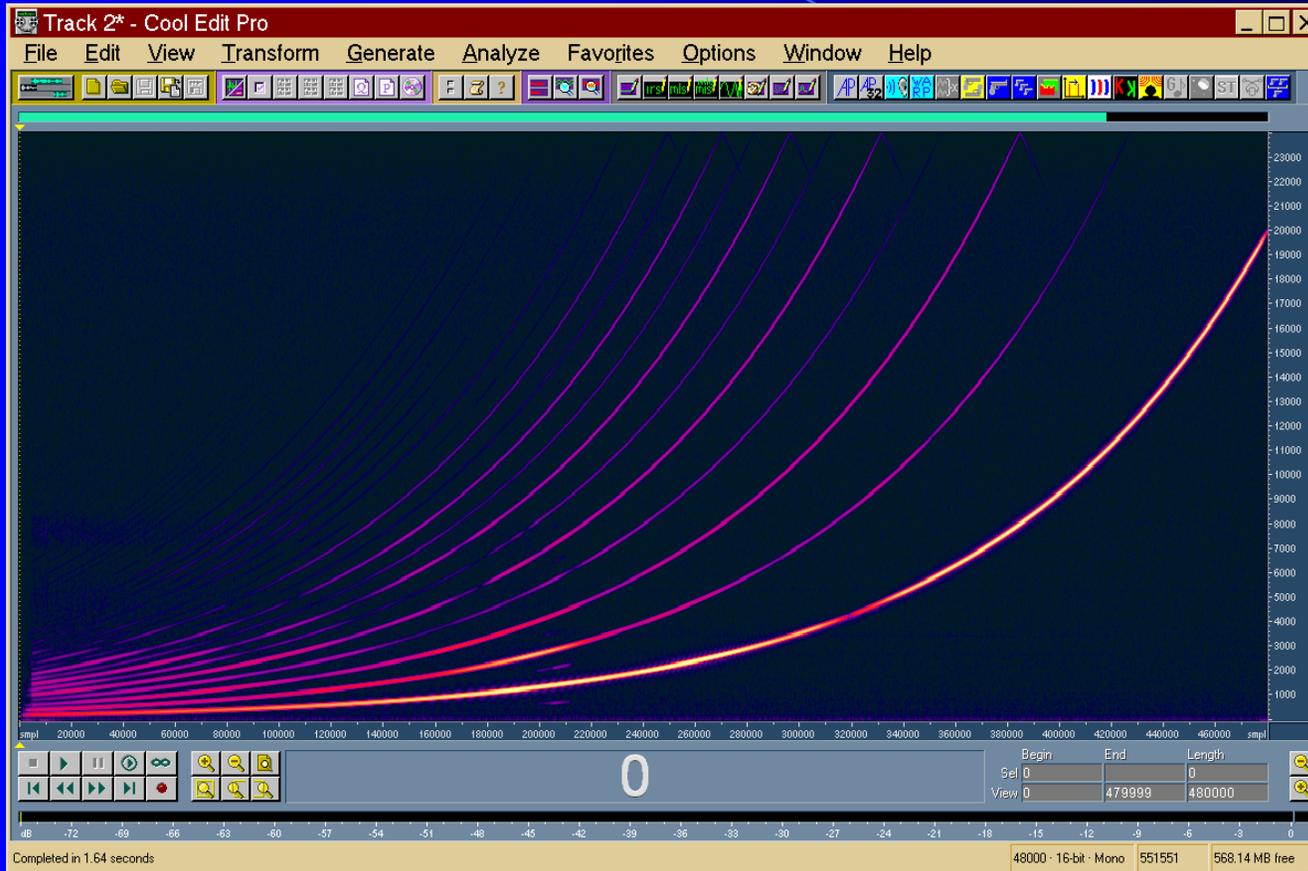


Inverse Filter  $z(t)$

# Test Signal – $x(t)$

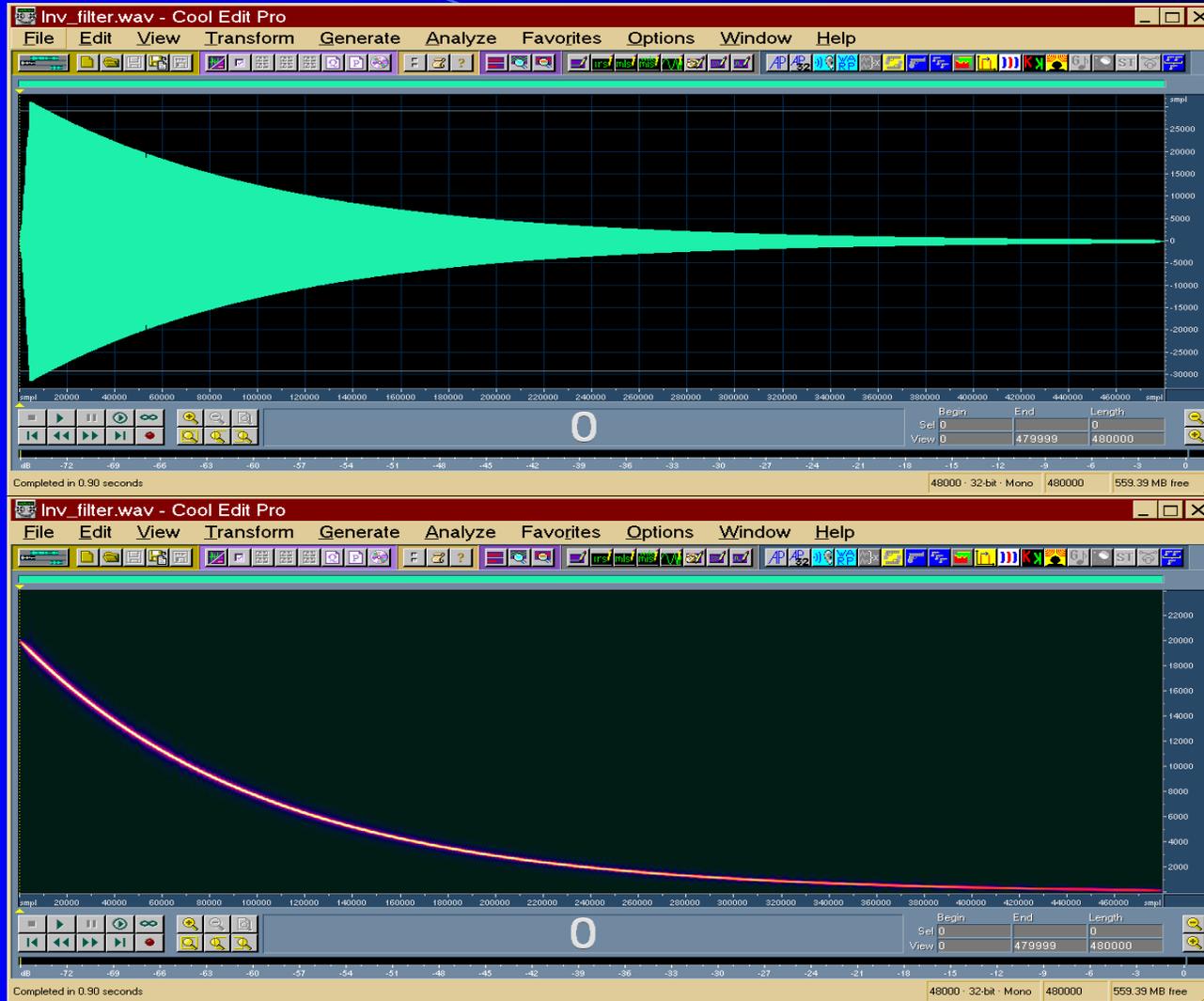


# Measured signal - $y(t)$



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

# Inverse Filter – $z(t)$



The deconvolution of the system's impulse response is obtained convolving the measured signal  $y(t)$  with the inverse filter  $z(t)$  [equalized, time-reversed  $x(t)$ ]

*24th AES International Conference*

# Result of the deconvolution



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

# Measurement Setup

- The measurement method incorporates all the known techniques:
  - Binaural
  - B-format (1<sup>st</sup> order Ambisonics)
  - WFS (Wave Field Synthesis, circular array)
  - ITU 5.1 surround (Williams MMA, OCT, INA, etc.)
  - Binaural Room Scanning
  - M. Poletti high-order virtual microphones
- This measurement setup has been named “Waves2003”, as it is being employed for the collection of impulse response to be employed together with the new convolution software being developed by KS Waves ltd.

# “Waves2003” Measurement Parameters

- Test Signal: pre-equalized sweep

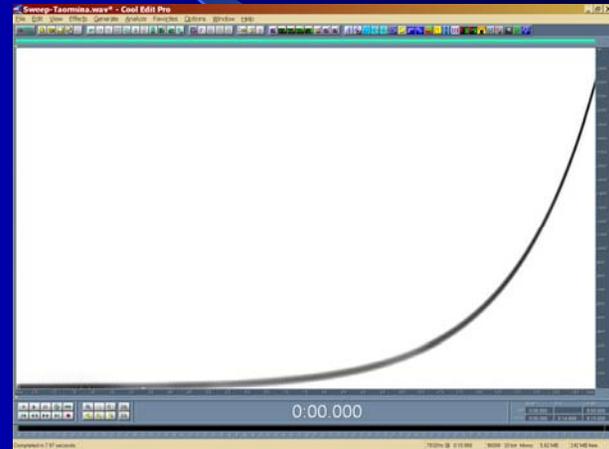
Start Frequency                      22 Hz

End Frequency                        22 kHz

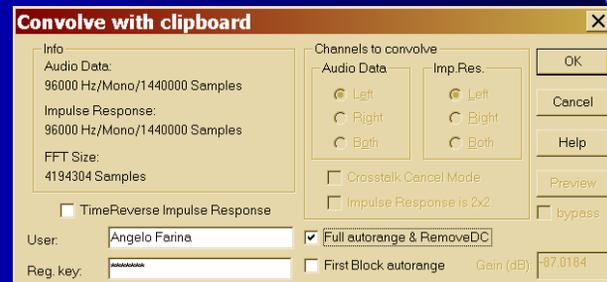
Sweep length                         15 s

Silence between sweeps            10 s

Type of sweep                        LOG

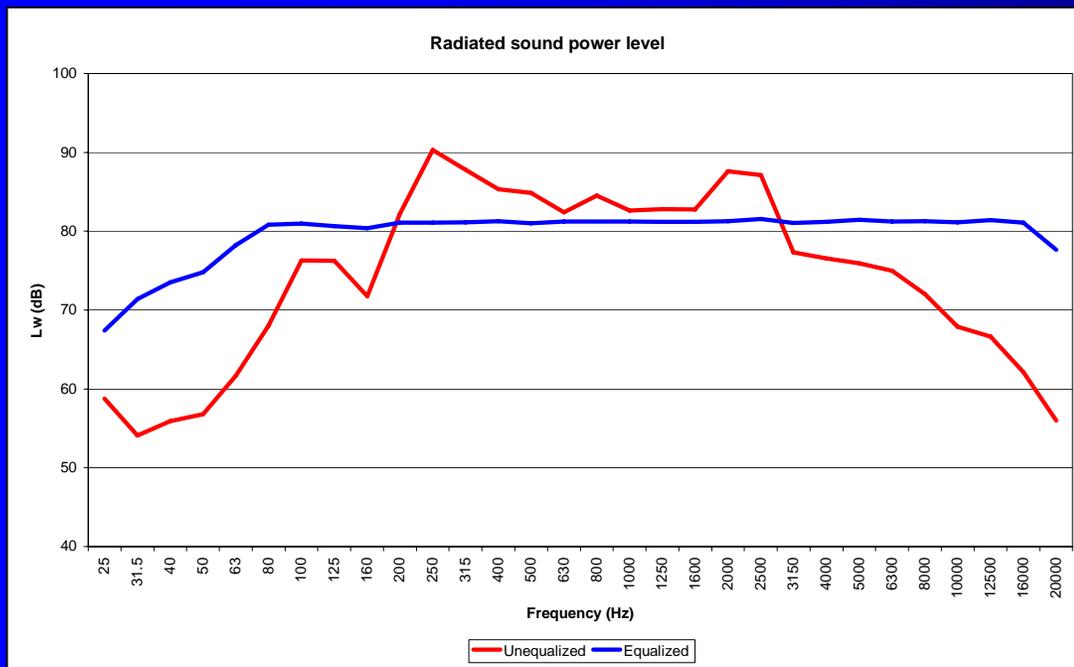


- Deconvolution:



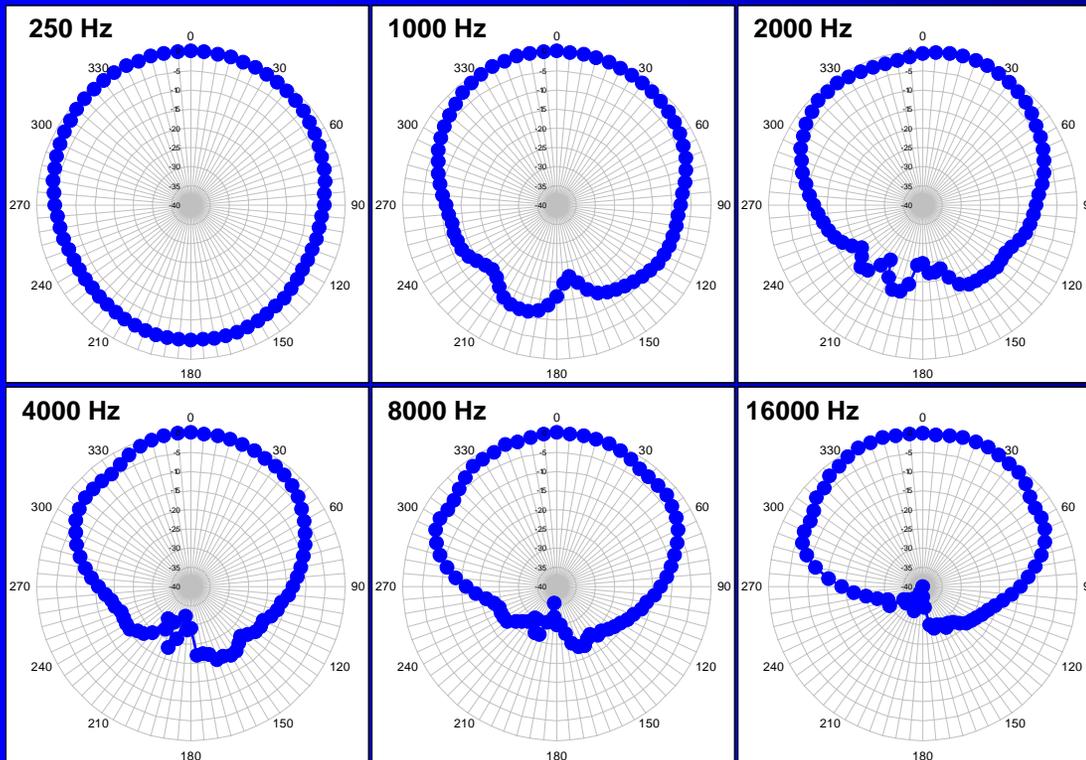
# Transducers (sound source #1)

- Equalized, omnidirectional sound source:
  - Dodechaedron for mid-high frequencies
  - Subwoofer



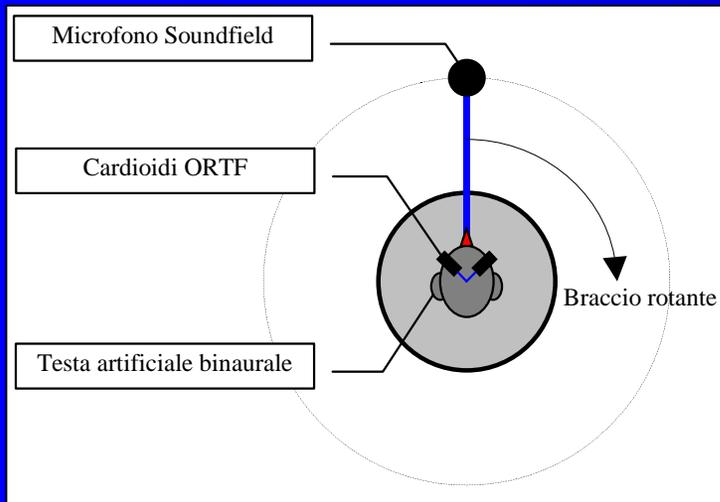
# Transducers (sound source #2)

- Genelec S30D reference studio monitor:
  - Three-ways, active multi-amped, AES/EBU
  - Frequency range 37 Hz – 44 kHz ( $\pm 3$  dB)



# Transducers (microphones)

- 3 types of microphones:
  - Binaural dummy head (Neumann KU-100)
  - 2 Cardioids in ORTF placement (Neumann K-140)
  - B-Format 4 channels (Soundfield ST-250)



# Other hardware equipment

- Rotating Table:
  - Outline ET-1

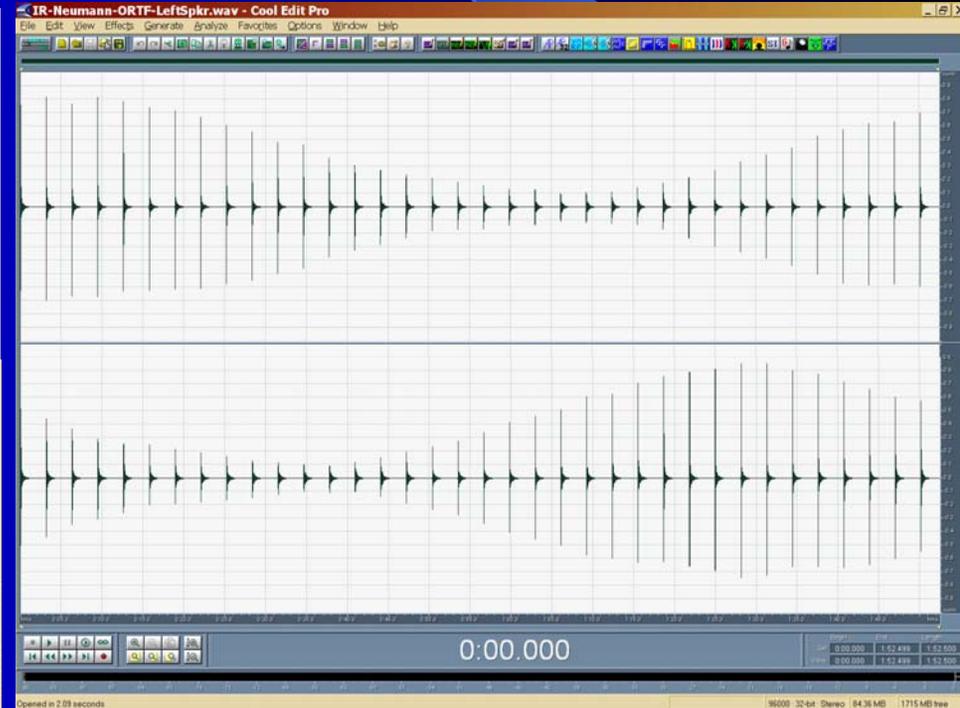
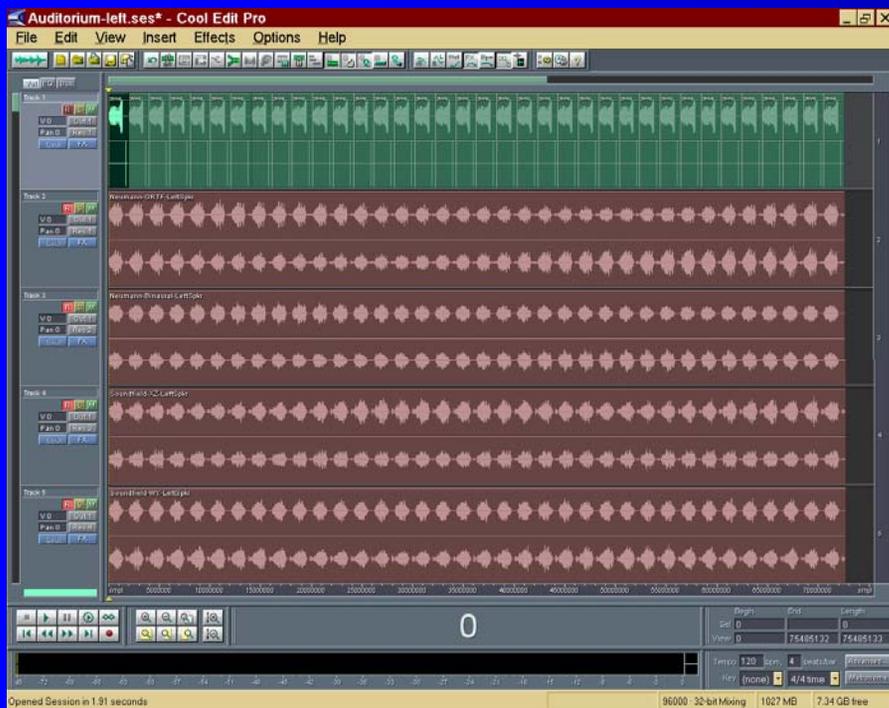


- Computer and sound card:
  - Signum Data Futureclient P-IV 1.8 GHz
  - Aardvark Pro Q-10 (8 ch., 96 kHz, 24 bits)

*24th AES International Conference*

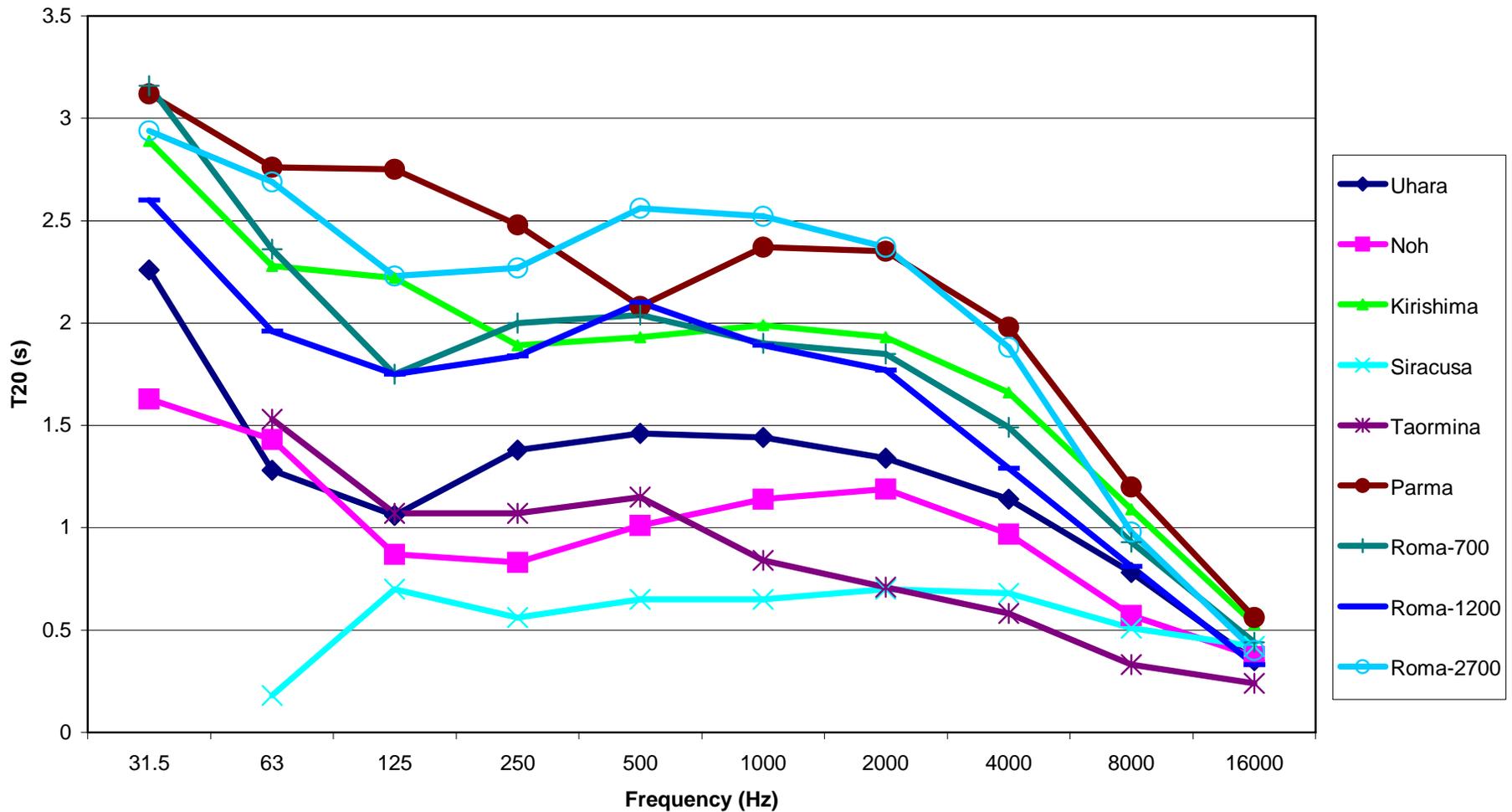
# Measurement procedure

- A single measurement session play backs 36 times the test signal, and simultaneously record the 8 microphonic channels



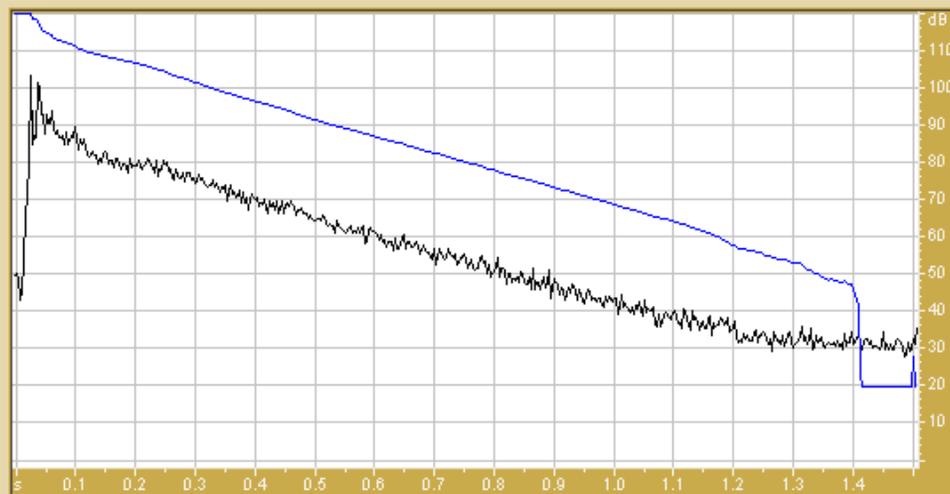
# Theatres measured

Reverberation Time T20



# Uhara Hall, Kobe, Japan

Acoustical Parameters according to ISO3382



OK Help

Save to File...

Copy to Clipboard

Store Reference Signal

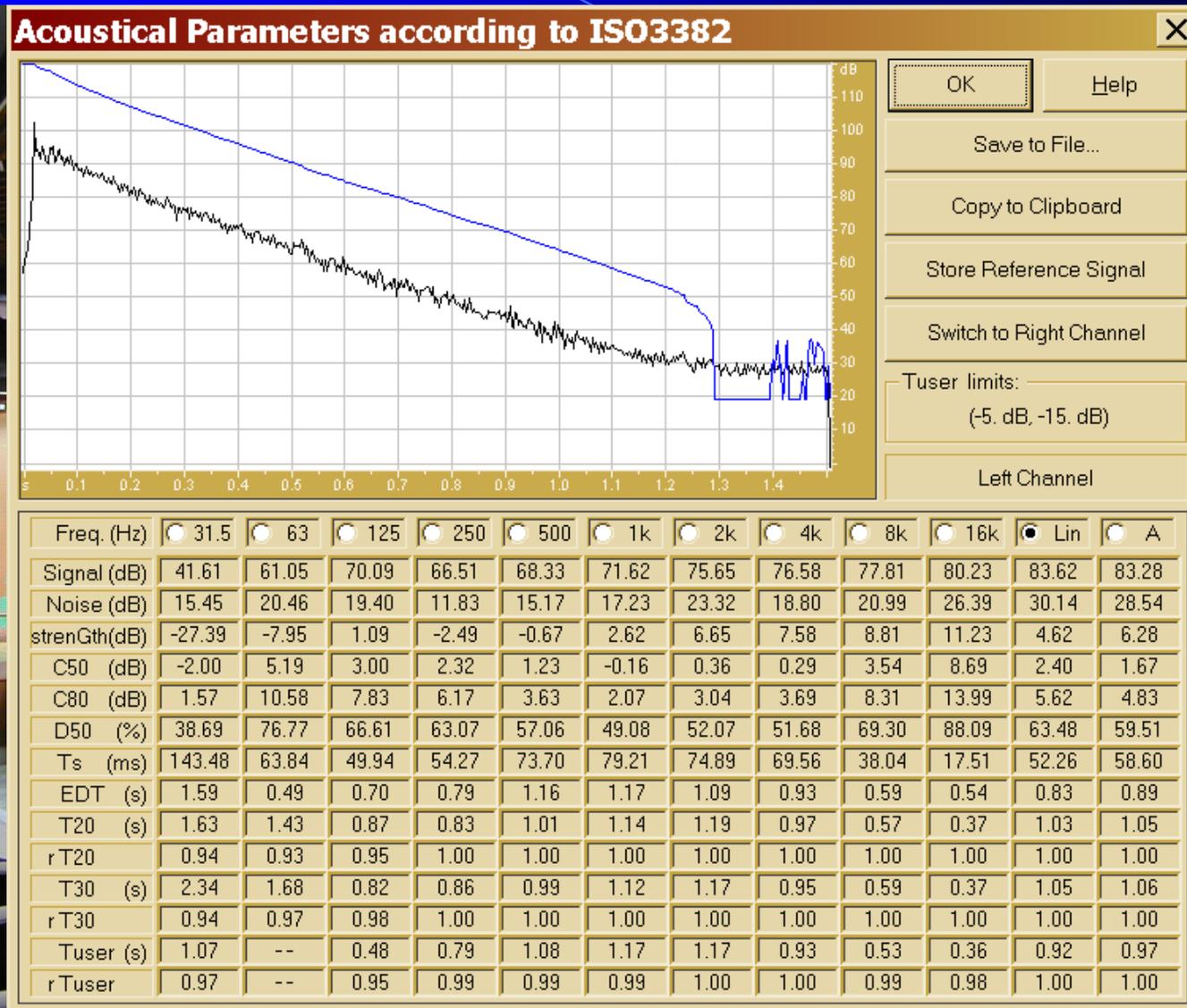
Switch to Right Channel

Tuser limits:  
(-5. dB, -15. dB)

Left Channel

Freq. (Hz)	<input type="radio"/> 31.5	<input type="radio"/> 63	<input type="radio"/> 125	<input type="radio"/> 250	<input type="radio"/> 500	<input type="radio"/> 1k	<input type="radio"/> 2k	<input type="radio"/> 4k	<input type="radio"/> 8k	<input type="radio"/> 16k	<input checked="" type="radio"/> Lin	<input type="radio"/> A
Signal (dB)	38.62	63.15	70.39	65.09	69.61	71.06	73.56	76.88	76.40	78.76	82.91	82.44
Noise (dB)	17.37	24.72	25.25	16.20	15.28	18.54	19.71	17.71	20.31	29.34	32.80	28.99
strenGth(dB)	-30.38	-5.85	1.39	-3.91	0.61	2.06	4.56	7.88	7.40	9.76	3.91	5.44
C50 (dB)	3.39	7.02	0.54	2.16	5.01	4.77	4.02	6.51	8.64	13.60	6.24	6.30
C80 (dB)	4.91	8.67	4.69	4.83	7.83	6.37	5.73	9.11	11.64	17.92	8.68	8.58
D50 (%)	68.59	83.45	53.11	62.19	76.02	75.01	71.60	81.74	87.97	95.82	80.80	81.02
Ts (ms)	102.03	56.49	82.94	74.39	45.93	51.61	53.52	33.91	23.70	15.08	36.27	36.17
EDT (s)	1.36	--	0.88	1.35	0.86	1.26	1.30	0.63	0.38	0.20	0.64	0.67
T20 (s)	2.26	1.28	1.06	1.38	1.46	1.44	1.34	1.14	0.78	0.35	1.22	1.24
r T20	0.97	0.98	0.96	1.00	0.98	0.99	1.00	0.99	0.99	0.98	1.00	1.00
T30 (s)	2.13	1.40	1.17	1.33	1.34	1.43	1.34	1.08	0.73	0.42	1.23	1.23
r T30	0.96	0.99	0.98	1.00	0.99	1.00	1.00	1.00	0.99	0.98	1.00	1.00
Tuser (s)	1.61	1.21	1.18	1.42	1.74	1.72	1.51	1.22	0.61	0.27	1.26	1.31
r Tuser	0.97	0.94	0.96	0.98	0.96	0.99	0.99	0.98	0.98	0.94	0.98	0.98

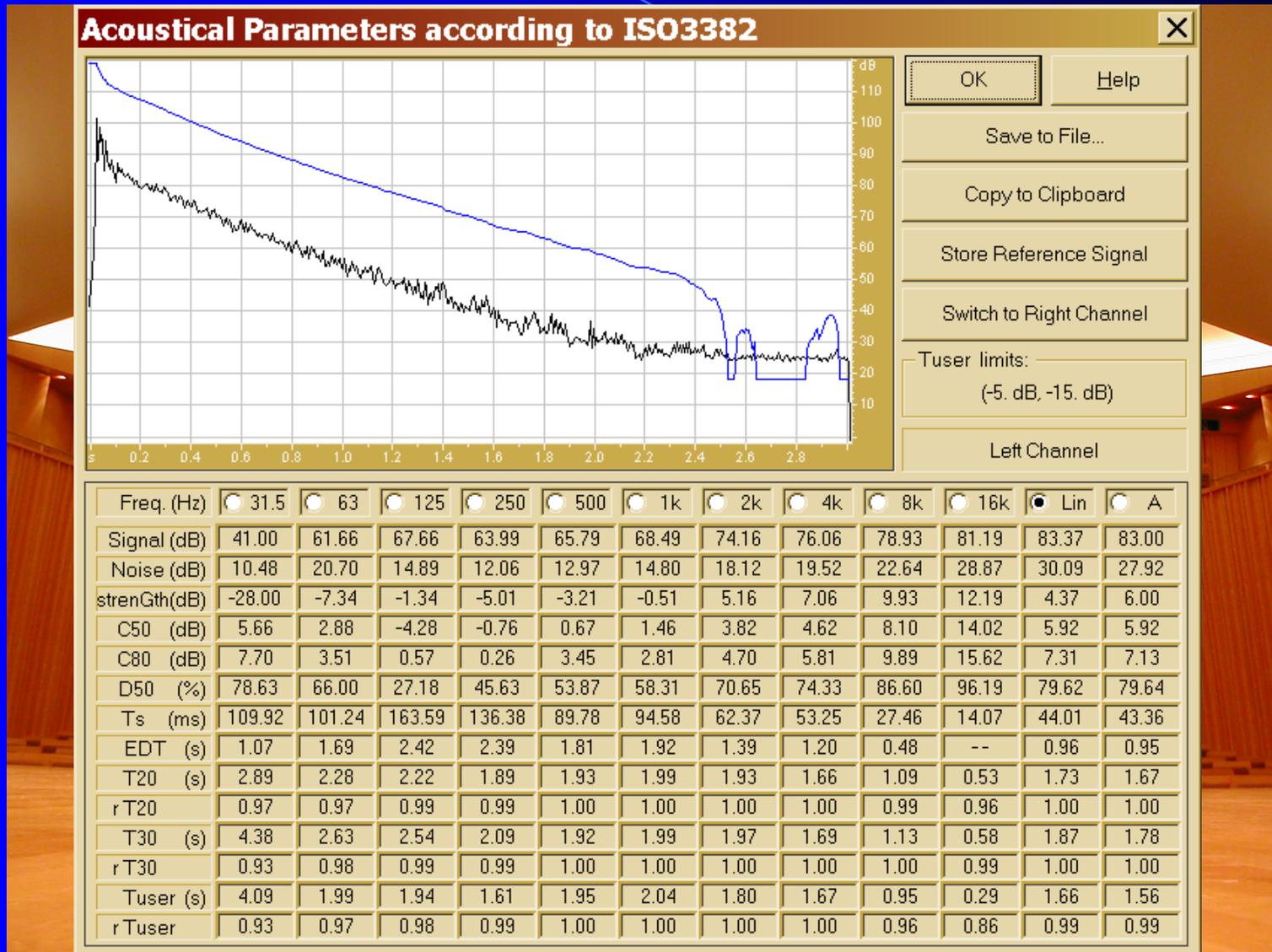
# Noh theater, Kobe, Japan



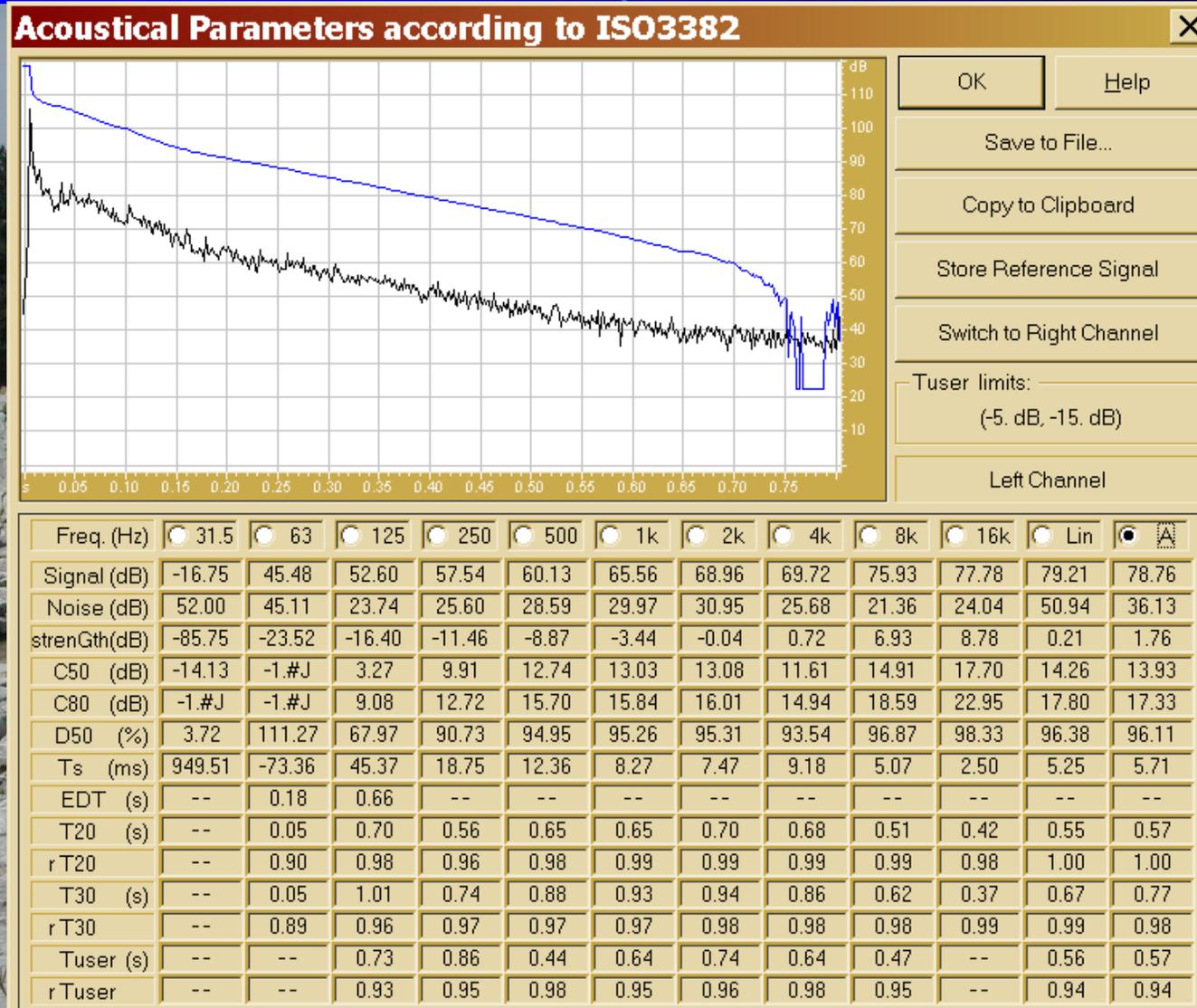
# Kirishima Concert Hall, Japan



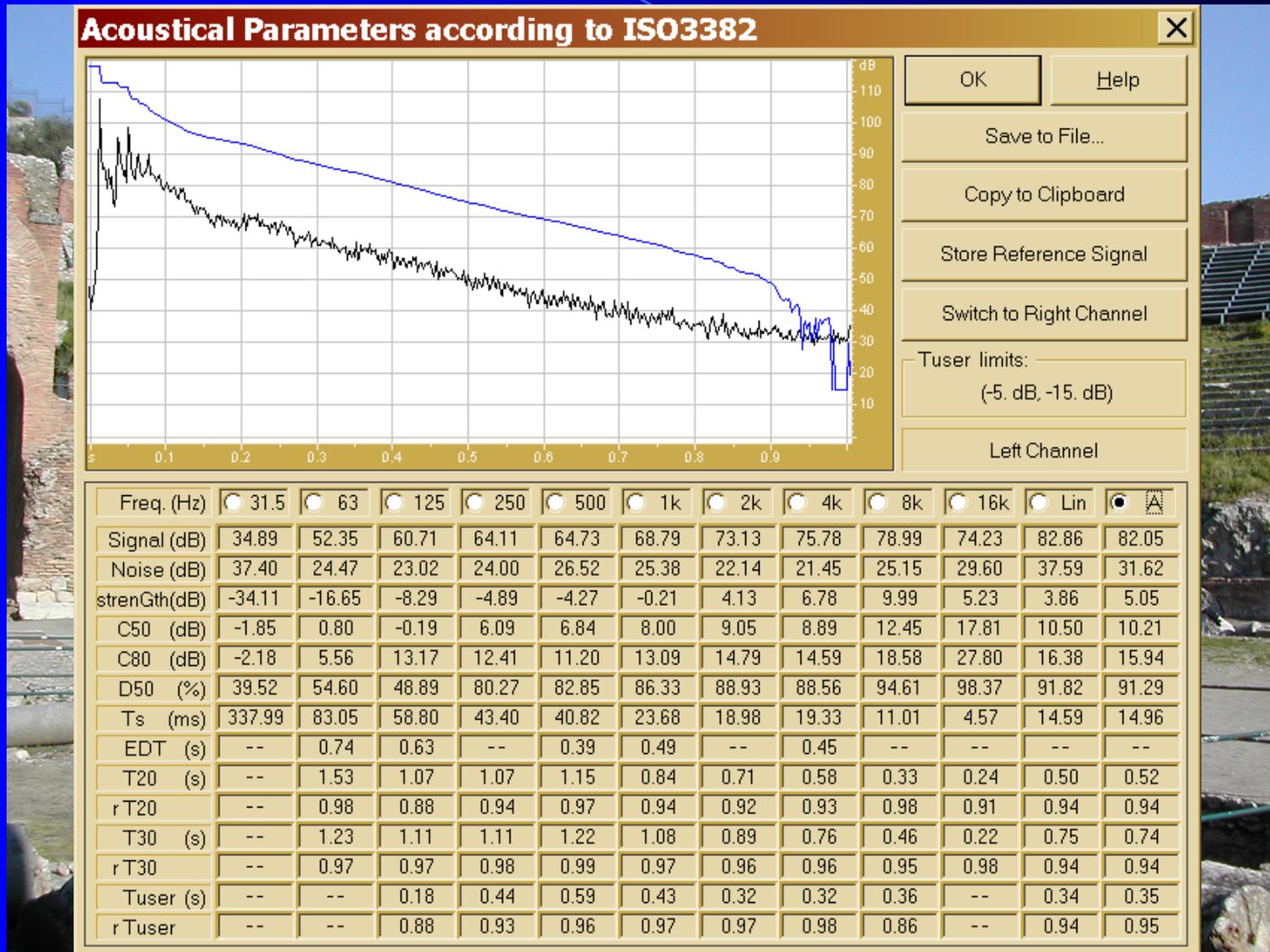
# Kirishima Concert Hall, Japan



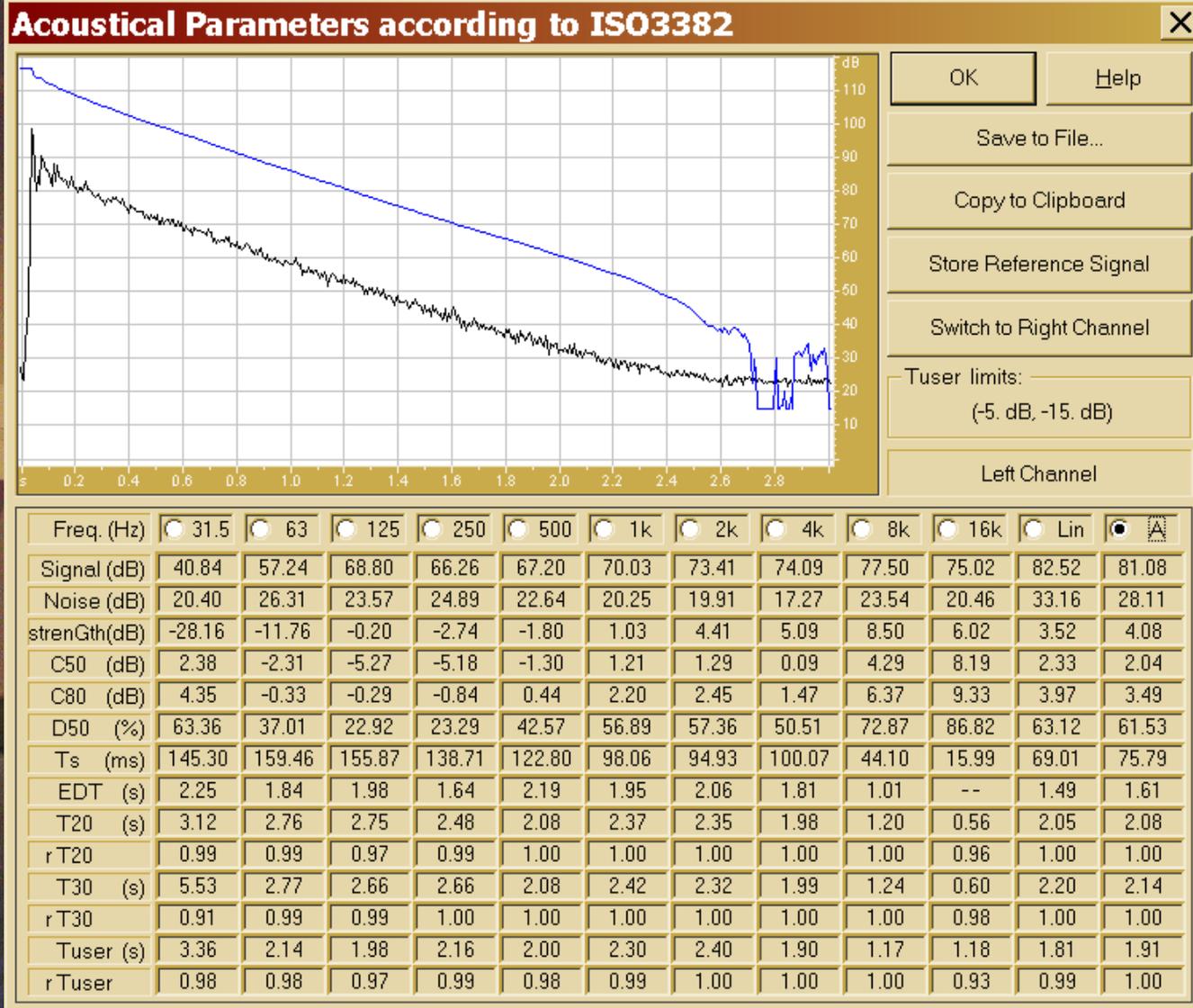
# Greek Theater in Siracusa



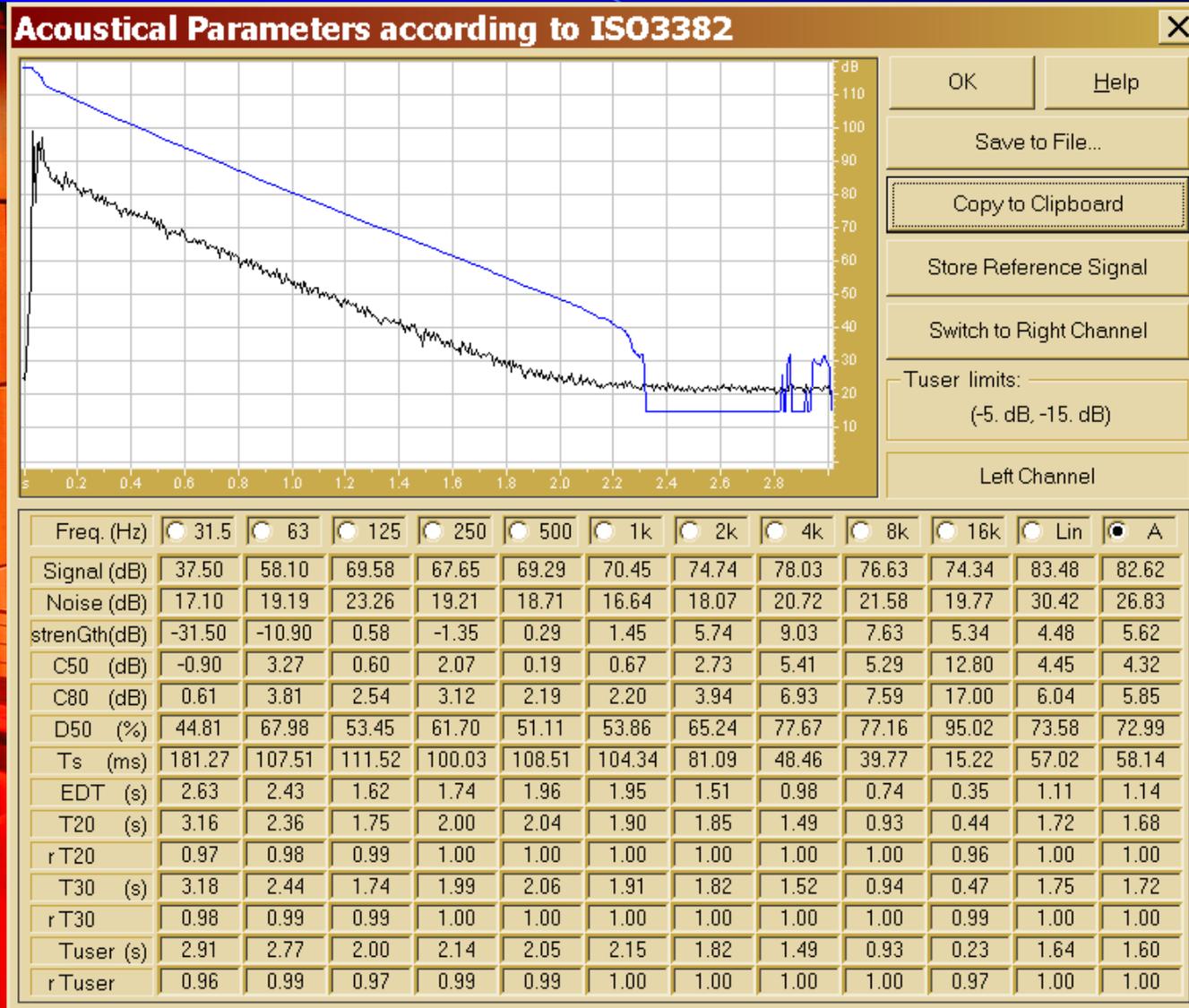
# Roman Theater in Taormina



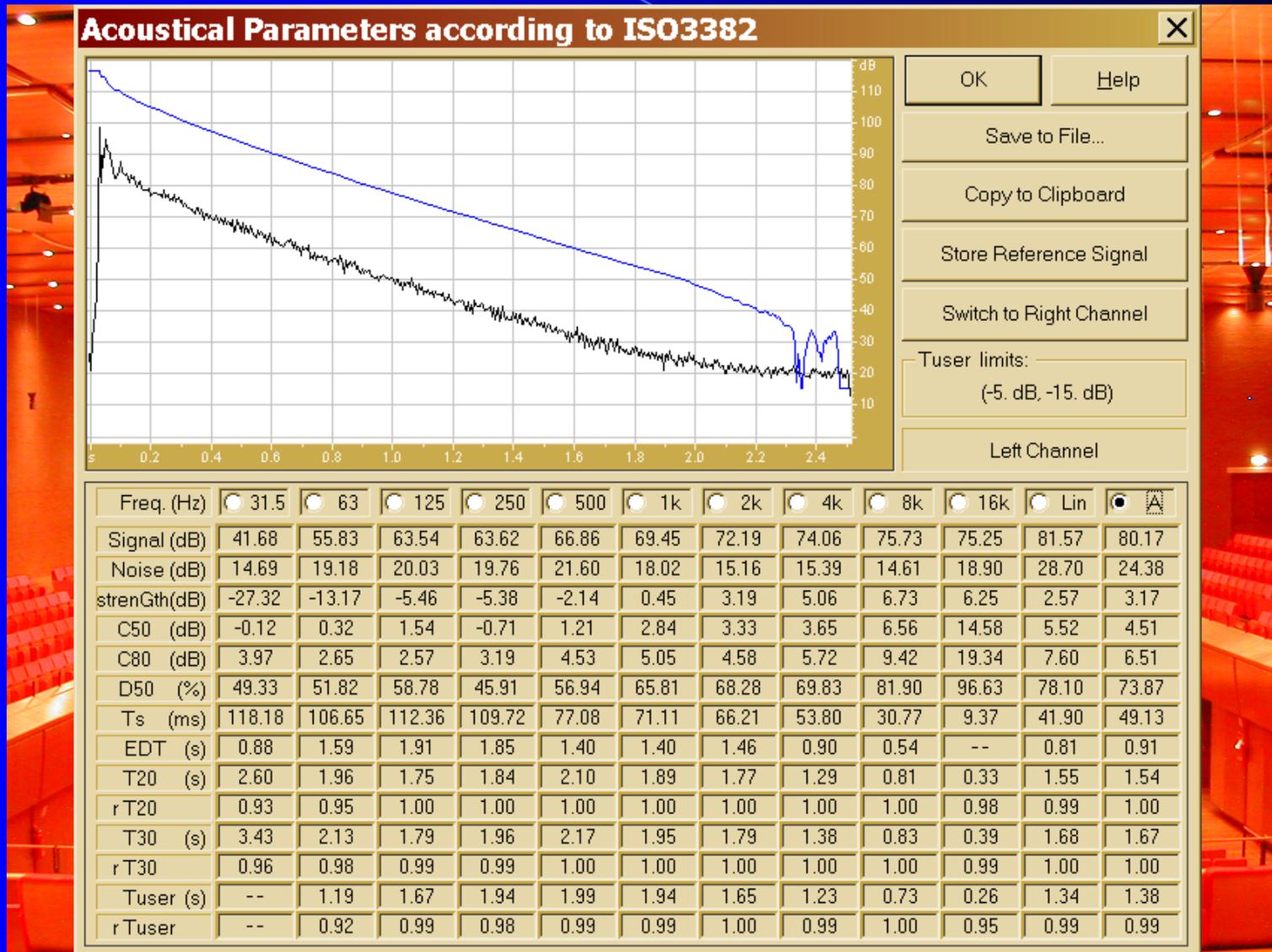
# Parma Auditorium, Italy



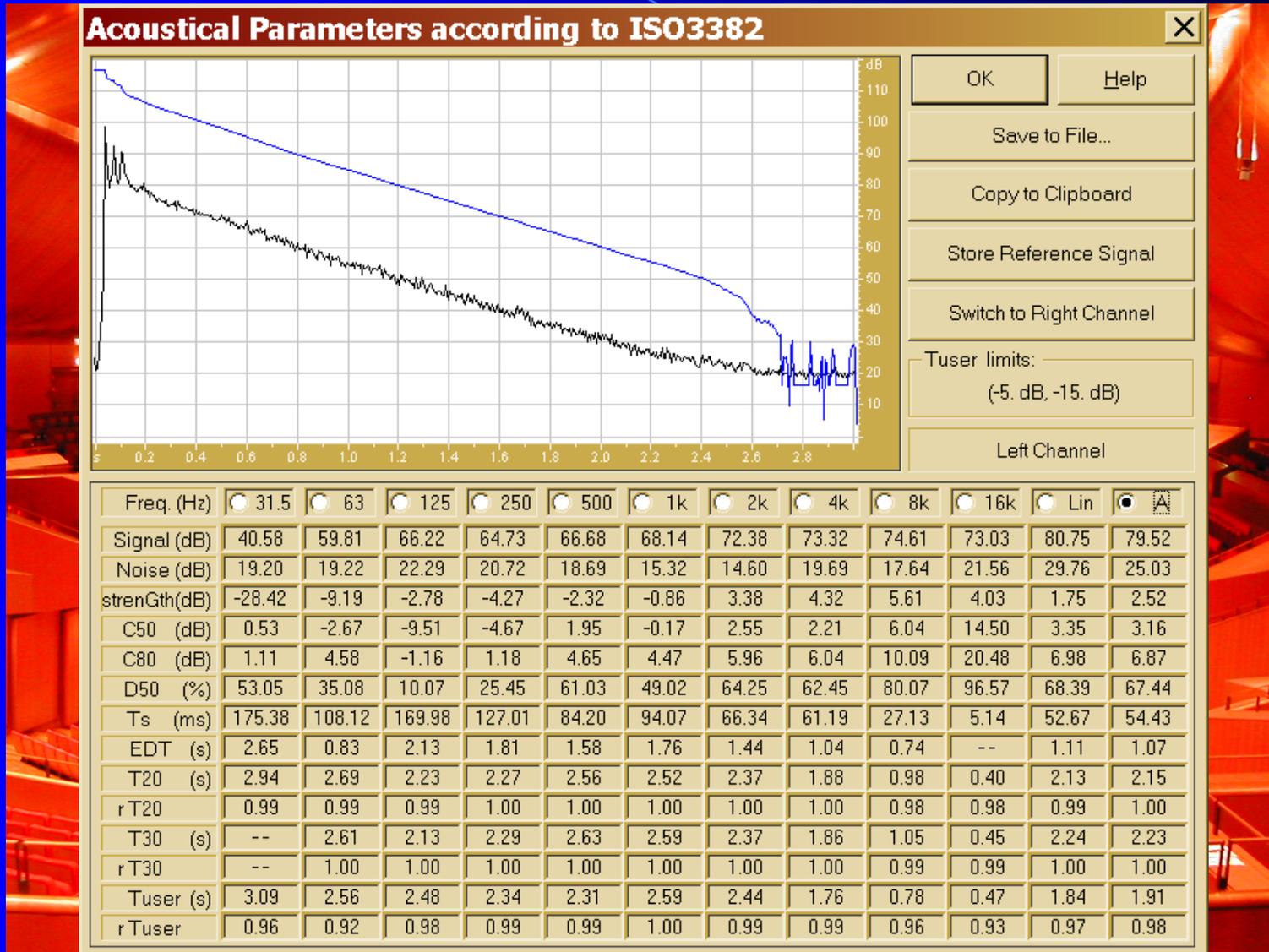
# Rome Auditorium, 700 seats



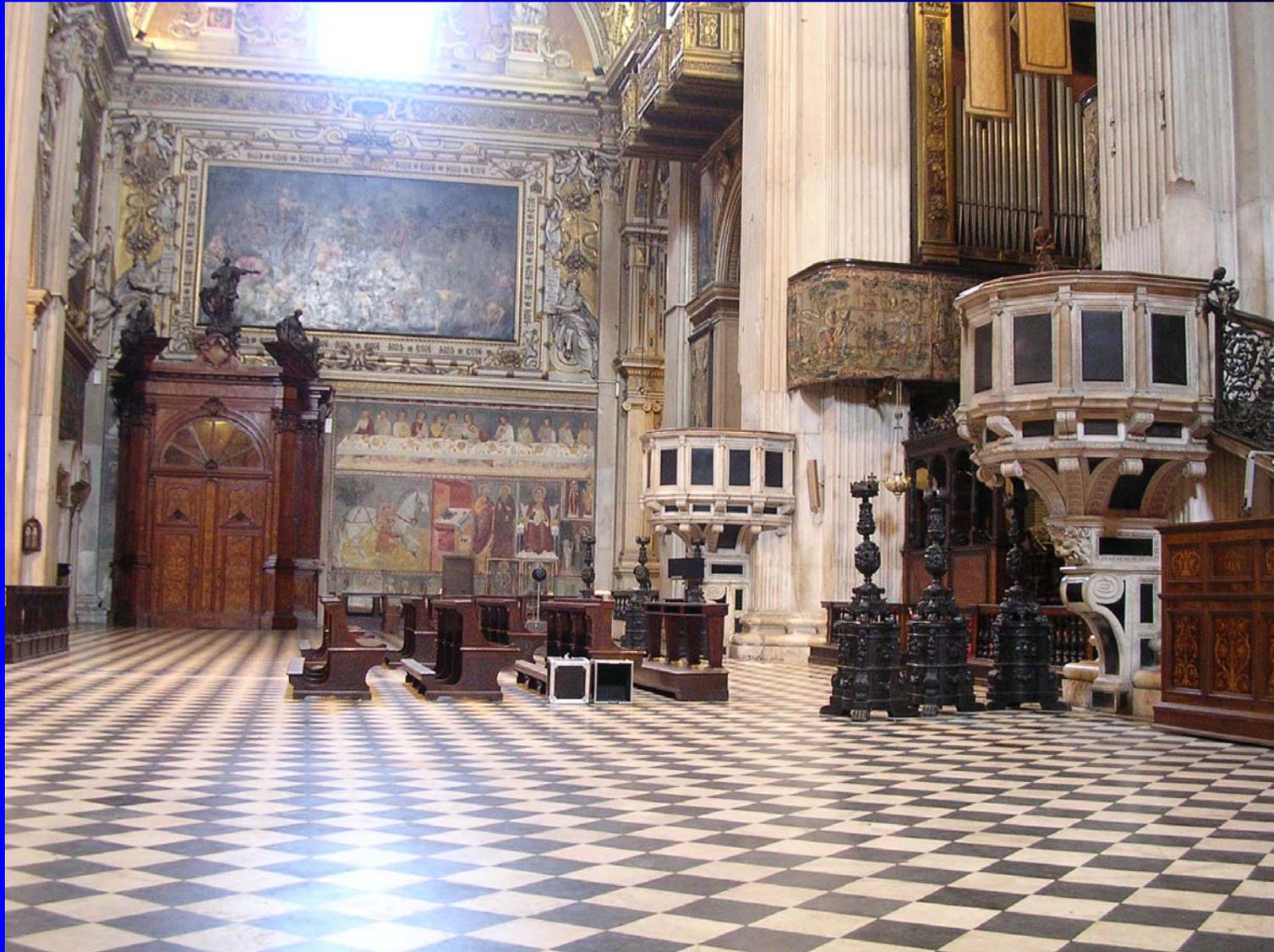
# Rome Auditorium, 1200 seats



# Rome Auditorium, 2700 seats



# Bergamo's Cathedral, Italy

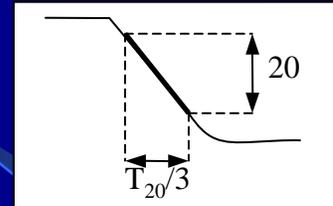


# Teatro Valli, Reggio Emilia, Italy



# Acoustical Parameters

- Reverberation Time  $T_{20}$ :
- Clarity  $C_{80}$ :
- Definition D:
- Center Time  $T_s$ :



$$C_{80} = 10 \cdot \lg \frac{\int_0^{80ms} p^2(\tau) \cdot d\tau}{\int_{80ms}^{\infty} p^2(\tau) \cdot d\tau}$$

$$D = \frac{\int_0^{50ms} p^2(\tau) \cdot d\tau}{\int_0^{\infty} p^2(\tau) \cdot d\tau} \cdot 100$$

$$T_s = \frac{\int_0^{\infty} \tau \cdot p^2(\tau) \cdot d\tau}{\int_0^{\infty} p^2(\tau) \cdot d\tau}$$

# Acoustical Parameters

● Strenght:

$$G = \text{SPL} - L_w + 31 \quad \text{dB}$$

● IACC:

$$\rho(\tau) = \frac{\int_{-\infty}^{\infty} h_d(\tau) \cdot h_s(\tau + t) \cdot d\tau}{\sqrt{\int_{-\infty}^{\infty} h_d^2(\tau) \cdot d\tau \cdot \int_{-\infty}^{\infty} h_s^2(\tau + t) \cdot d\tau}}$$

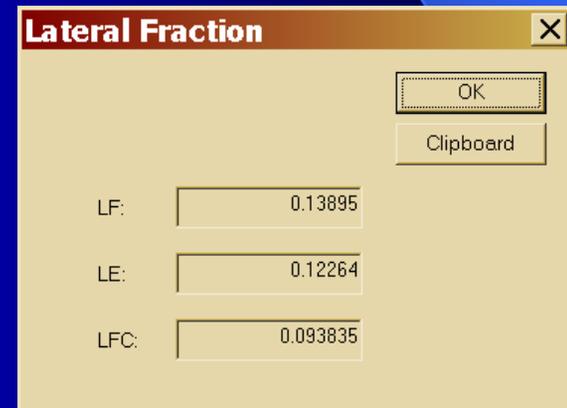
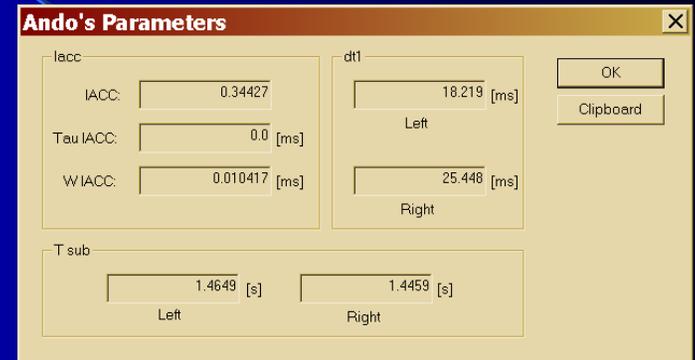
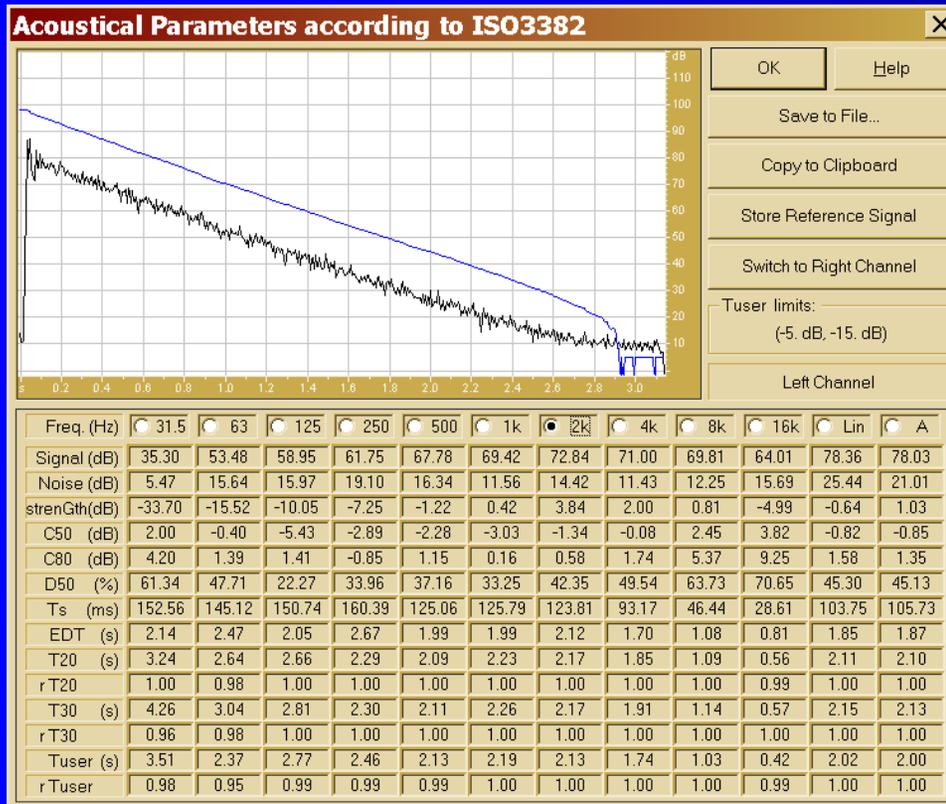
● LF:

$$LF = \frac{\int_{0ms}^{80ms} h_Y^2(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_W^2(\tau) \cdot d\tau}$$

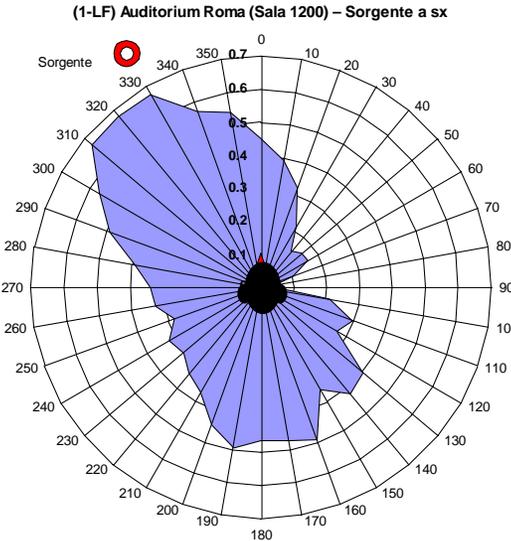
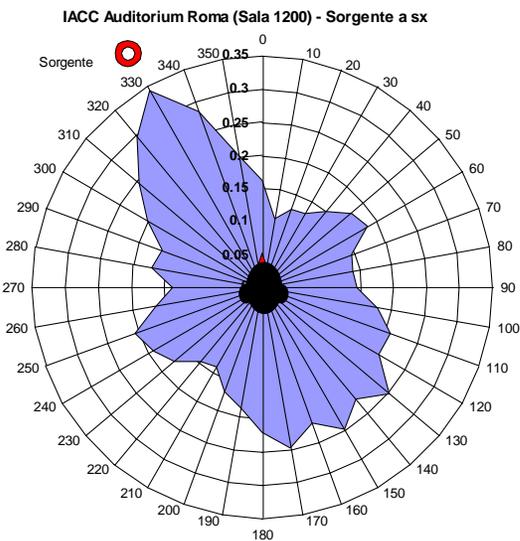
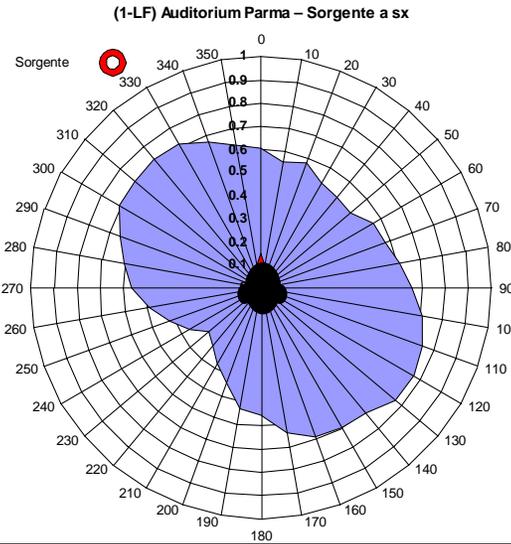
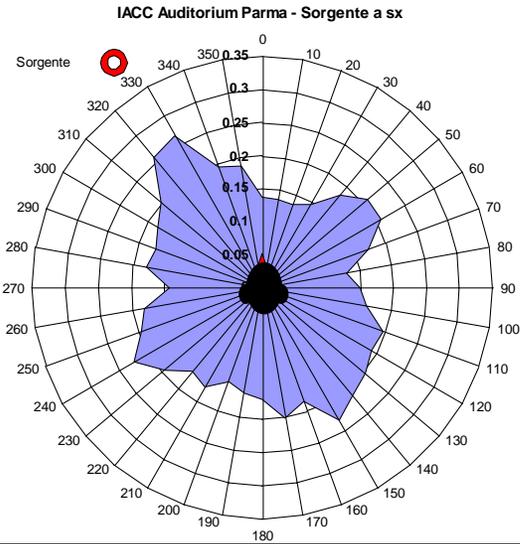
● LFC:

$$LFC = \frac{\int_{0ms}^{80ms} h_Y(\tau) \cdot h_W(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_W^2(\tau) \cdot d\tau}$$

# Analysis of spatial attributes



# Polar diagrams of IACC and (1-LF)



Auditorium	1-LF	IACC
Parma	0.725	0.266
Roma	0.676	0.344

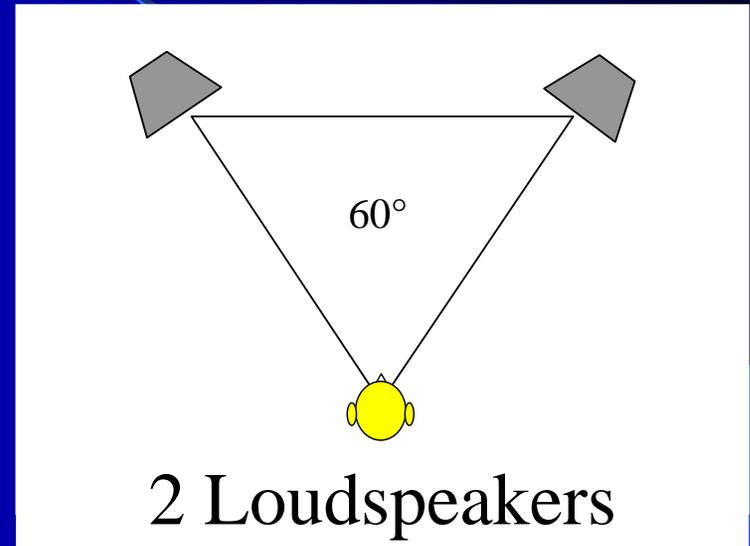
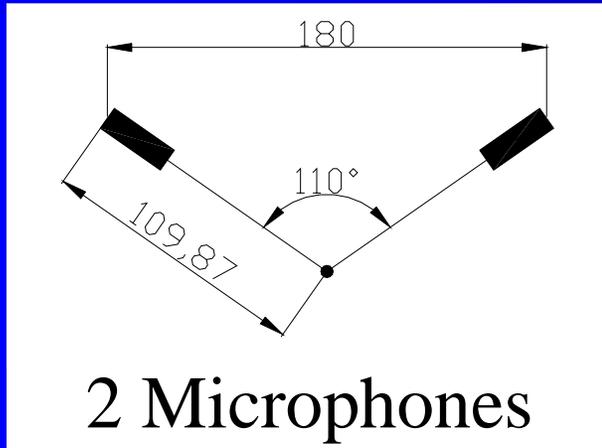
# Auralization by convolution

- The basic method consists in convolution of a dry signal with a set of impulse responses corresponding to the required output format for surround (2 to 24 channels).
- The convolution operation can nowadays be implemented very efficiently on a modern PC through an ancient algorithm (equally-partitioned FFT processing, Stockam 1966).

# Auralization types

- Stereo (ORTF on 2 standard loudspeakers at +/- 30°)
- Rotation-tracking reproduction on headphones (Binaural Room Scanning)
- Full 3D Ambisonics 1<sup>st</sup> order (decoding the B-format signal)
- ITU 5.1 (from different 5-mikes layouts)
- 2D Ambisonics 3<sup>rd</sup> order (from Mark Poletti's circular array microphone)
- Wave Field Synthesis (from the circular array of Soundfield microphones)
- Hybrid methods (Ambiophonics)

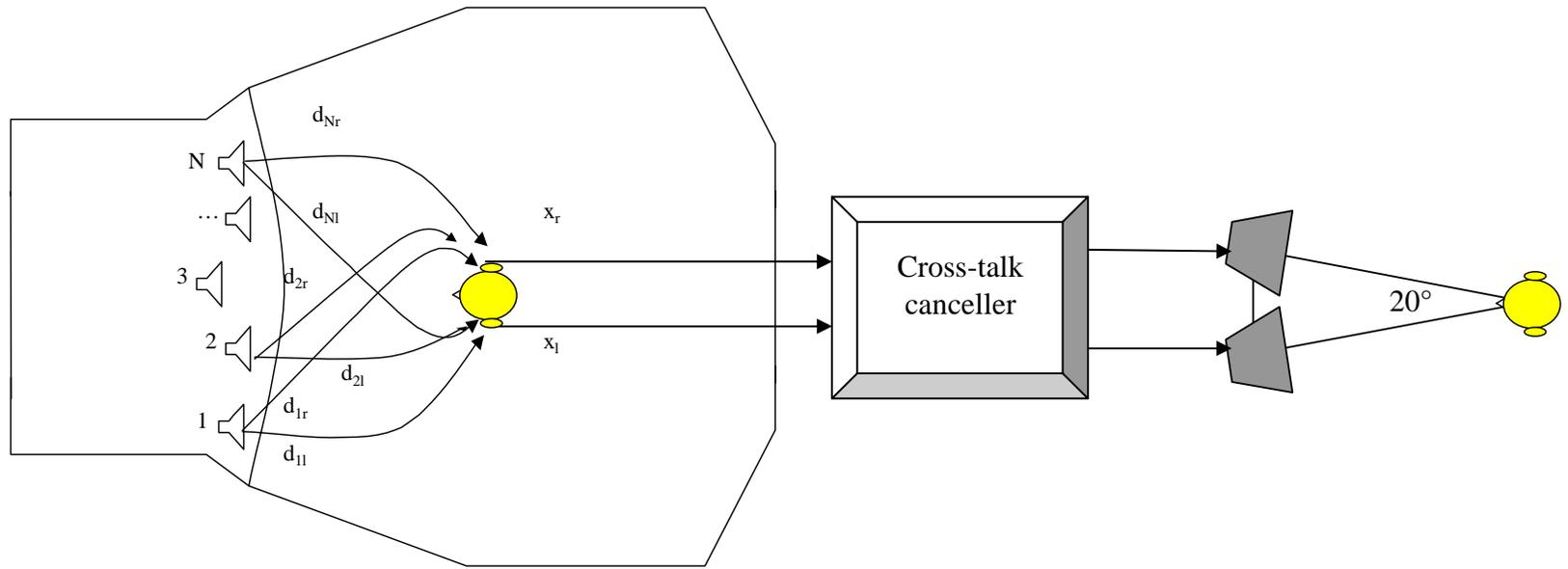
# ORTF Stereo



- Playback occurs over a pair of loudspeakers, in the standard configuration at angles of  $\pm 30^\circ$ , each being fed by the signal of the corresponding microphone

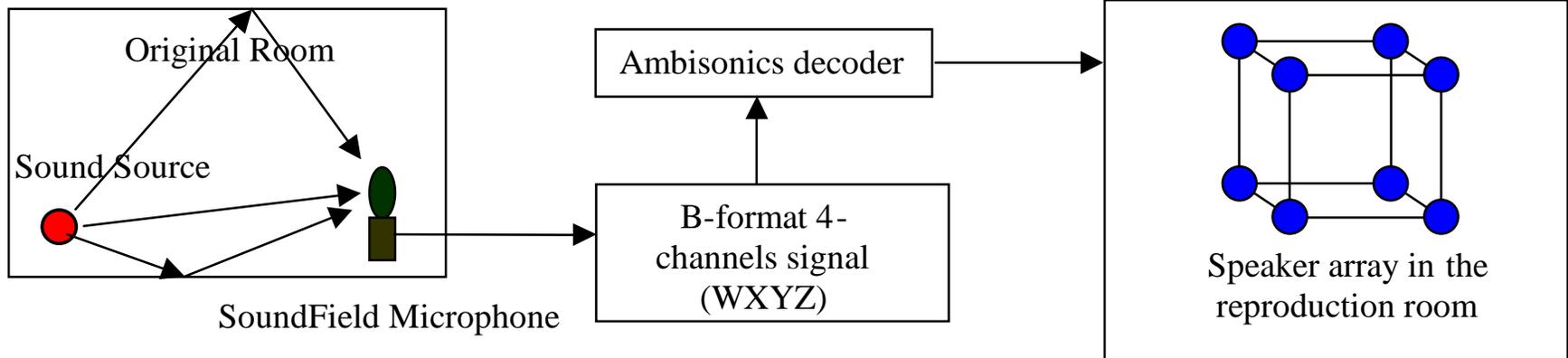
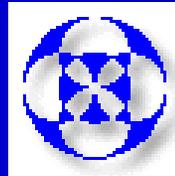
# Binaural (Stereo Dipole)

Original 2-channels recording of the signals coming from N sources



- Reproduction occurs over 2 loudspeakers angled at  $\pm 10^\circ$ , being fed through a “cross-talk cancellation” digital filtering system

# Ambisonics 3D 1<sup>st</sup> order



- Reproduction occurs over an array of 8-24 loudspeakers, through an Ambisonics decoder

# ITU 5.1 surround

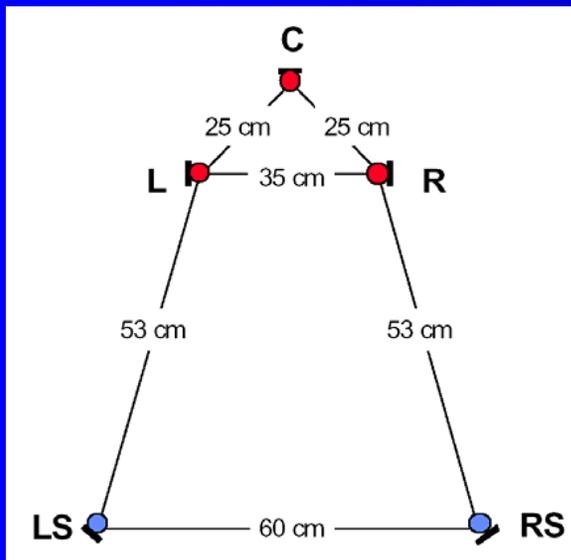
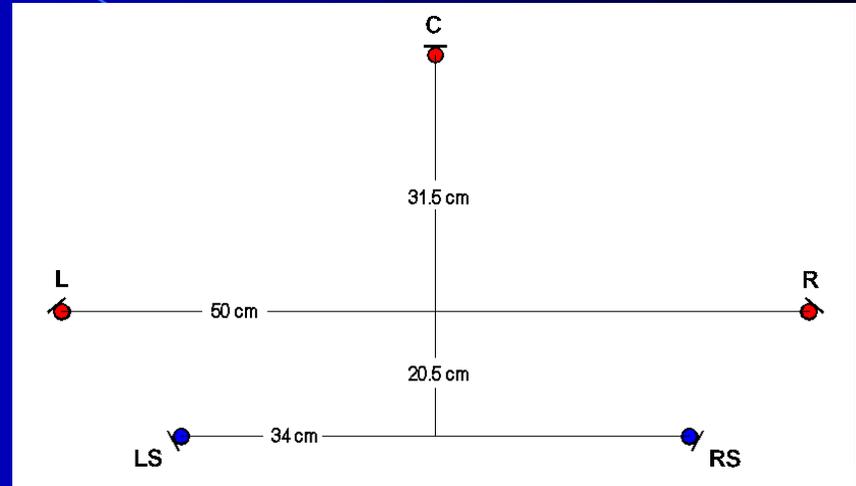
## ● Williams MMA

### Schematic of the setup

C : Cardioid,  $0^\circ$

L, R : Cardioid,  $\pm 40^\circ$

LS, RS : Cardioid,  $\pm 120^\circ$



## ● INA-5

### Schematic of the setup

C : Cardioid,  $0^\circ$

L, R : Cardioid,  $\pm 90^\circ$

LS, RS : Cardioid,  $\pm 150^\circ$

# ITU 5.1 surround

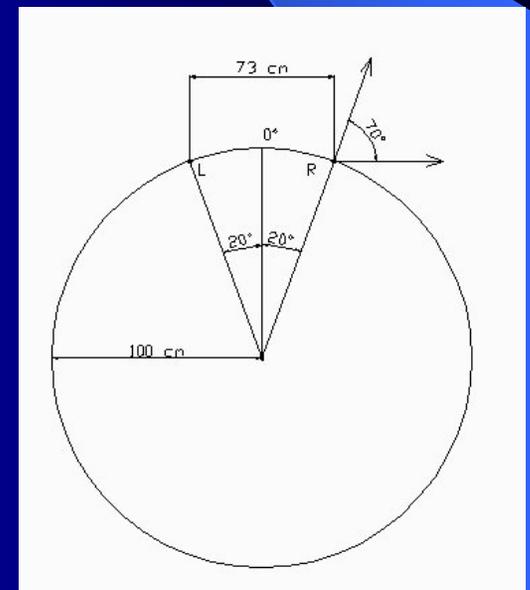
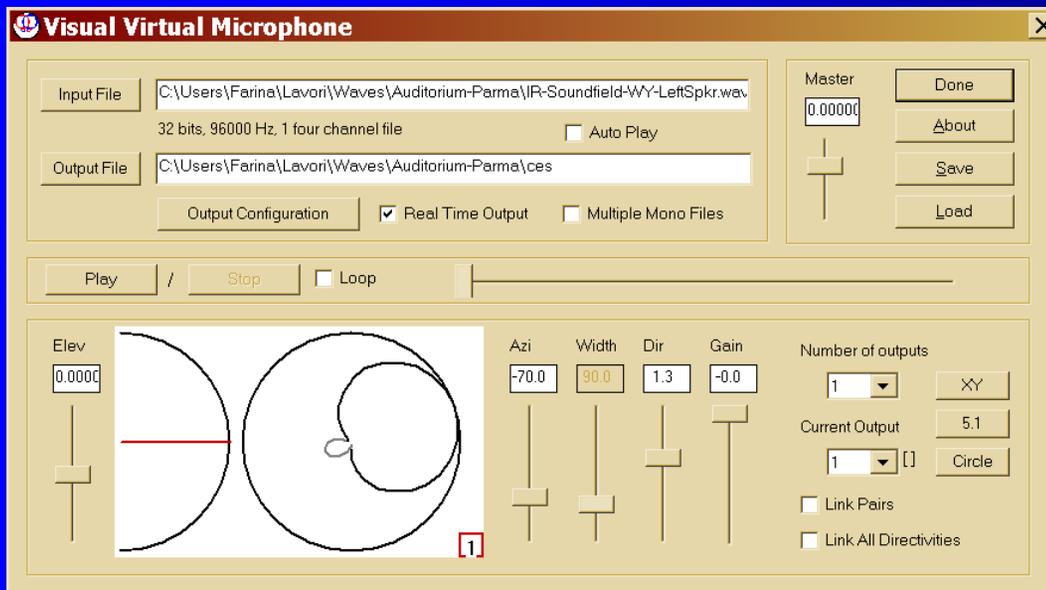
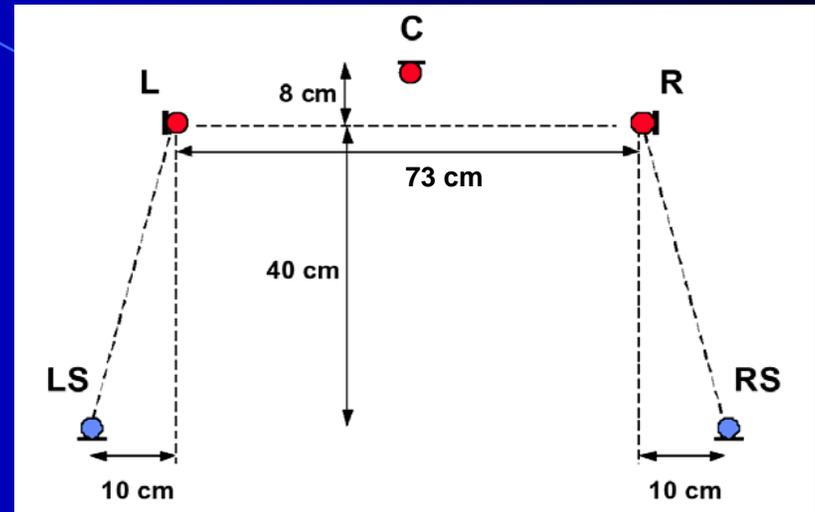
## ● OCT

### Schematic of the setup

C : Cardioid,  $0^\circ$

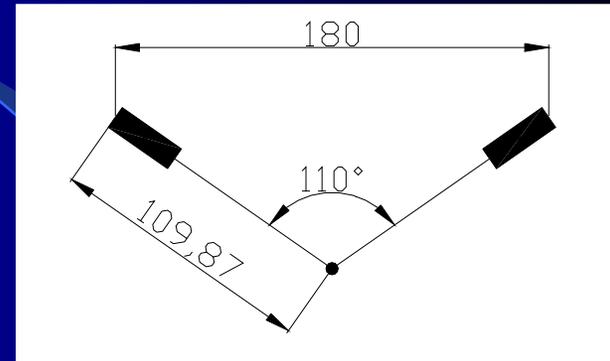
L, R : Super Cardioid,  $\pm 90^\circ$

LS, RS : Cardioid,  $\pm 180^\circ$

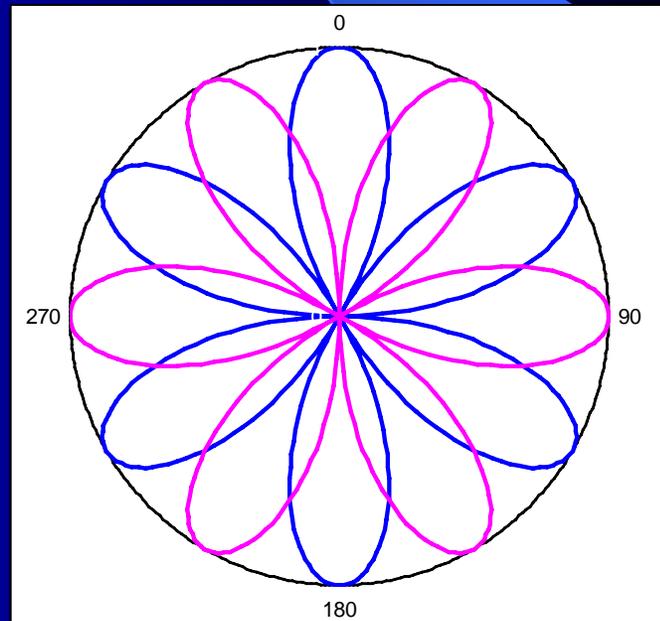


# Virtual high-order microphones (M. Poletti)

- One of the two ORTF cardioid is employed, which samples 36 positions along a 100 mm-radius circumference

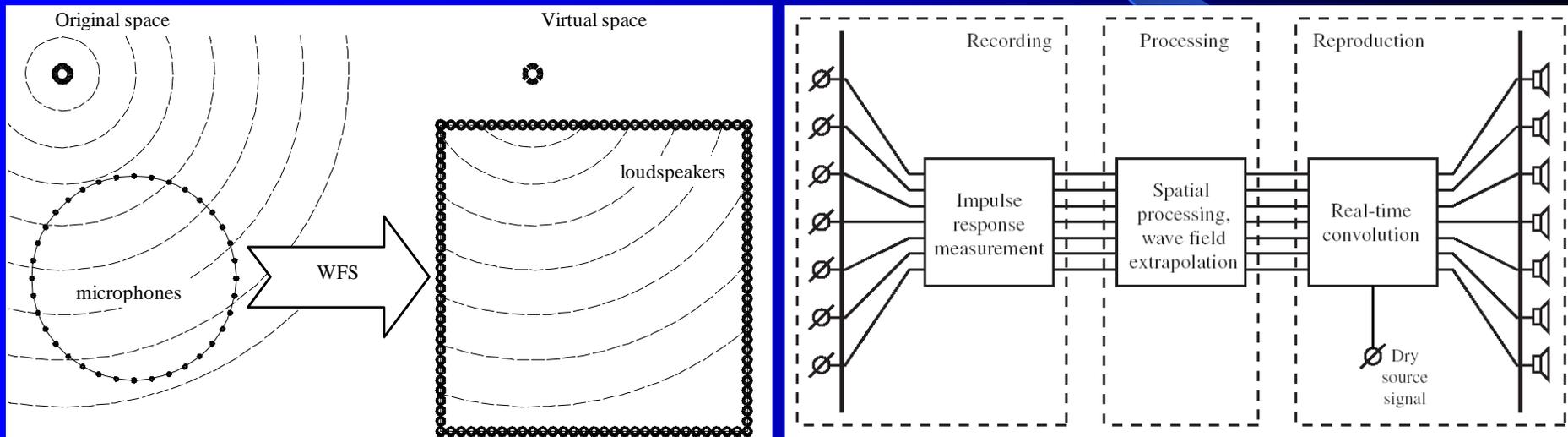


From these 36 impulse responses it is possible to derive the response of cylindrical harmonics microphones (2D Ambisonics) up to 5th order.



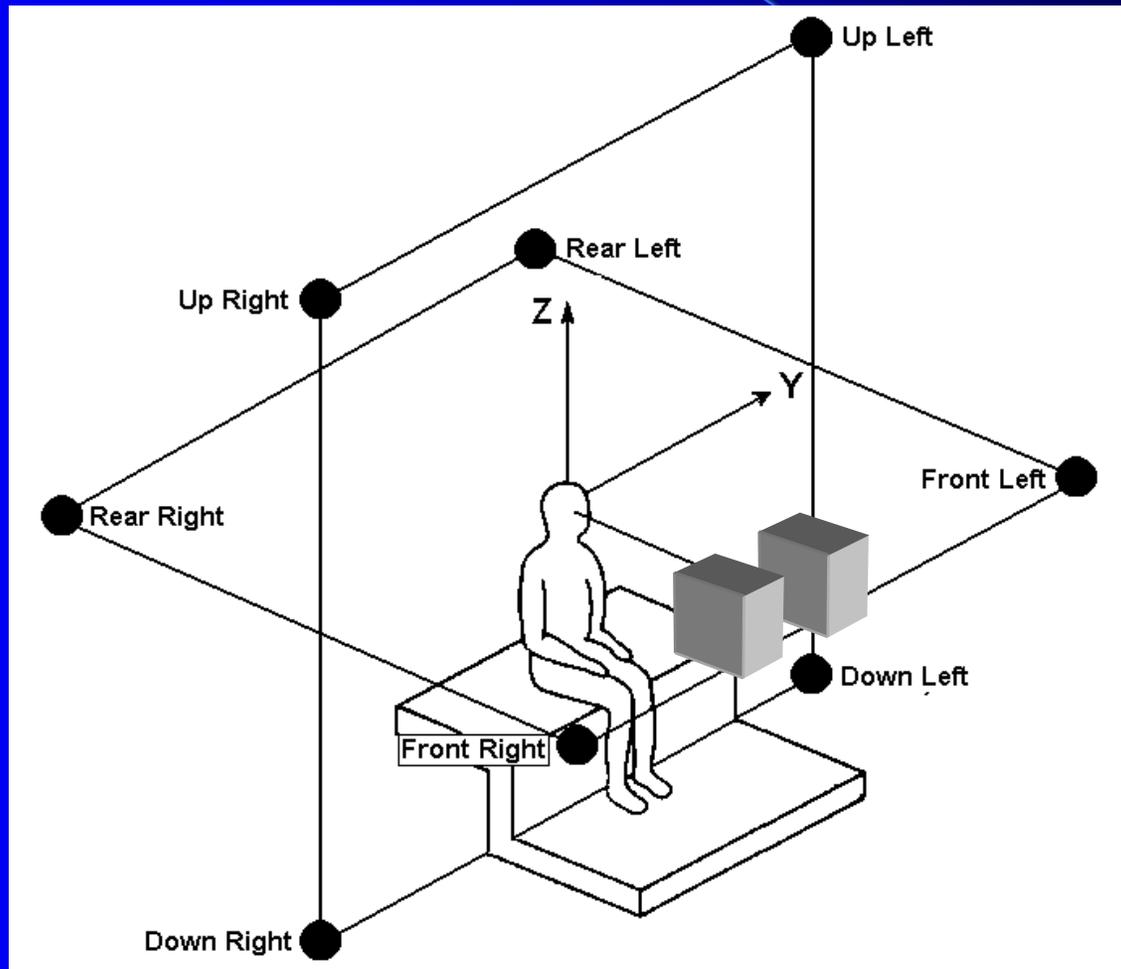
# Wave Field Synthesis (WFS)

- Flow diagram of the process



# Hybrid methods (Ambiophonics)

- Ambiophonics 3D (10 loudspeakers):



# Conclusions

- Main advantages of the new measurement method “Waves 2003”:
  - Almost all previously known measurement techniques are incorporated in a single, coherent approach
  - The spatial informations are accurately sampled, making it possible to store, analyze and preserve these “3D acoustical photographies” of existing musical spaces for the posterity
  - The impulse response are stored in many different formats, allowing for their usage for surround productions with today technologies (ITU 5.1, 1st order Ambisonics) and future, more advanced methods (high order Ambisonics, WFS, Ambiophonics)