



International Symposium on Room Acoustics : Design and Science 2004



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Advanced techniques for measuring and reproducing spatial sound properties of auditoria



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and

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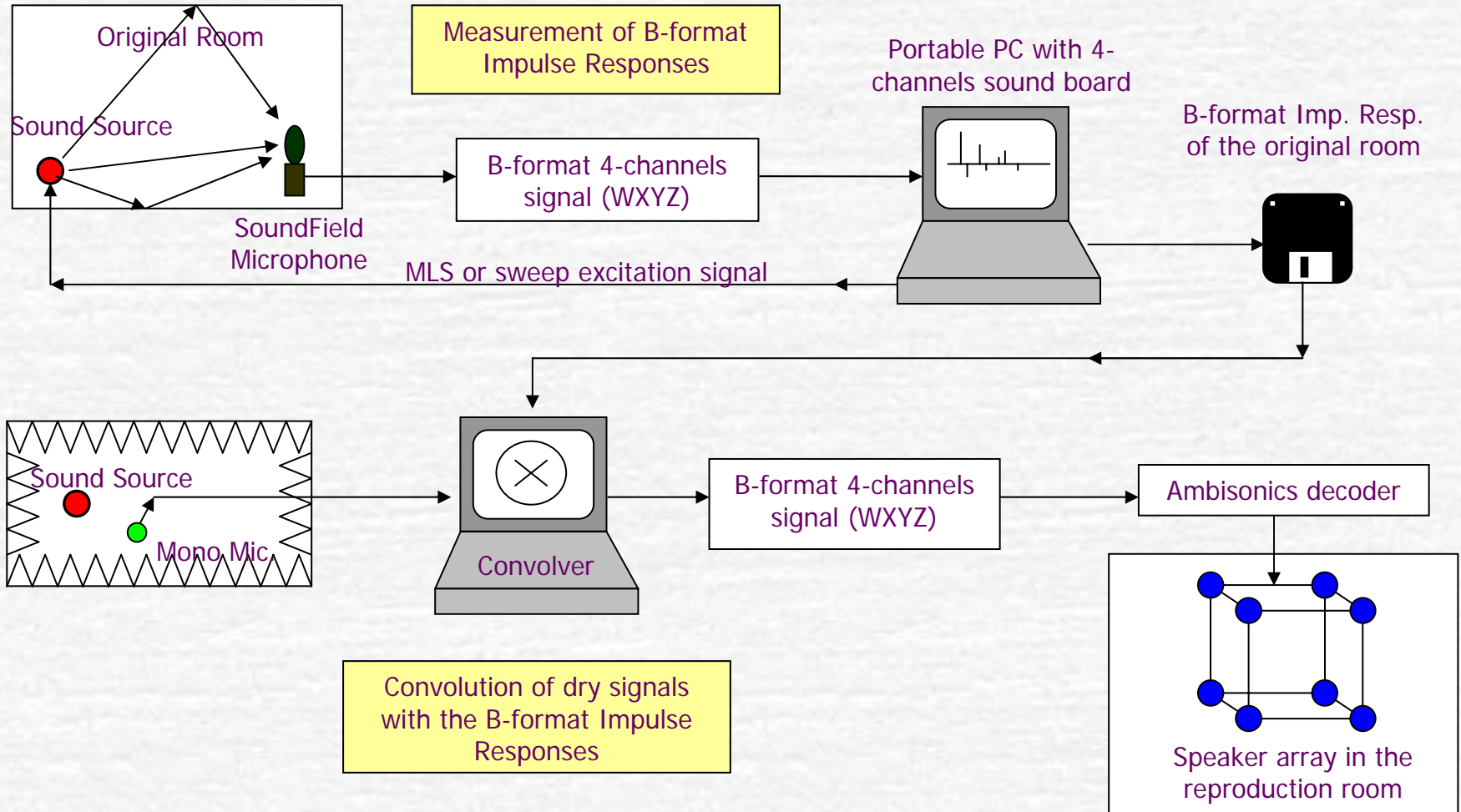
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Topics

- This paper is a tribute to M. Gerzon, who had foreseen 3D impulse response measurements and 3D Auralization obtained by convolution.
- The advantages (and disadvantages) of employing measured IRs
- Comparison between Auralizations based on calculated and measured IRs (e.g. Theatre "La Fenice", Venice)
- Different approaches to 3D Auralization

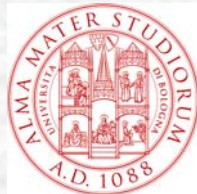


Concept (Gerzon, 1975)



The measurements of IRs:

1. In case something happens to the original space (e.g.: 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
3. Several configurations of listening rooms could be developed, by means of multichannel auralization, for subjective tests



Theatre la Fenice, Venice



- The first theatre was realised in 1792 by Gian Antonio Selva, after the burning of Teatro San Benedetto
- In December 1836 the theatre burned down again and was rebuilt by G. and T. Meduna the year after
- The theatre was closed in 1995 for maintenance; it had to open again in February 1, 1996, but it burned two days before (January 29, 1996)
- A few weeks before the fire, L. Tronchin measured binaural impulse responses

Impulse Responses of La Fenice

- In 27 positions a series of binaural impulse responses (with gun shots) was recorded
- Each recording is consequently a stereo file at 16 bits, 48 kHz
- During measurements the room was perfectly fitted, whilst the stage was empty (no scenery)



Point n. 12

Example n. 1 – La Fenice



Stop

Overture alle Nozze di Figaro di Mozart

- [Dry music](#)
- [Convolution with experimental I.R. \(pt. 12\)](#)
- [Convolution with simulated IR](#)

Preludio al primo atto della Traviata di G.Verdi

- [Dry music](#)
- [Convolution with experimental I.R. \(pt. 12\)](#)
- [Convolution with simulated IR](#)

Example n. 2 – Paris vs ...la *petit Paris*



Cité de la Musique, Parigi



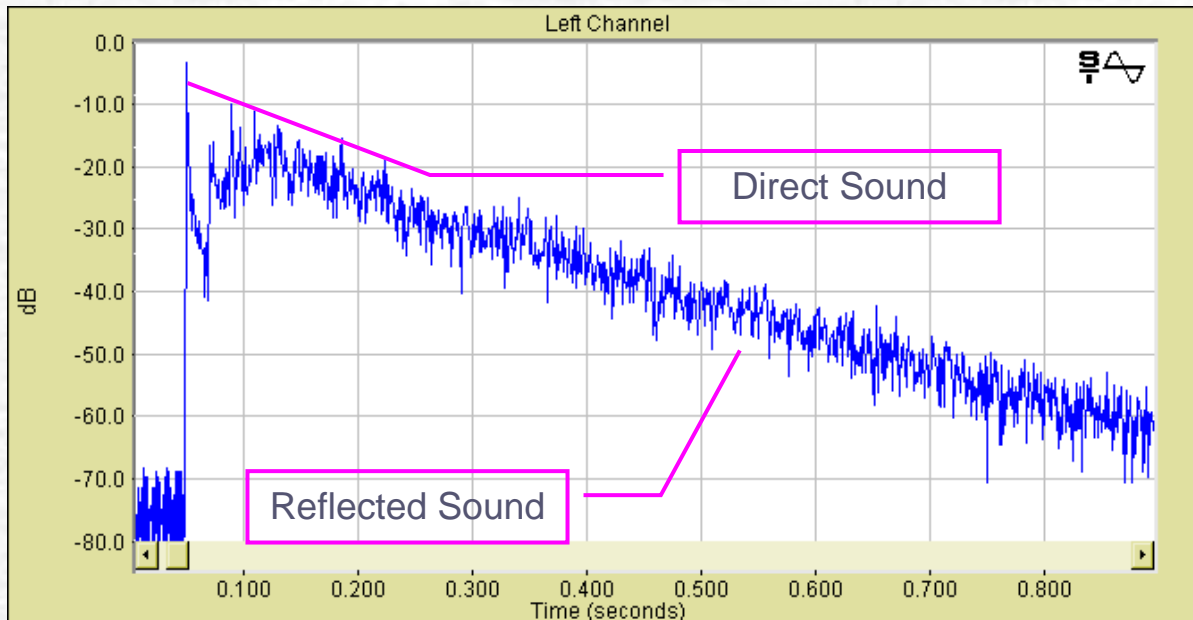
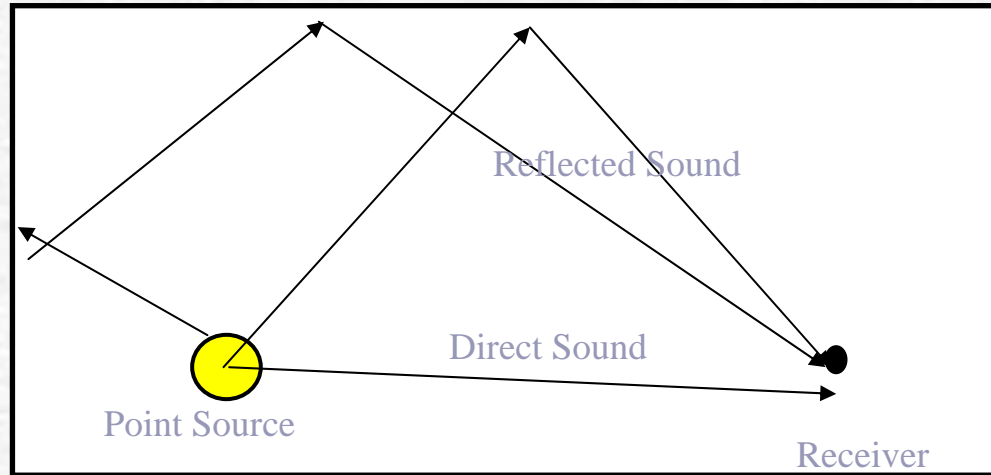
Auditorium di Parma

Stop

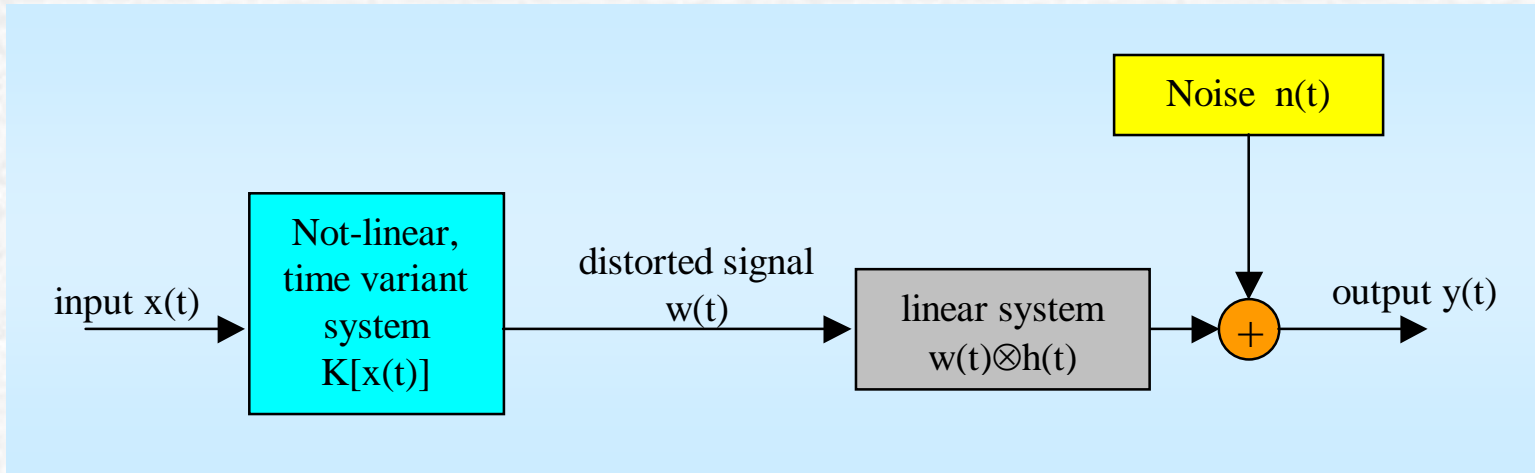
Advanced IR capture and rendering (project)

- Description of the measurement technique
- Analysis of some acoustical parameters of some theaters 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, musical production and for scientific Auralization tests

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 $n(t)$.

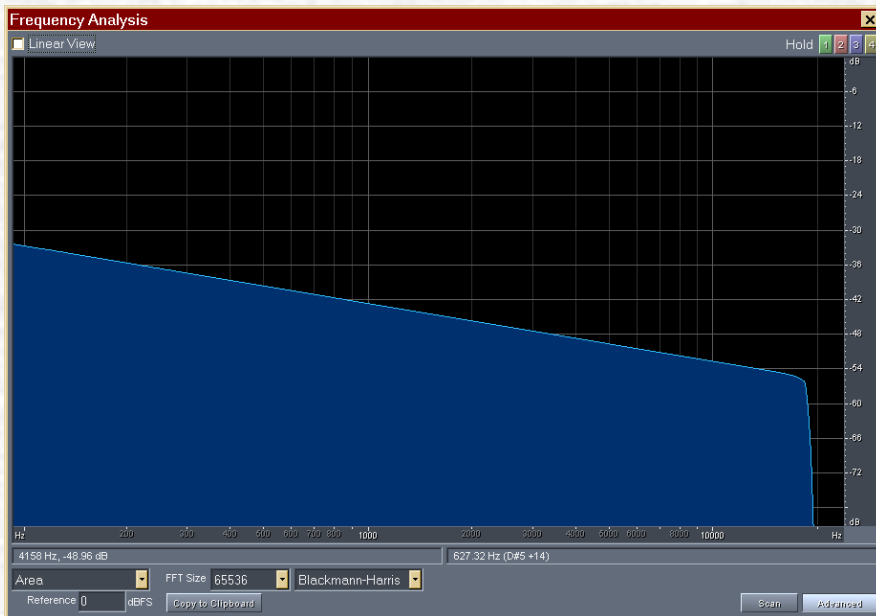
Test signal: Log Sine Sweep

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

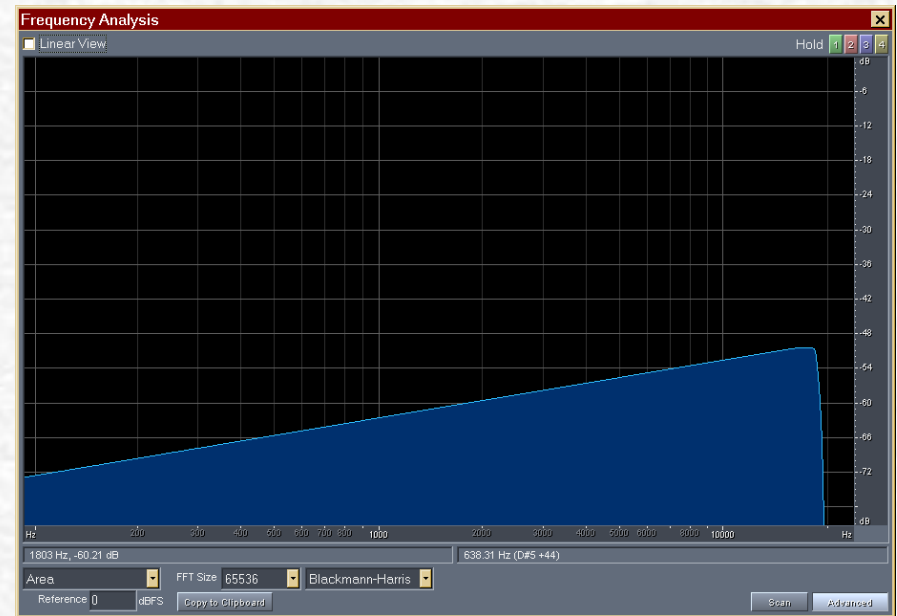
$$x(t) = \sin \left[\frac{2 \cdot \pi \cdot f_1 \cdot T}{\ln \left(\frac{f_2}{f_1} \right)} \cdot \left(e^{\frac{t}{T} \cdot \ln \left(\frac{f_2}{f_1} \right)} - 1 \right) \right]$$

Deconvolution of Log Sine Sweep

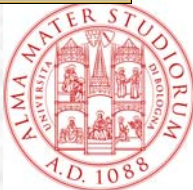
The "time reversal mirror" technique is employed: the system's impulse response is obtained by convolving the measured signal $y(t)$ with the time-reversal of the test signal $x(-t)$. As the log sine sweep does not have a "white" spectrum, proper equalization is required



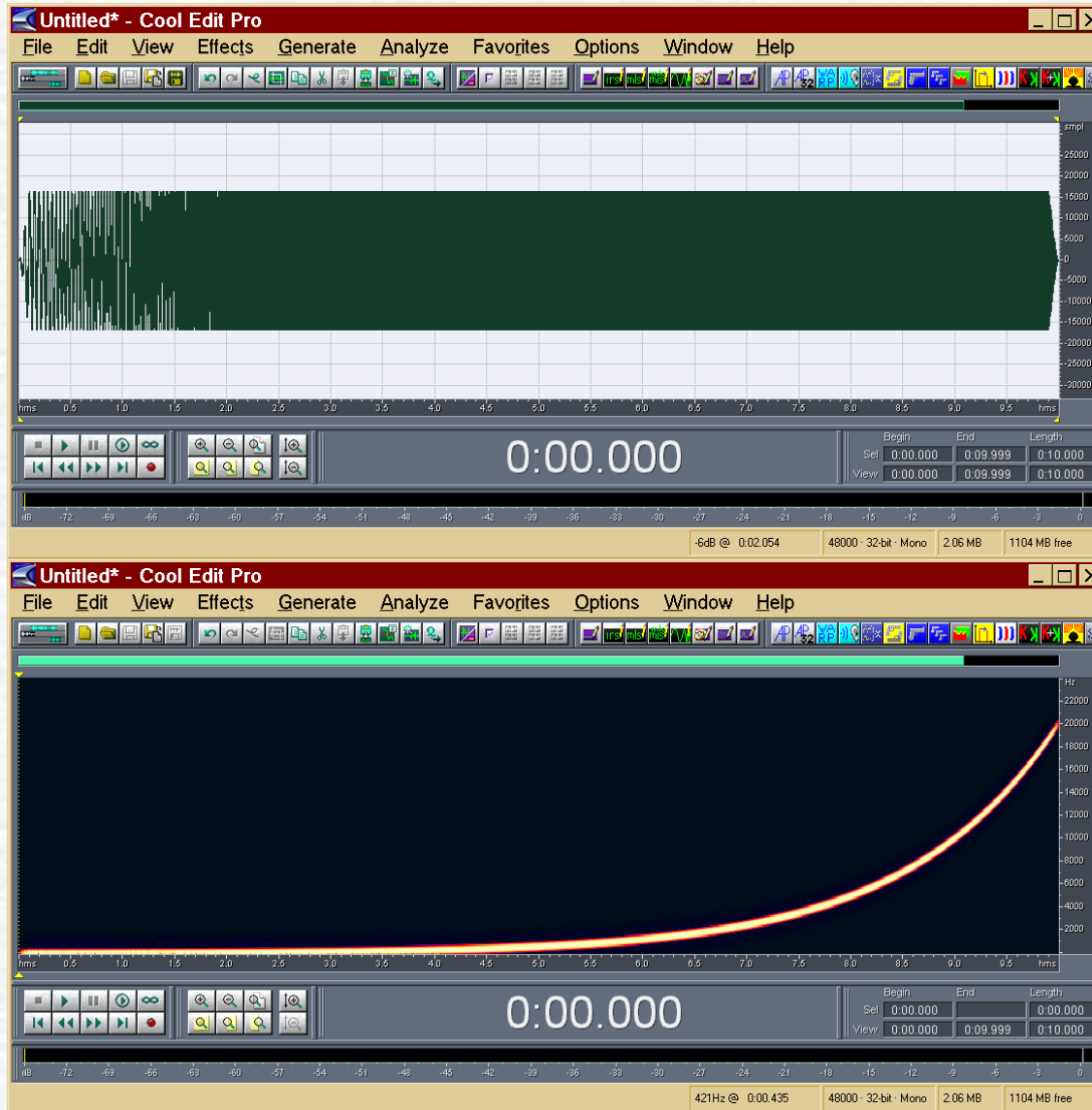
Test Signal $x(t)$



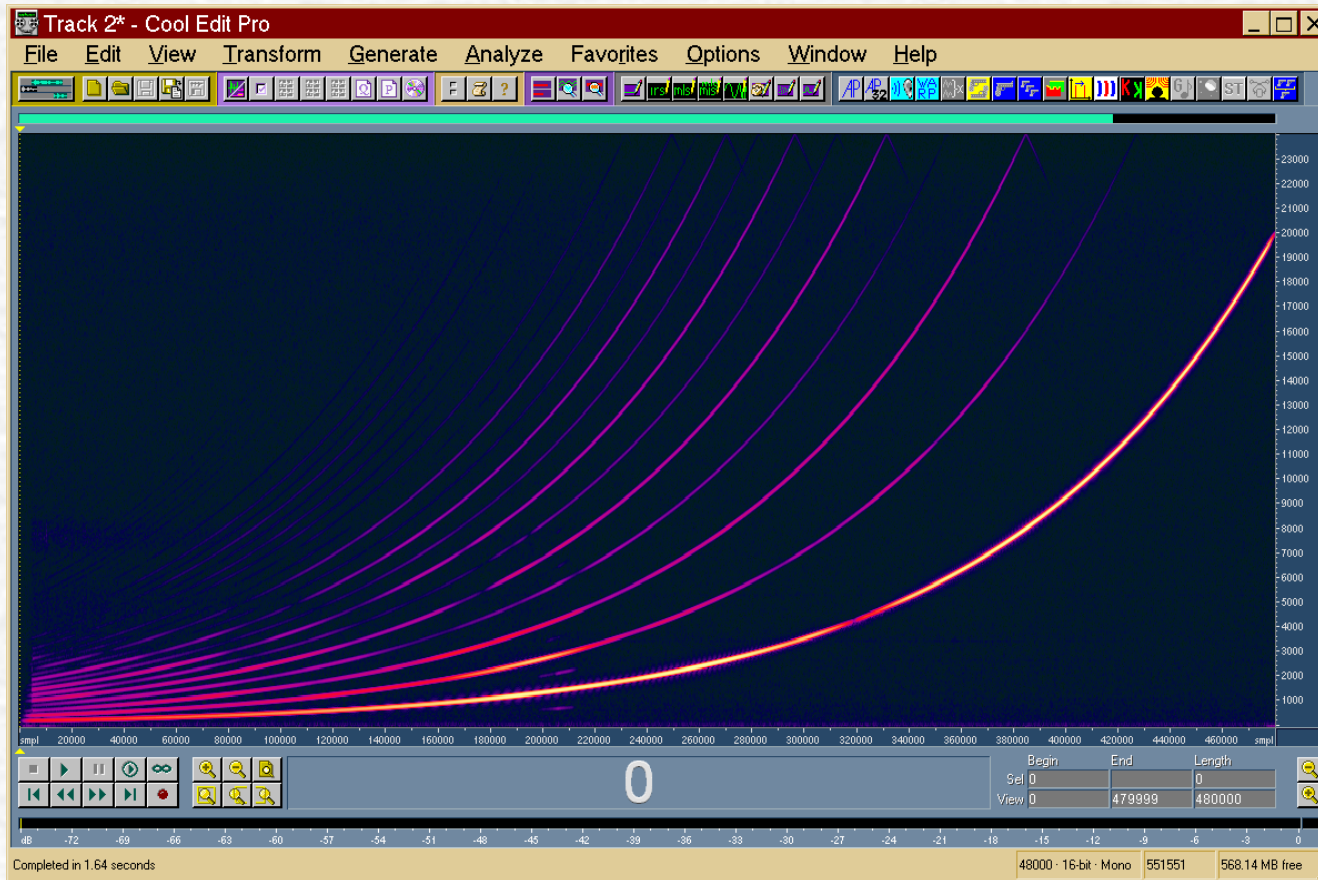
Inverse Filter $z(t)$



Test Signal – $x(t)$

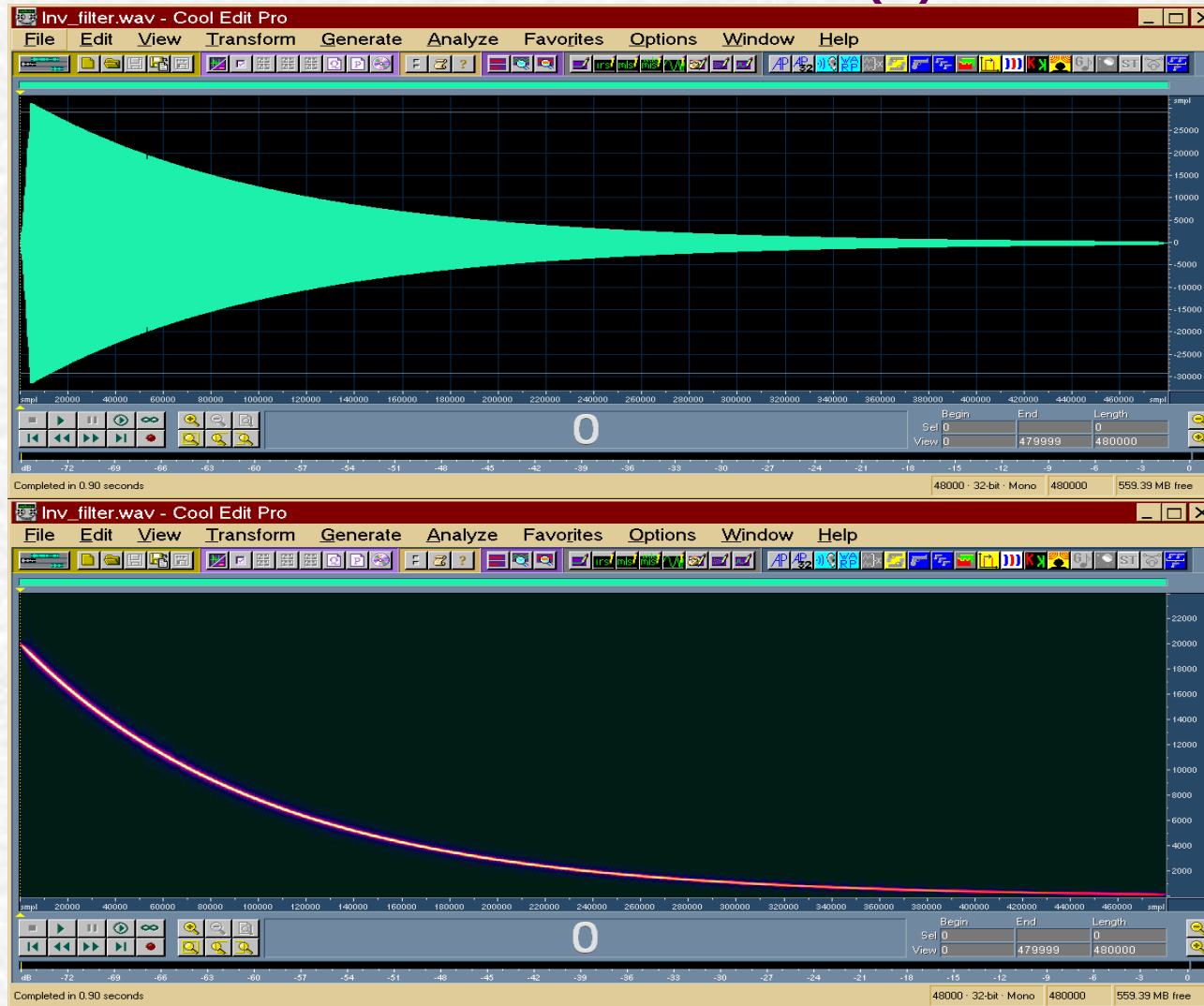


Measured signal - $y(t)$



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

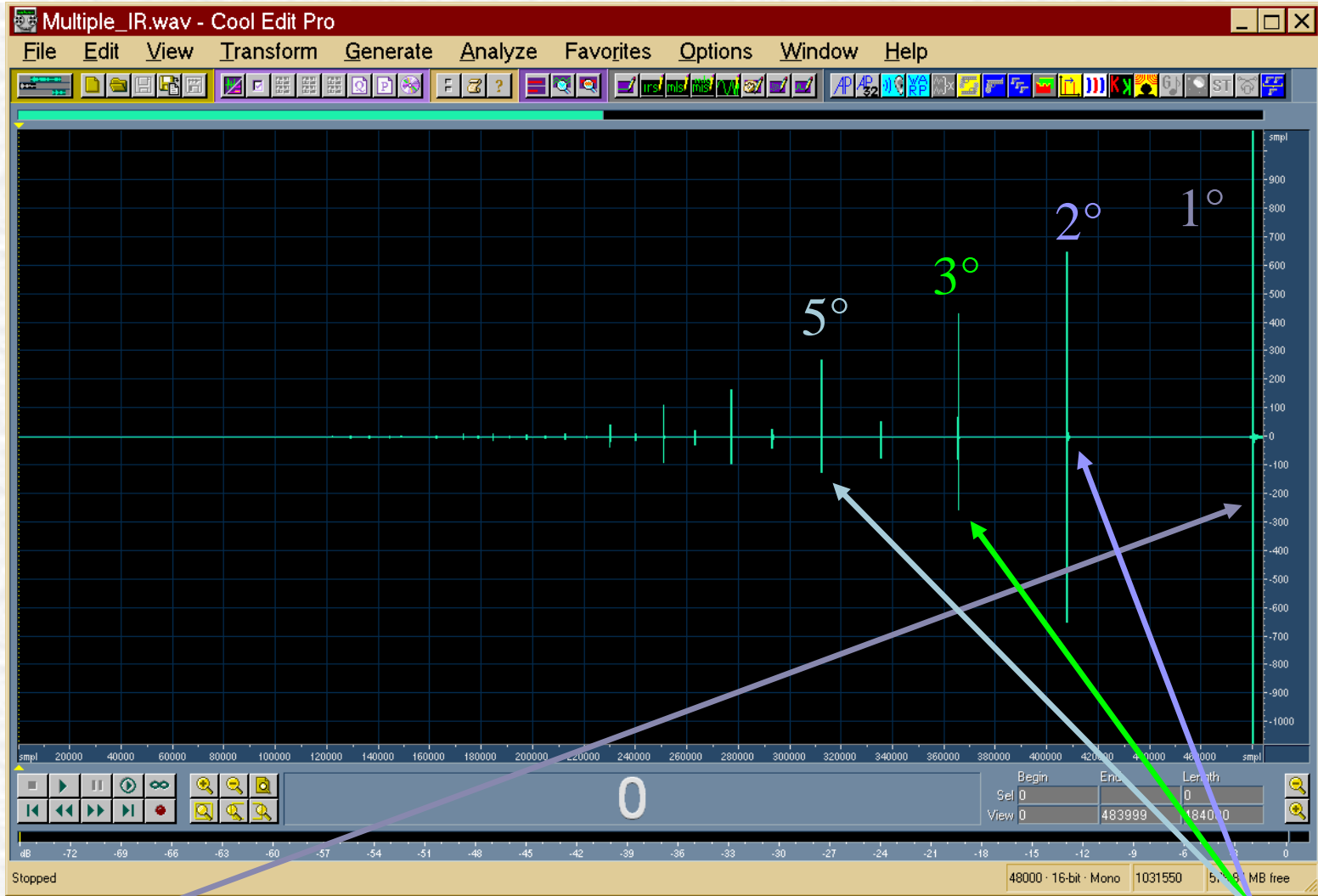
Inverse Filter – $z(t)$



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



Result of the deconvolution



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

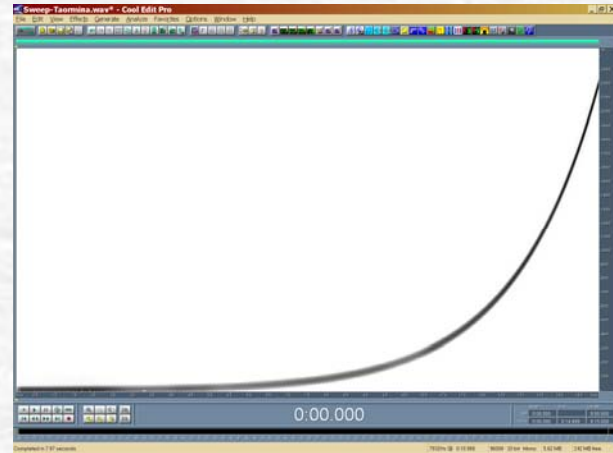
Measurement Setup

- The measurement method incorporates all the known techniques:
 - Binaural
 - B-format (1st 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
- Any multichannel auralization systems nowadays available is supported

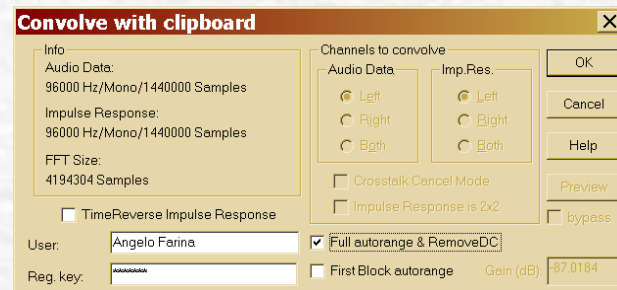
Measurement Parameters

- 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

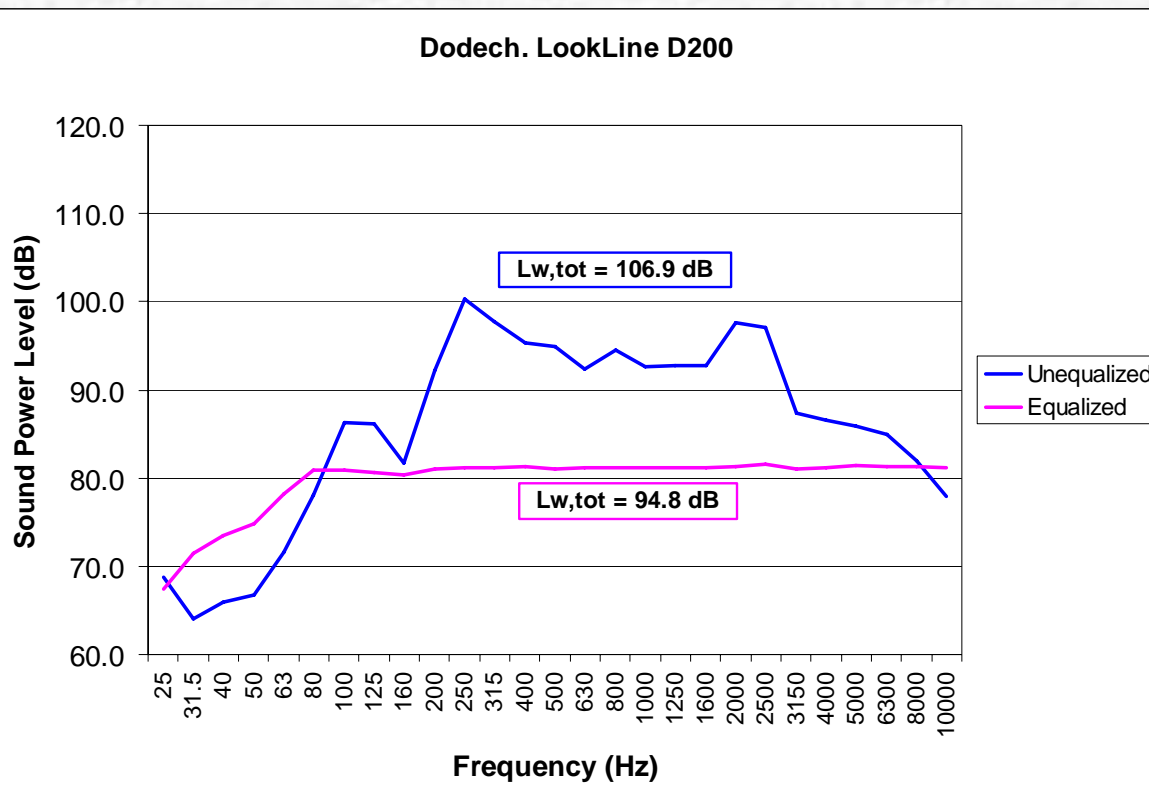


- Deconvolution:



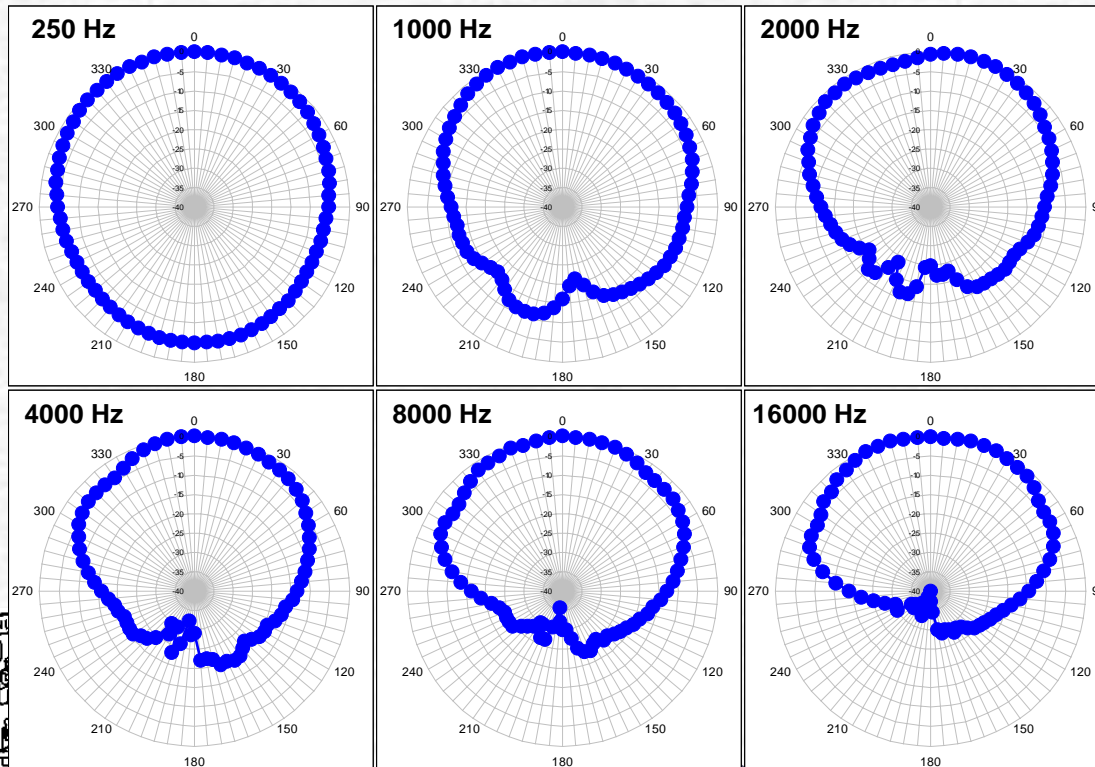
Transducers (sound source #1)

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



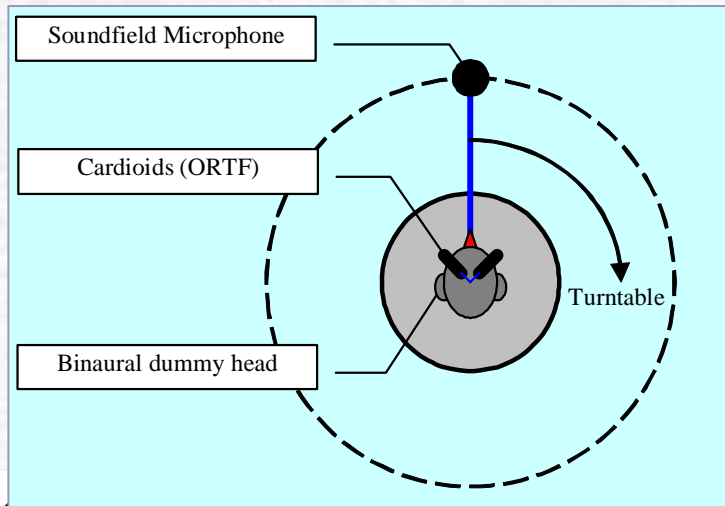
Transducers (sound source #2)

- Genelec S30D reference studio monitor:
 - Three-ways, active multi-amped, AES/EBU
 - Frequency range 37 Hz – 44 kHz (+/- 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

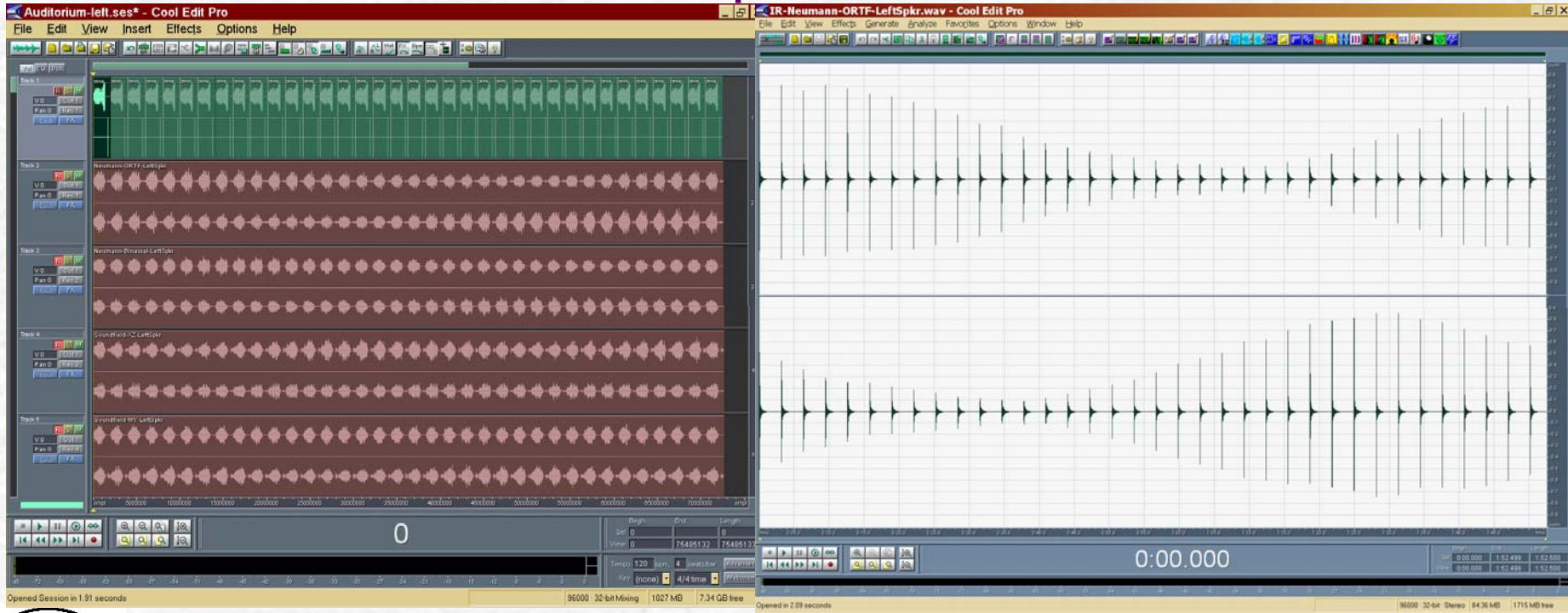


- Computer and sound card:
 - Signum Data Futureclient P-IV 1.8 GHz
 - Aardvark Pro Q-10 (8 ch., 96 kHz, 24 bits)



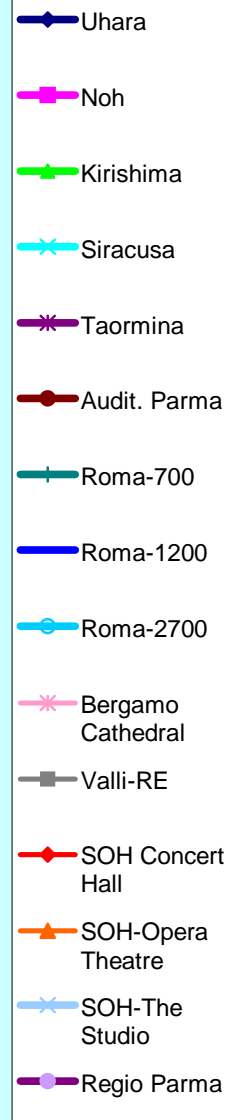
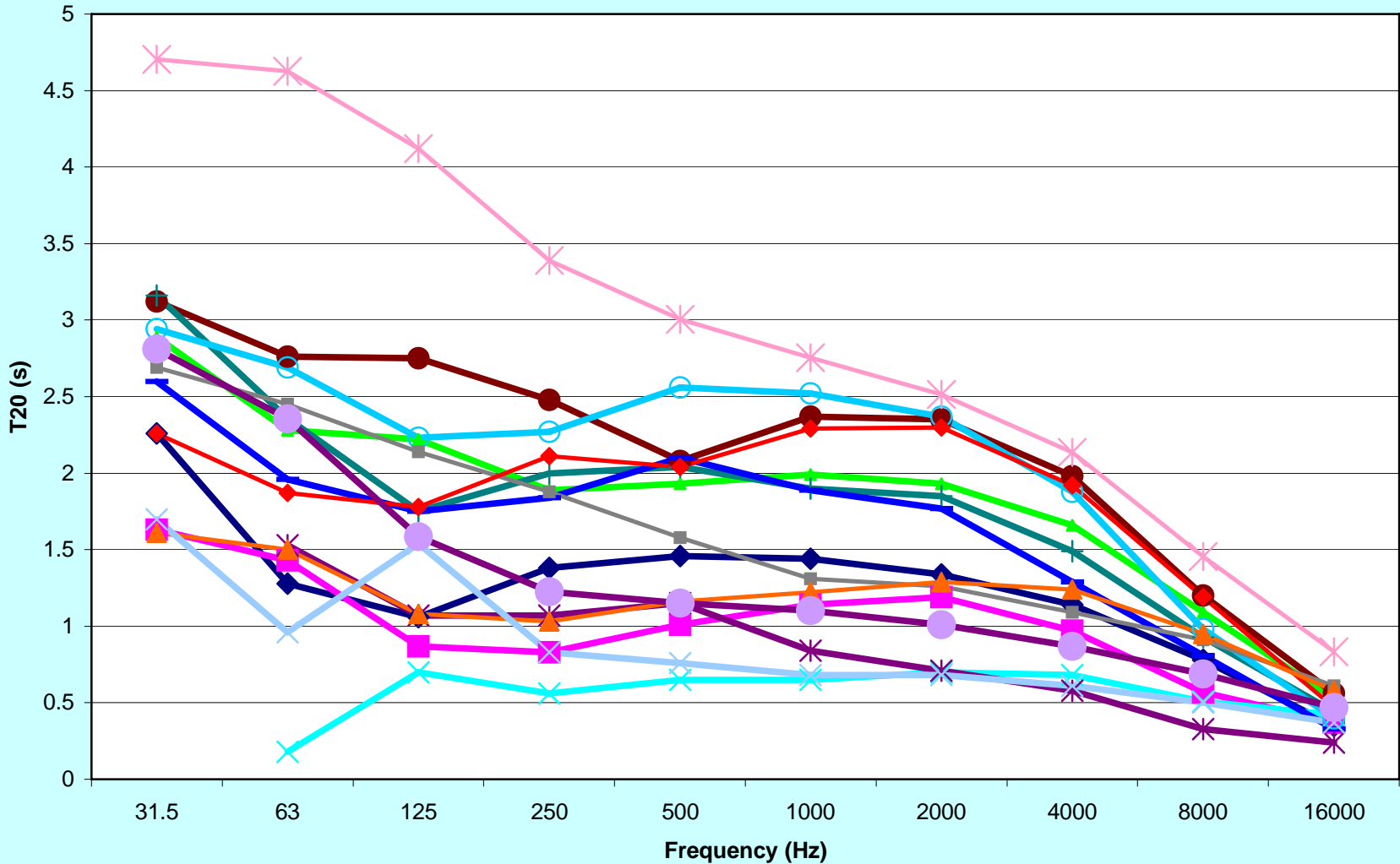
Measurement procedure

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

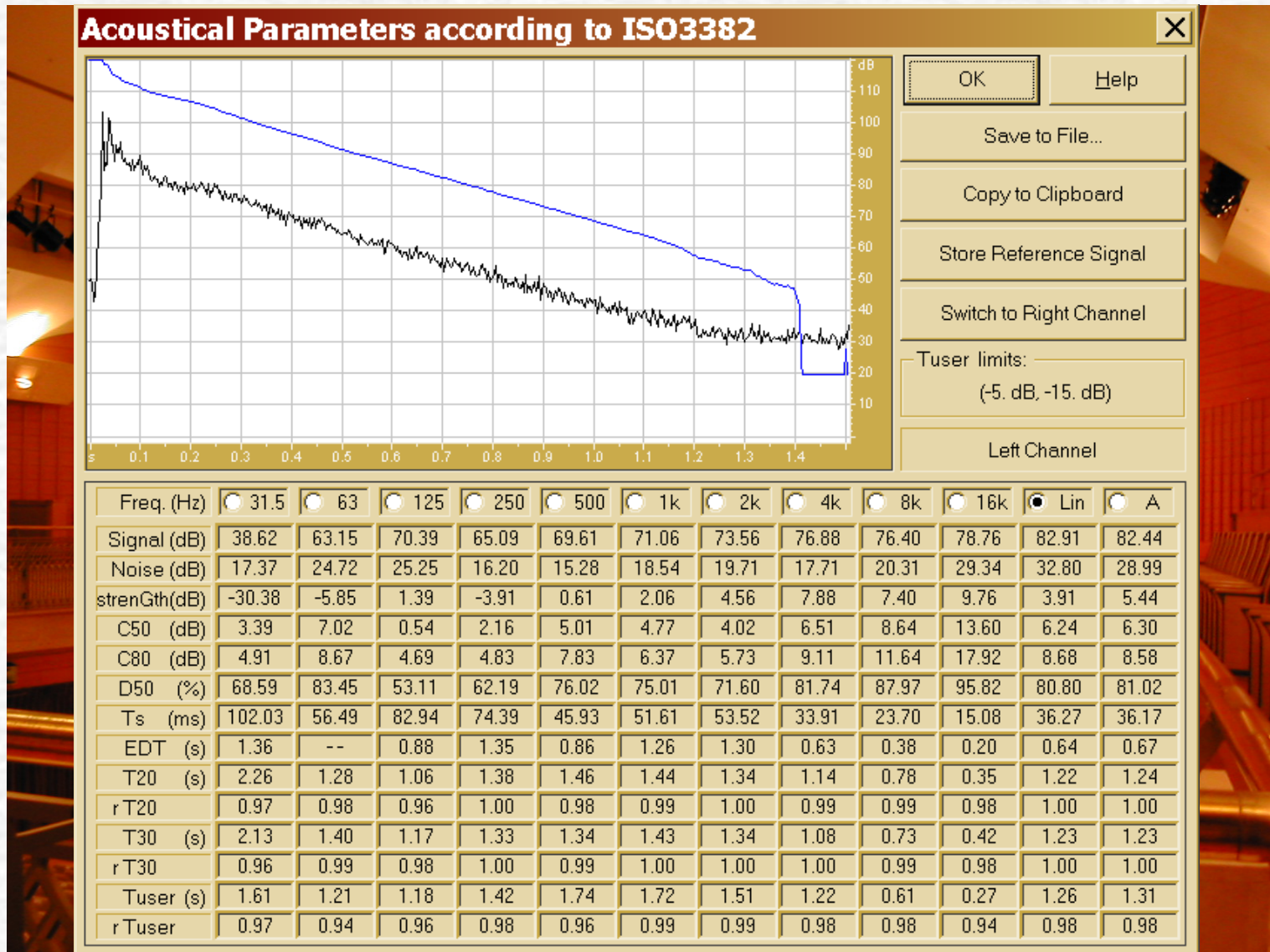


Theatres measured

Reverberation Time T20

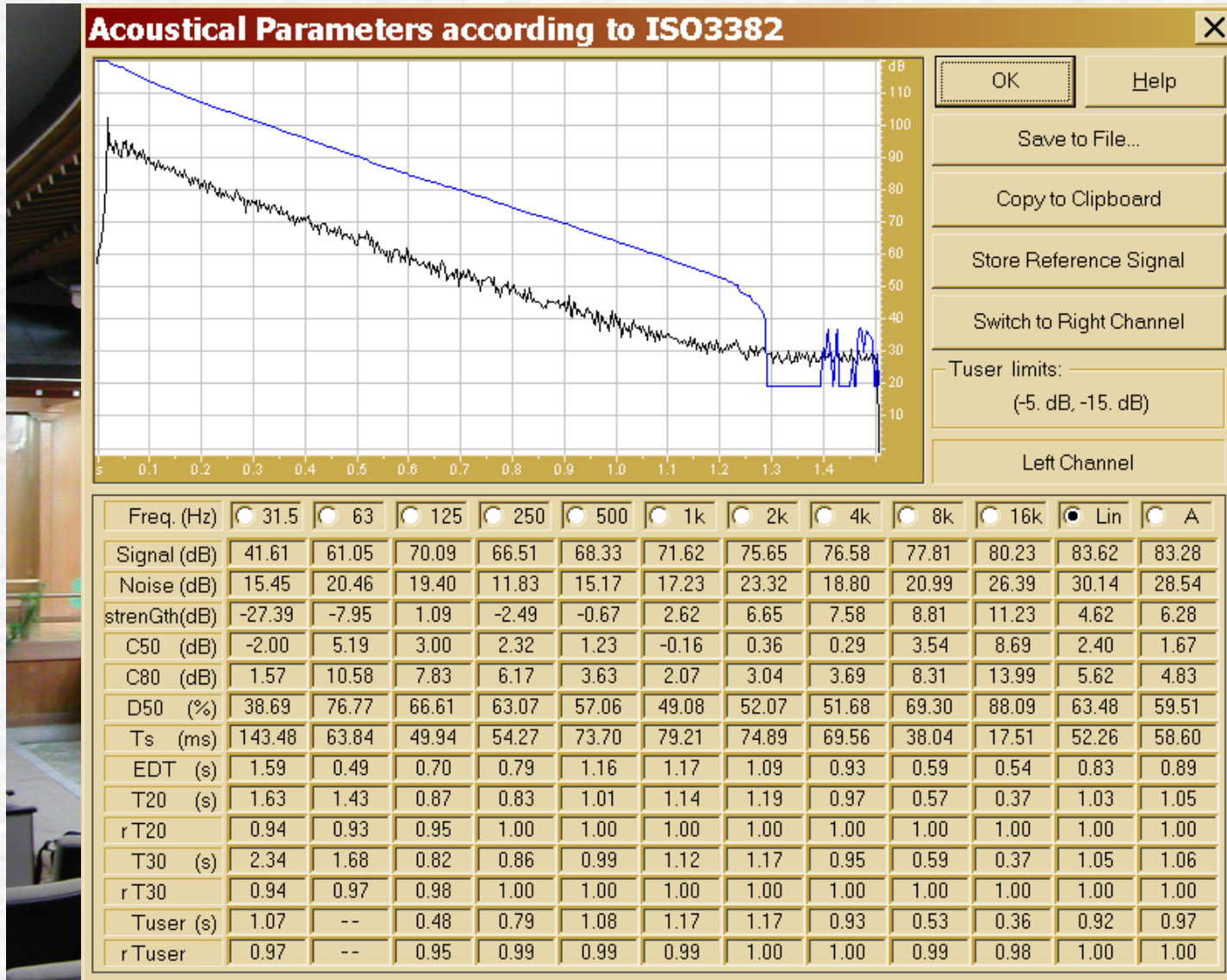


Uhara Hall, Kobe, Japan



$T_{20} = 1.44 \text{ s}$

Noh theater, Kobe, Japan



$T_{20} = 1.14 \text{ s}$



International Conference on "Room acoustics: Design and science"

Kirishima Concert Hall, Japan

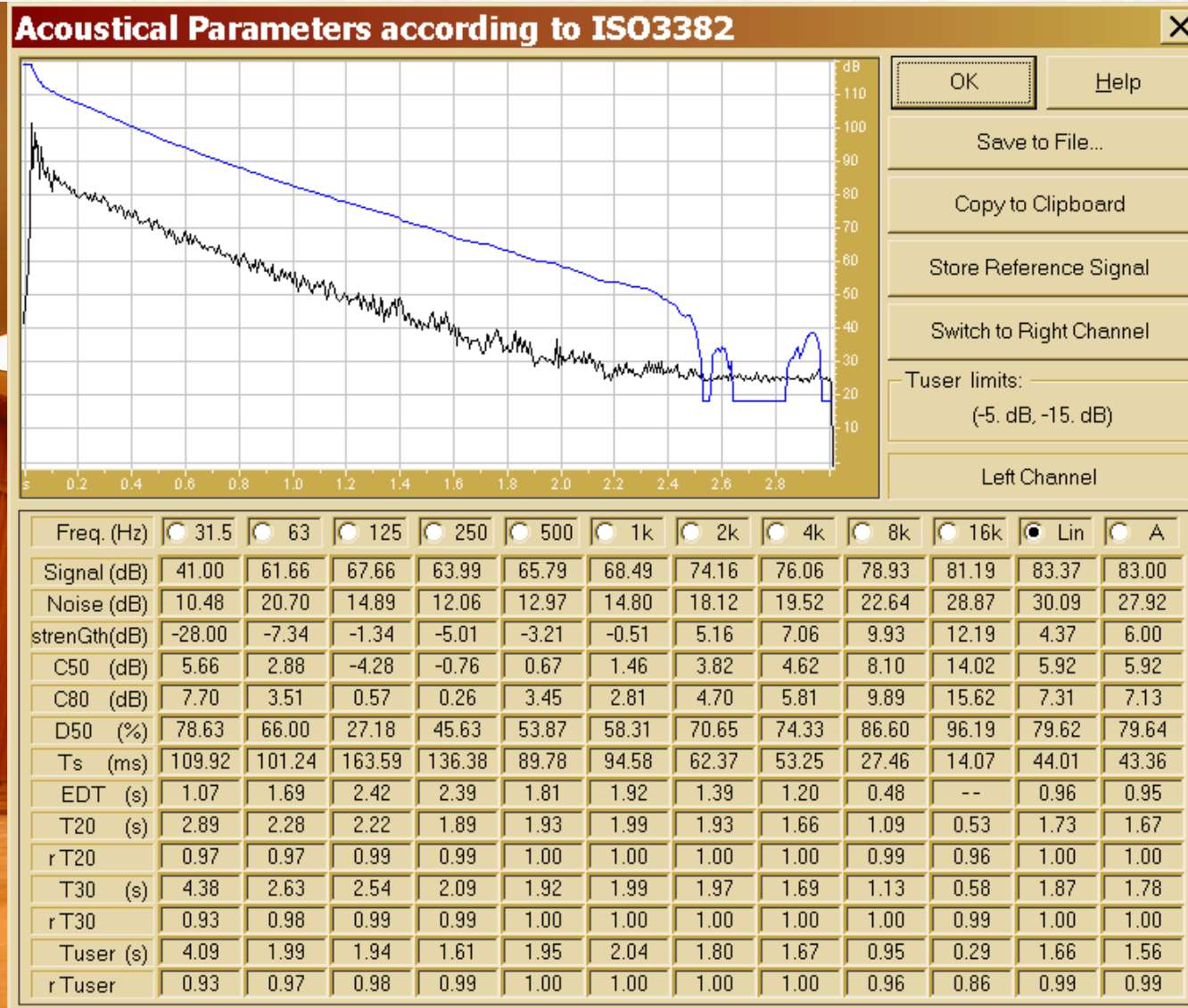


10-Sep-08

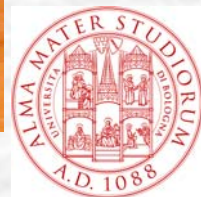
Farina, Tronchin, RADS



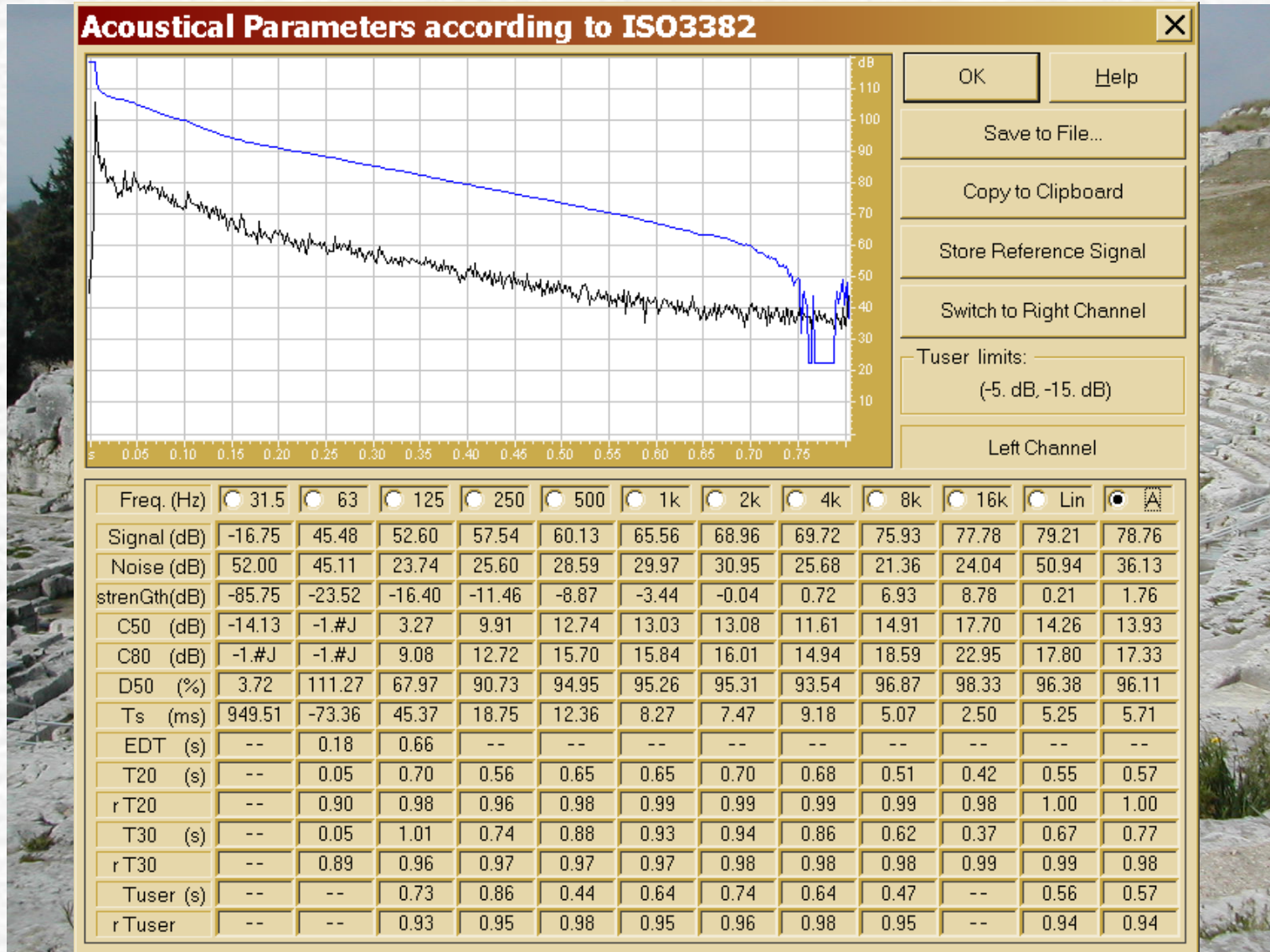
Kirishima Concert Hall, Japan



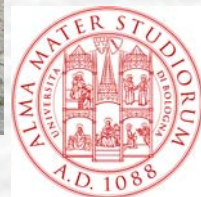
$T_{20} = 1.93 \text{ s}$



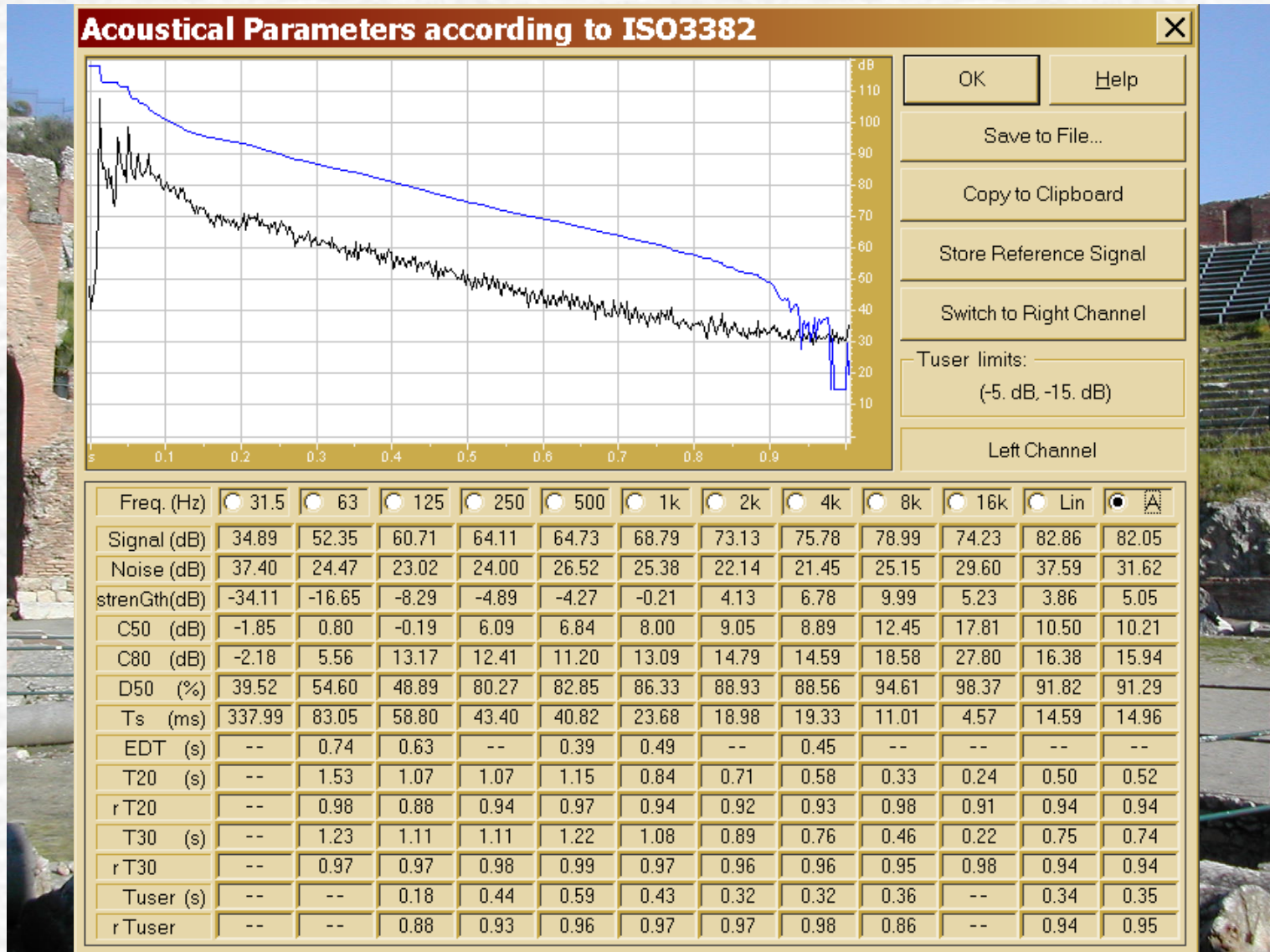
Greek Theater in Siracusa



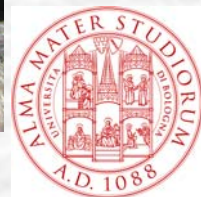
$T_{20} = 0.65 \text{ s}$



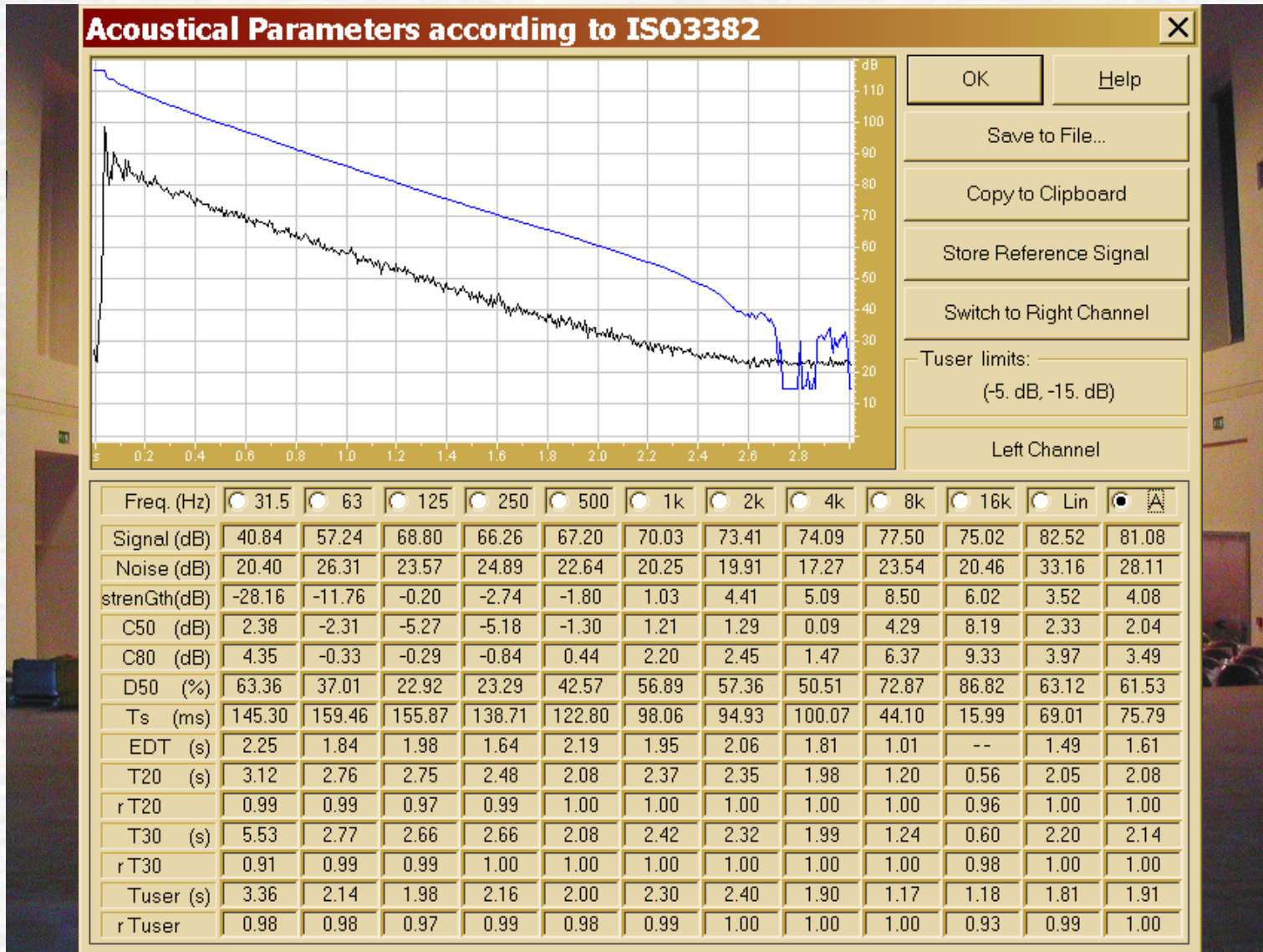
Roman Theater in Taormina



$T_{20} = 1.15 \text{ s}$



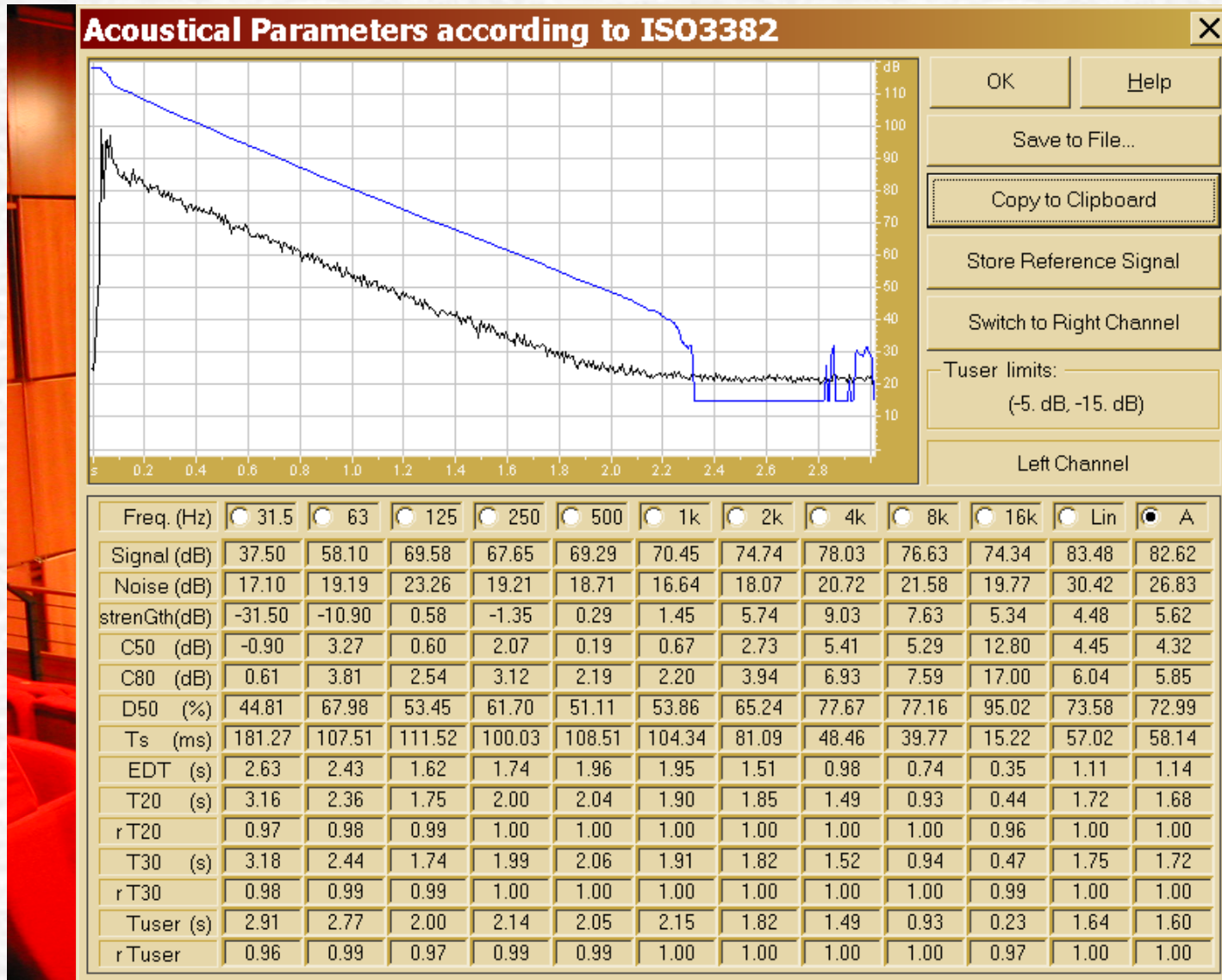
Parma Auditorium, Italy



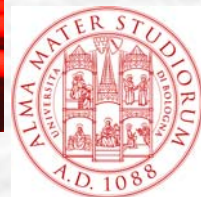
$T_{20} = 2.08 \text{ s}$



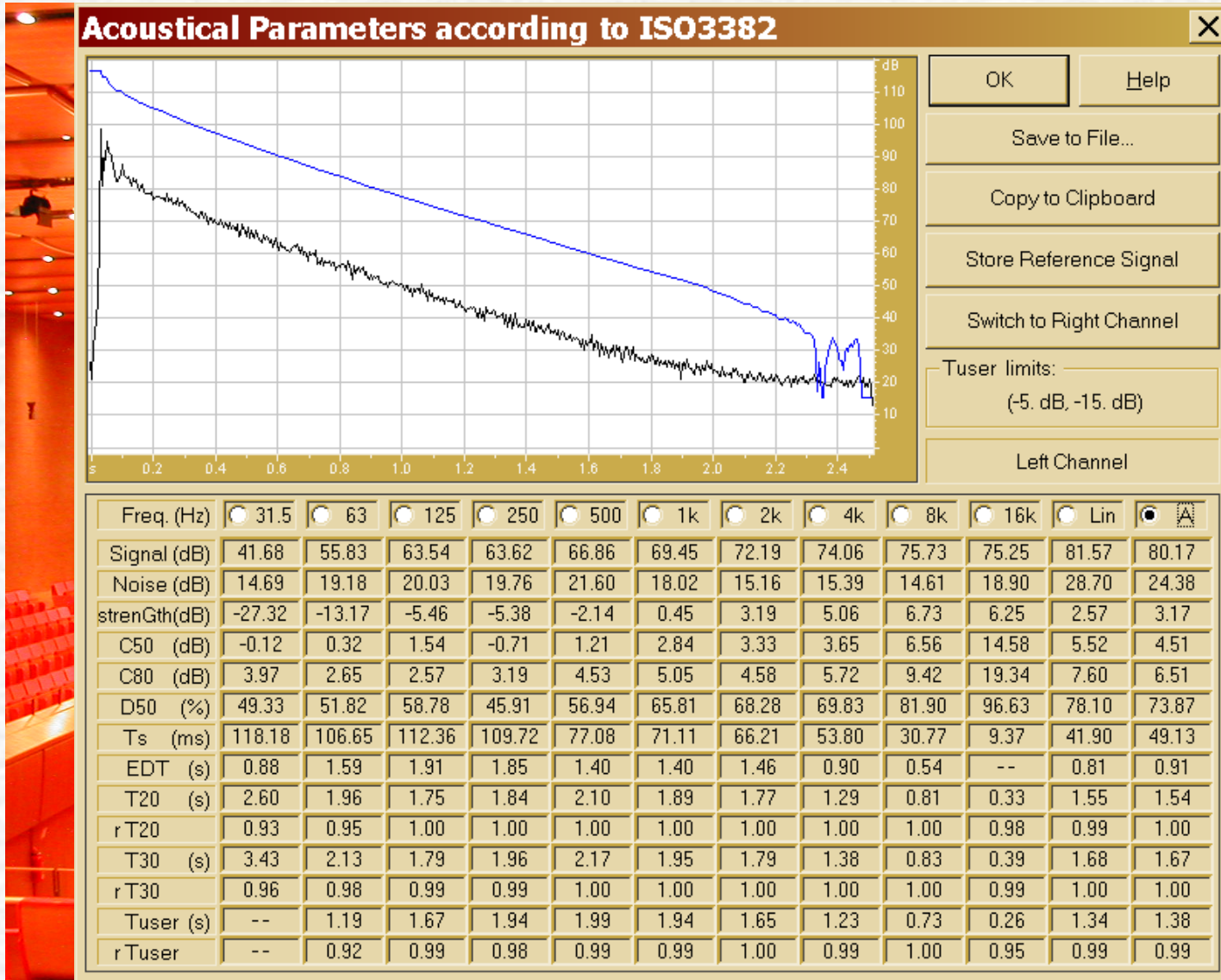
Rome Auditorium, 700 seats



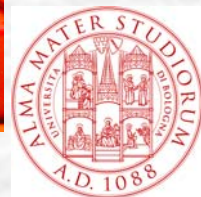
$T_{20} = 2.04 \text{ s}$



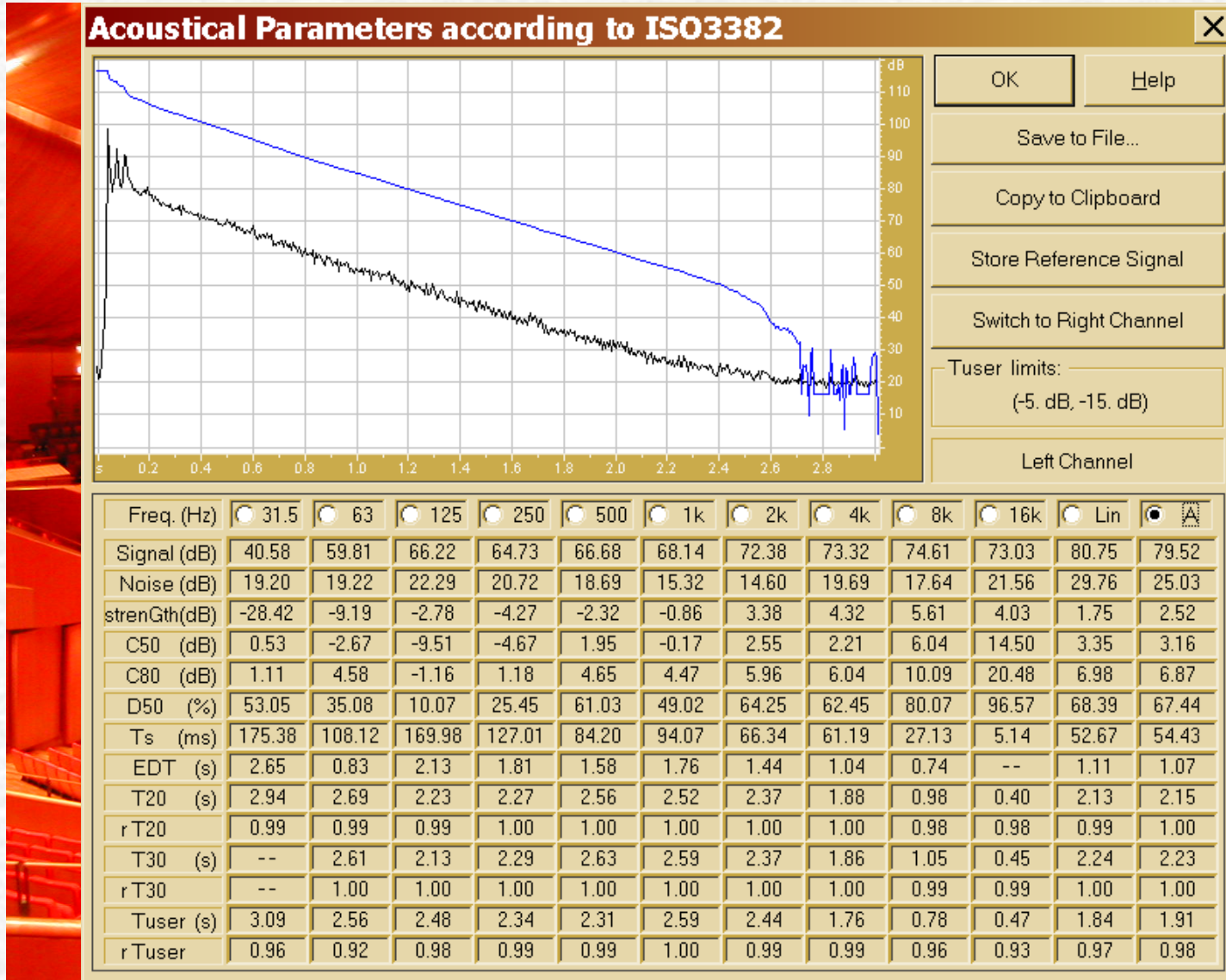
Rome Auditorium, 1200 seats



$T_{20} = 2.10 \text{ s}$



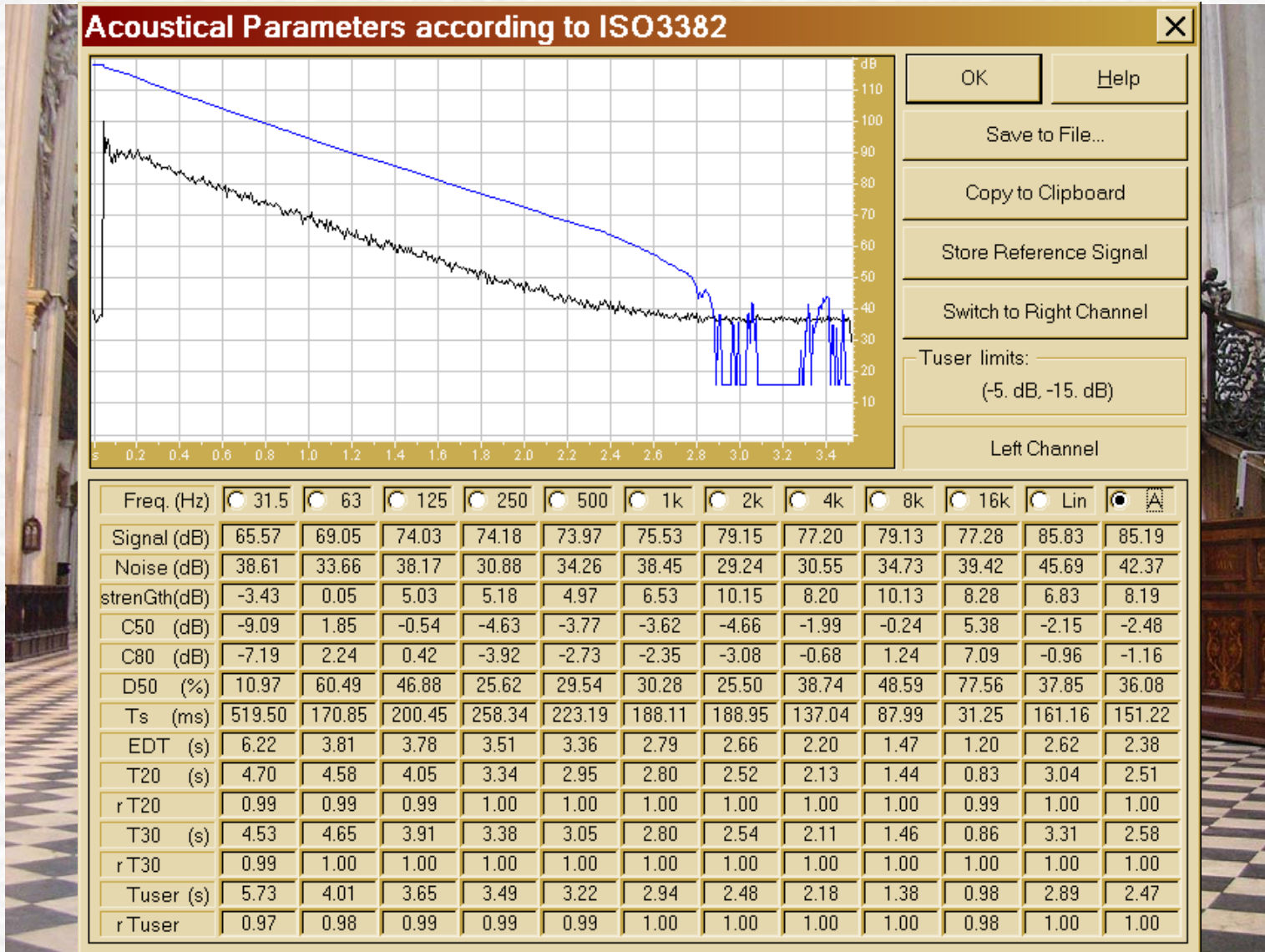
Rome Auditorium, 2700 seats



$T_{20} = 2.56 \text{ s}$



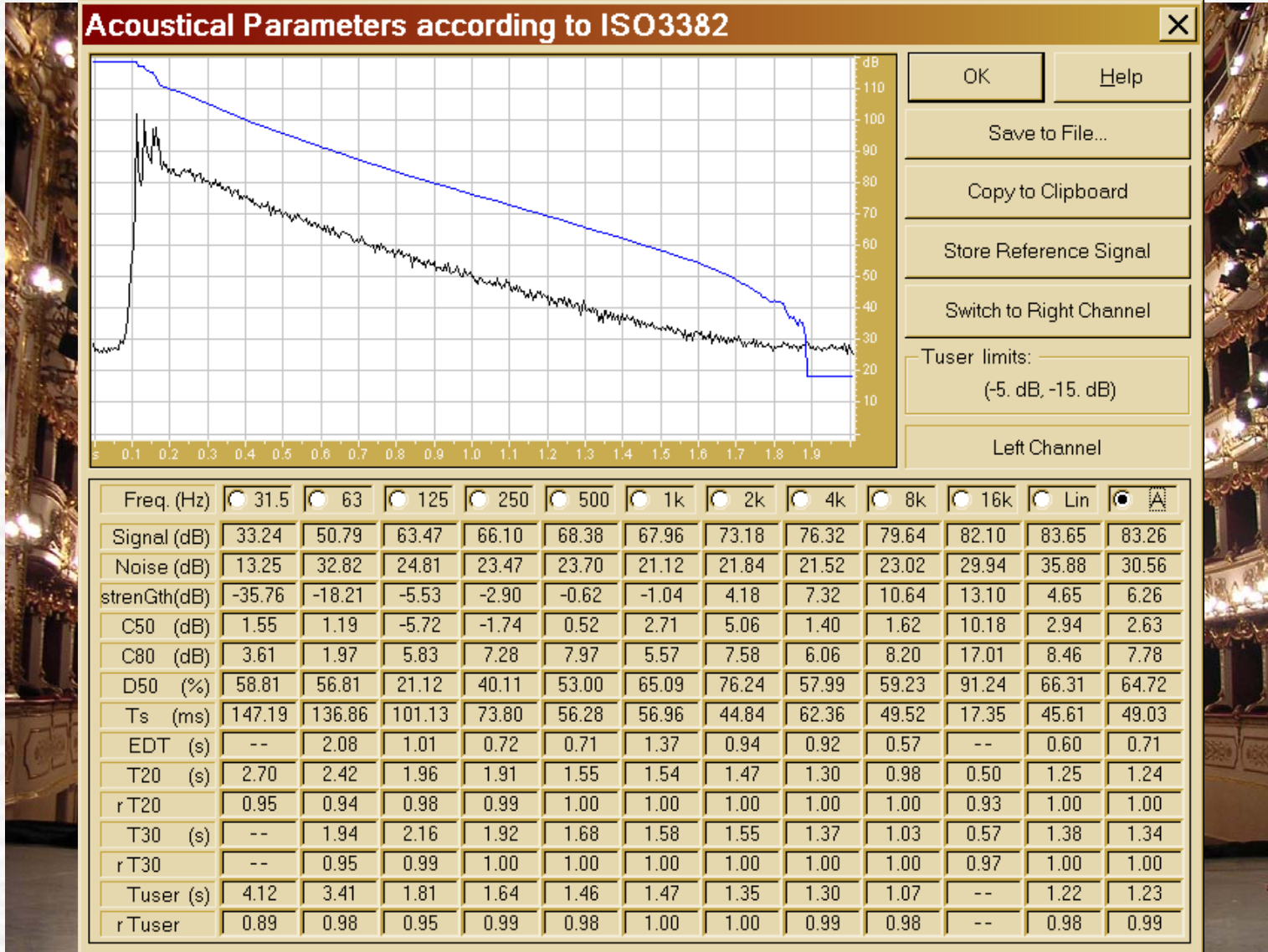
Bergamo's Cathedral, Italy



$T_{20} = 2.95 \text{ s}$



Teatro Valli, Reggio Emilia, Italy



$T_{20} = 1.55 \text{ s}$





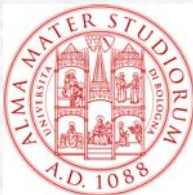
International Conference on "Room acoustics: Design and science"

Sydney Opera House

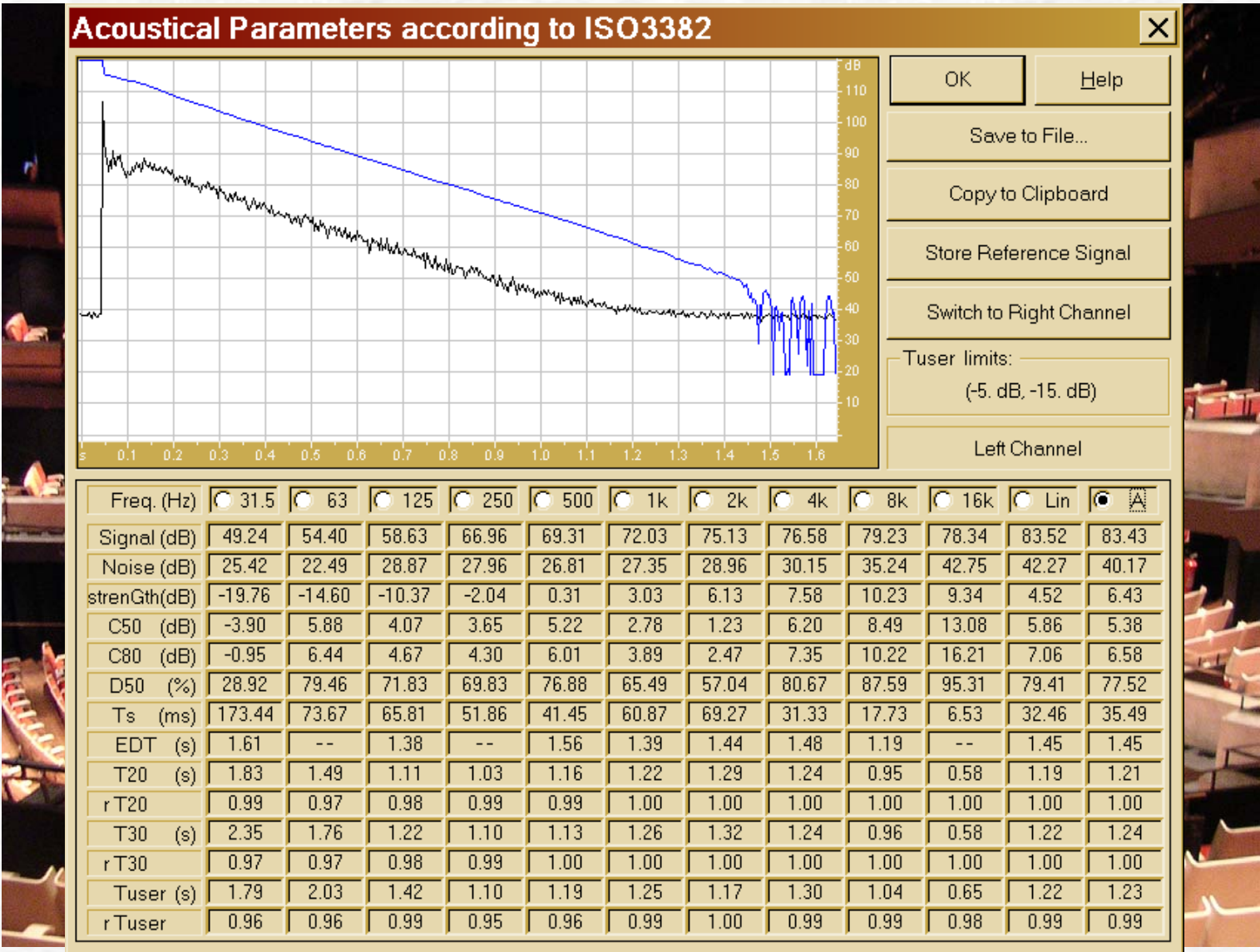


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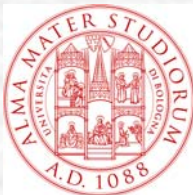
Farina, Tronchin, RADS



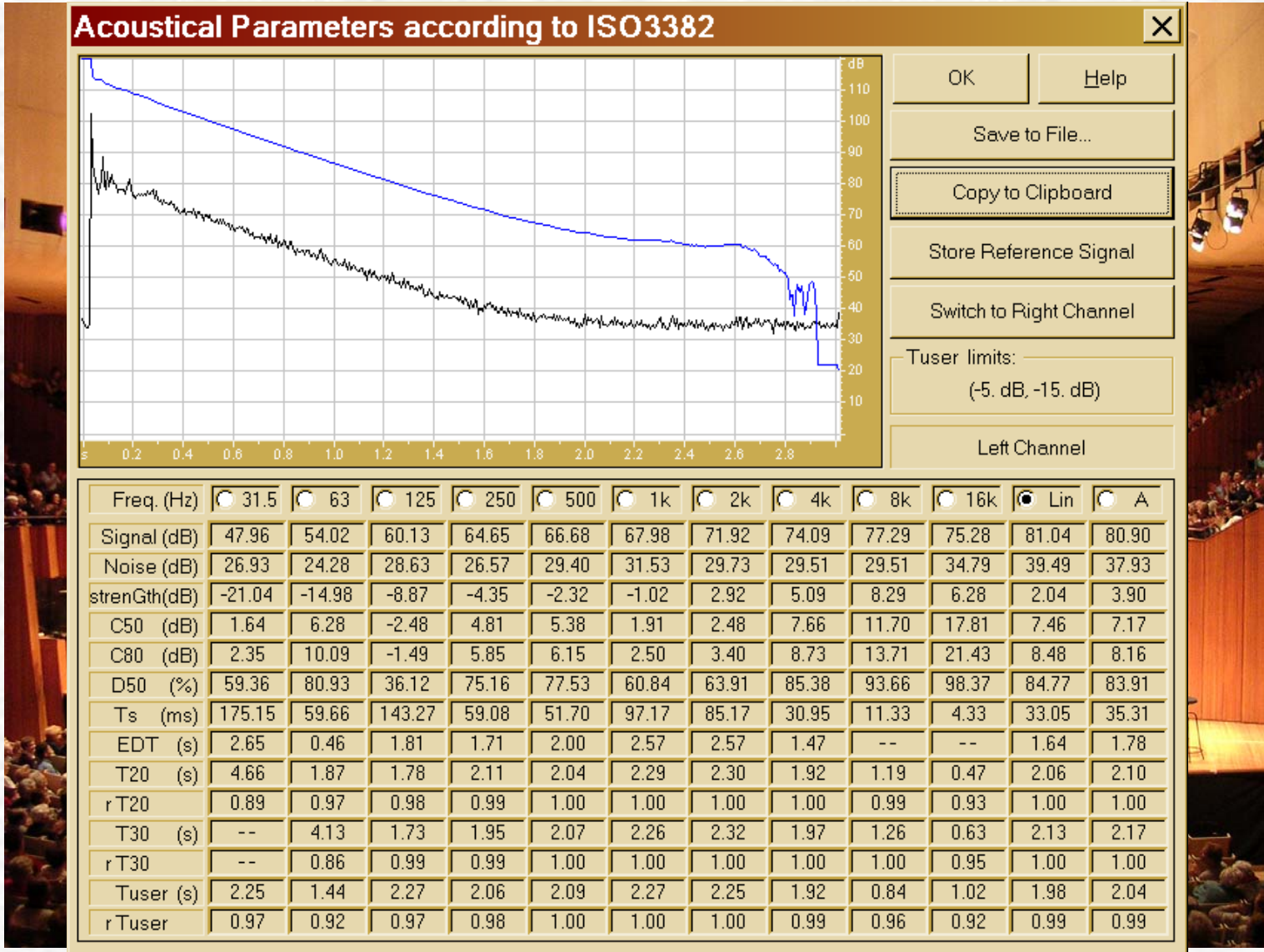
Sydney Opera House – opera theatre



$T_{20} = 1.16 \text{ s}$

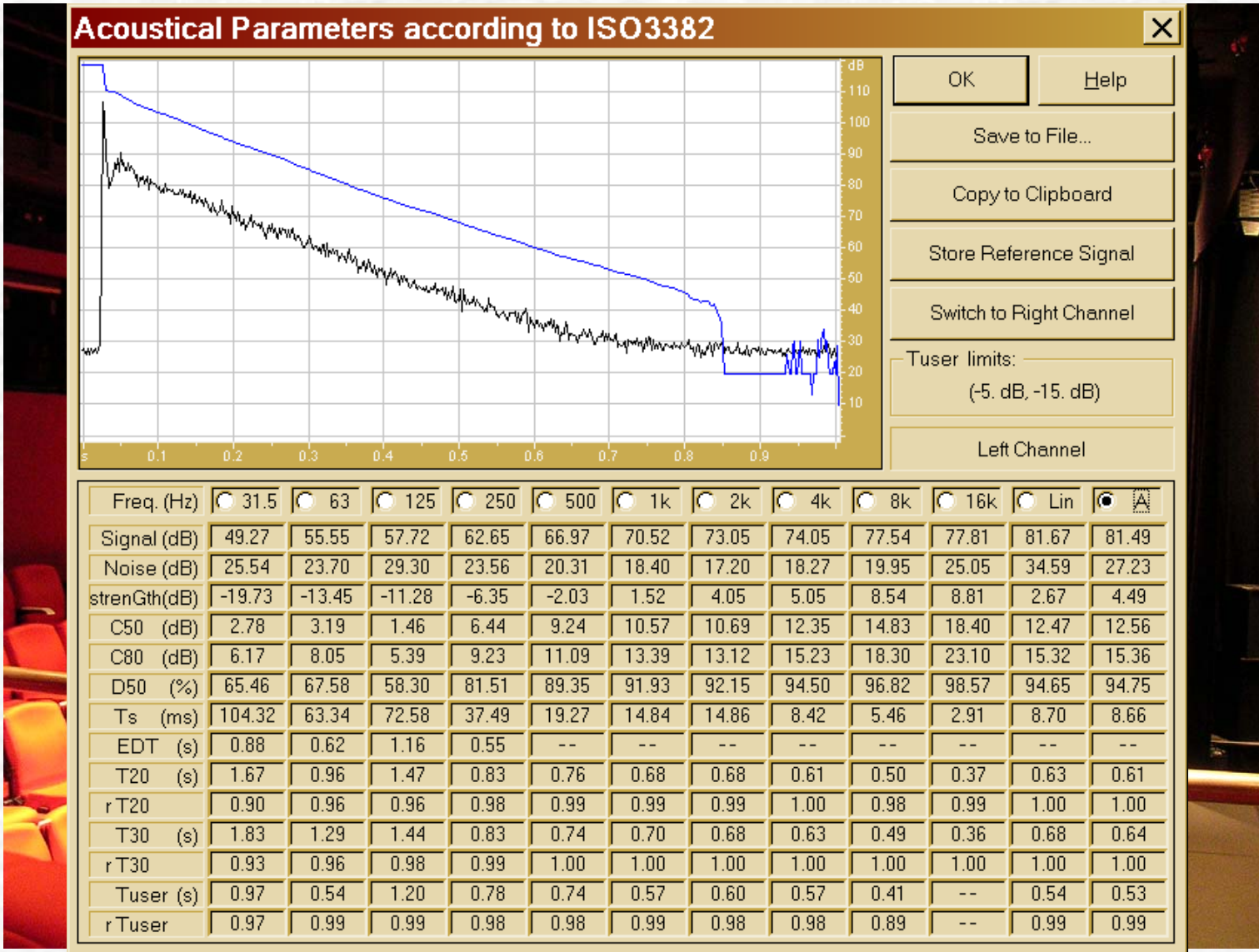


Sydney Opera House – concert hall



$T_{20} = 2.04 \text{ s}$

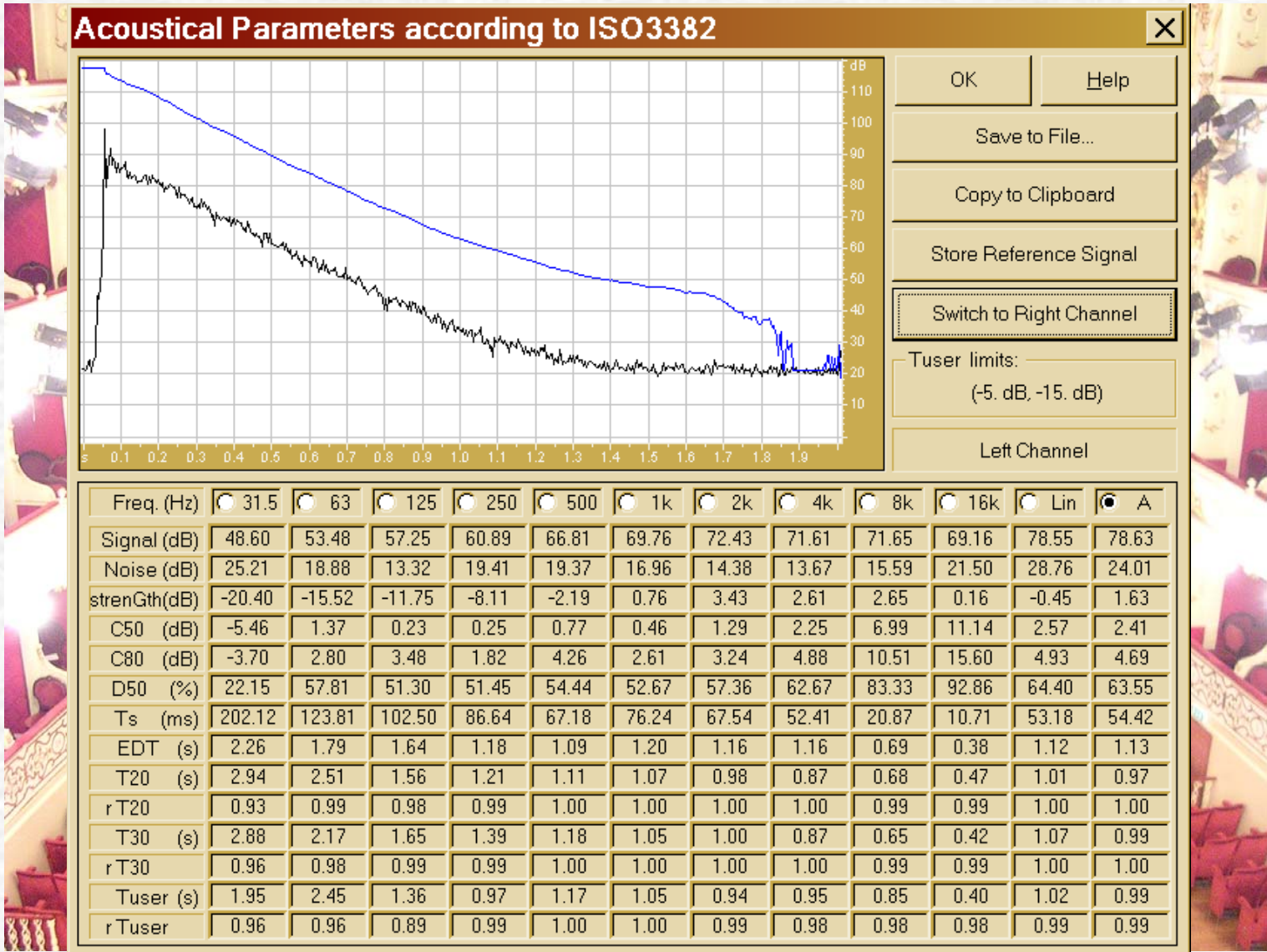
Sydney Opera House – the studio



$T_{20} = 0.76 \text{ s}$



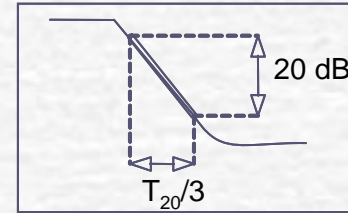
Teatro Regio in Parma (Italy)



$T_{20} = 1.11 \text{ s}$

Acoustical Parameters (ISO 3382)

- 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 (ISO 3382)

- **Strength:** $G = \text{SPL} - L_W + 31 \quad \text{dB}$

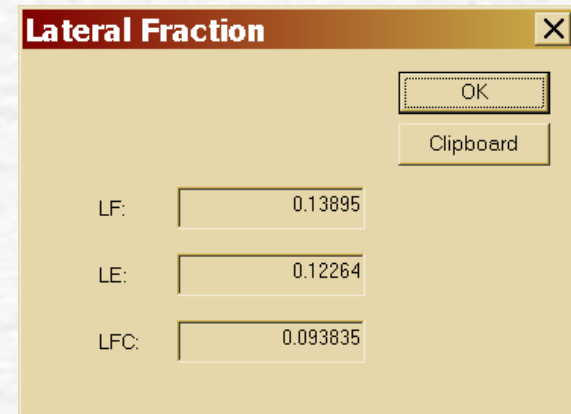
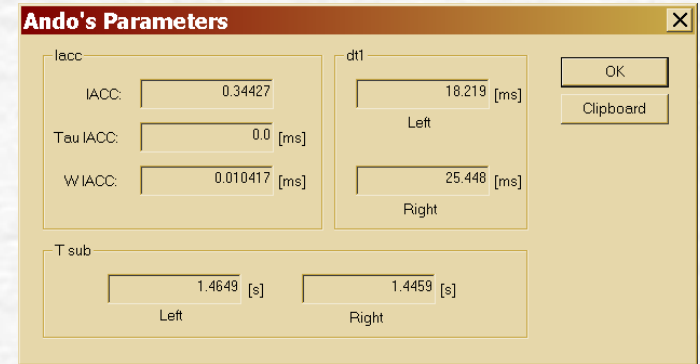
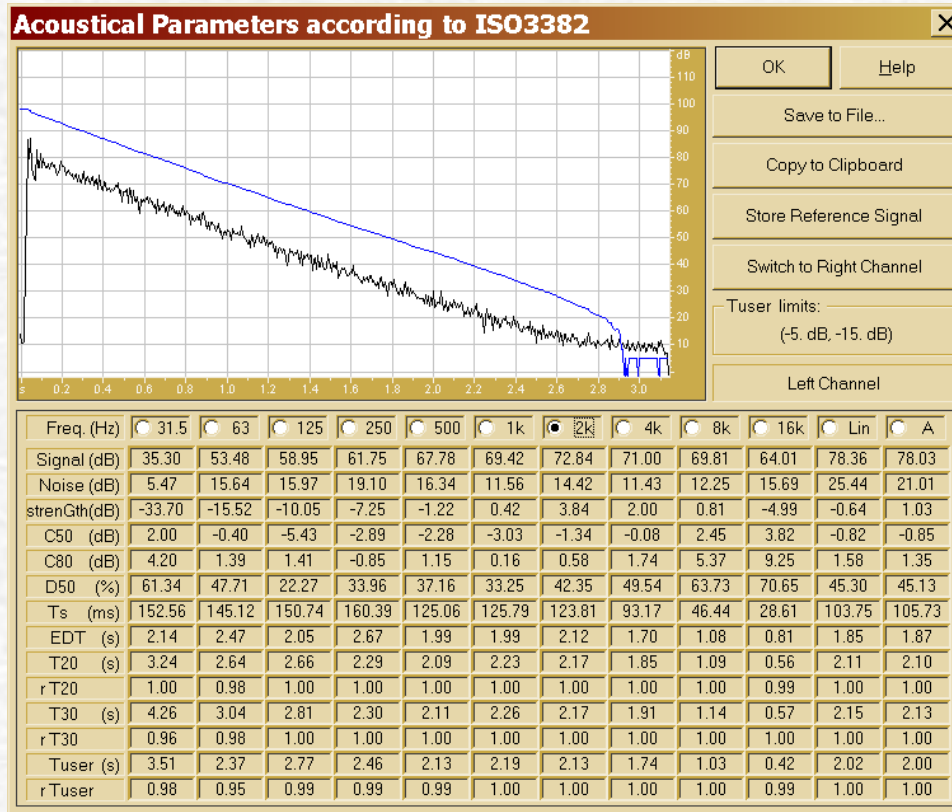
- **IACC:**
$$\rho(\tau) = \frac{\int_{-\infty}^{\infty} h_d(\tau) \cdot h_s(\tau+t) \cdot dt}{\sqrt{\int_{-\infty}^{\infty} h_d^2(\tau) \cdot d\tau \cdot \int_{-\infty}^{\infty} h_s^2(\tau+t) \cdot dt}}$$

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

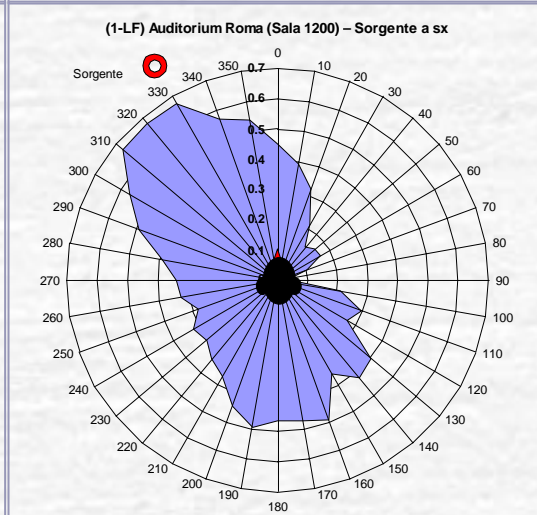
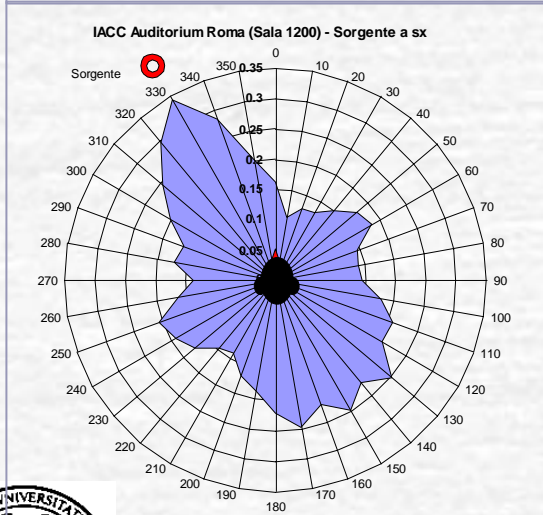
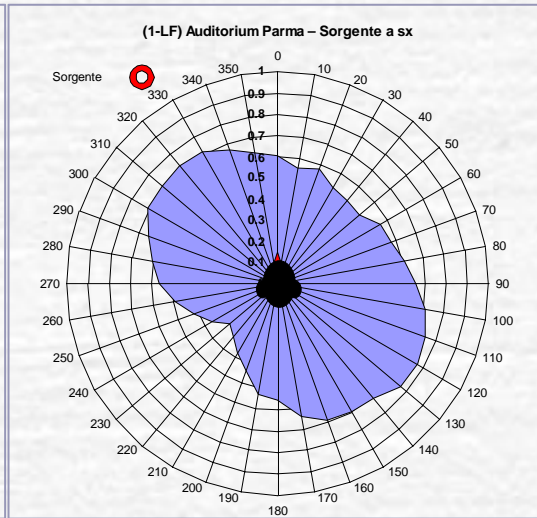
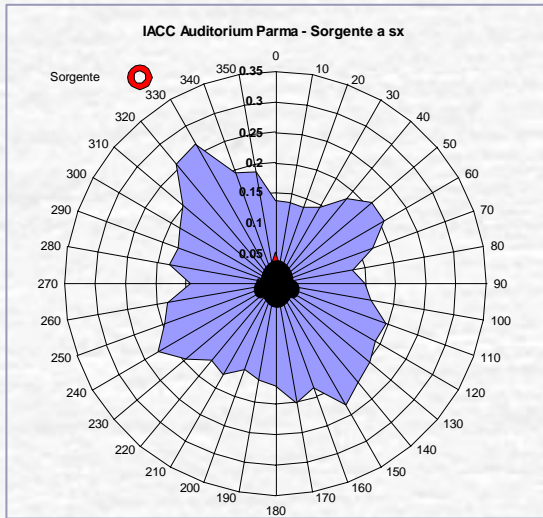
- **LFC:**
$$LFC = \frac{\int_{0ms}^{80ms} h_Y(\tau) \cdot h_W(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_W^2(\tau) \cdot d\tau}$$



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).

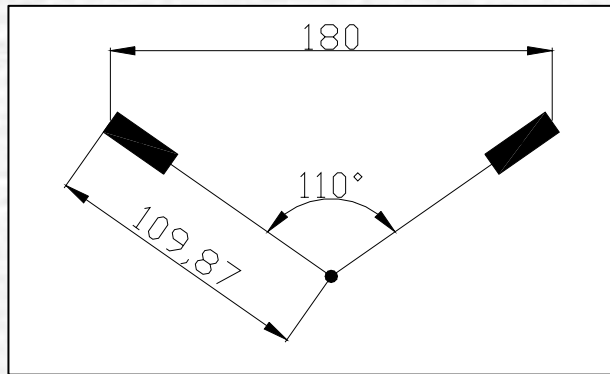


Auralization types

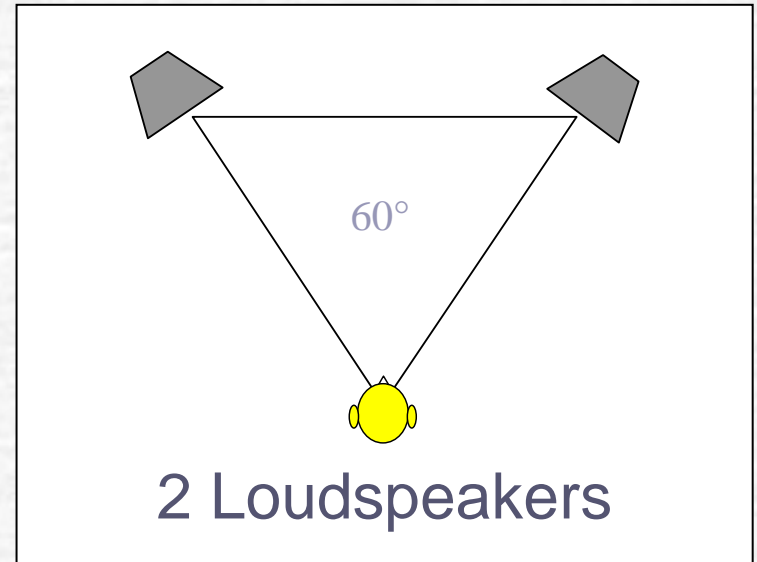
- Stereo (ORTF on 2 standard loudspeakers at $\pm 30^\circ$)
- Rotation-tracking reproduction on headphones (Binaural Room Scanning)
- Full 3D Ambisonics 1st order (decoding the B-format signal)
- ITU 5.1 (from different 5-mikes layouts)
- 2D Ambisonics 3rd order (from Mark Poletti's circular array microphone)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



ORTF Stereo



2 Microphones

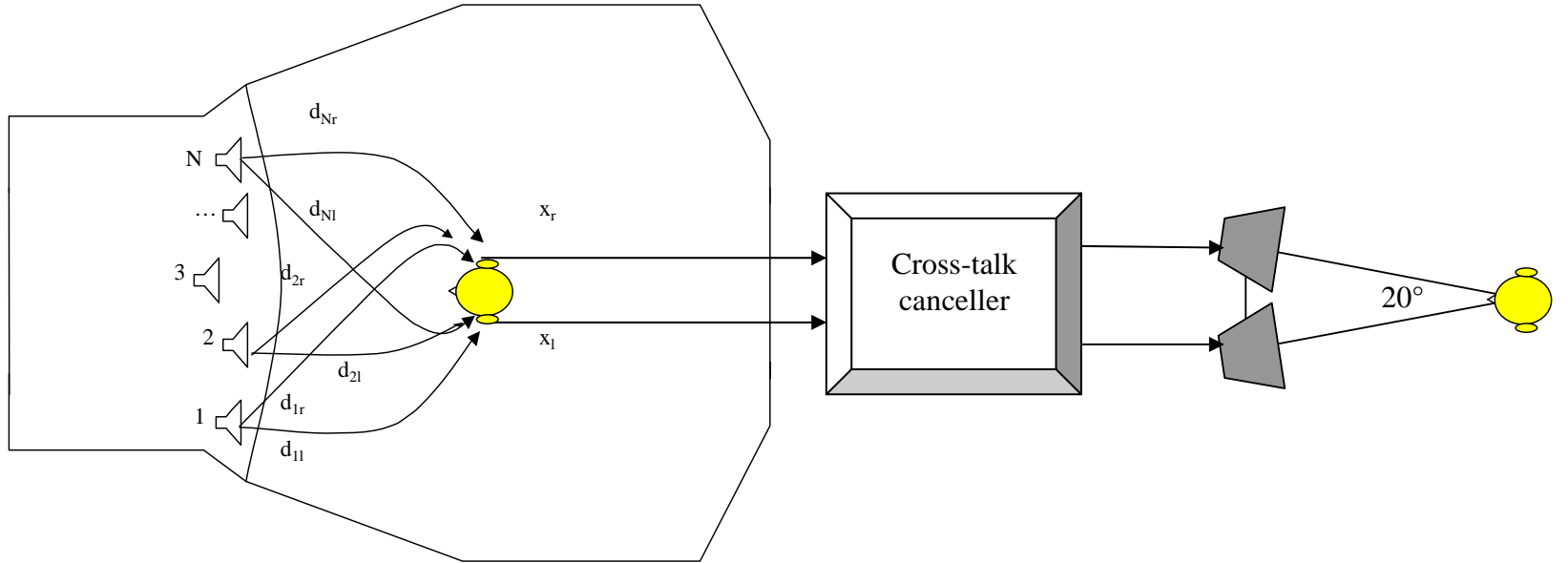


2 Loudspeakers

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

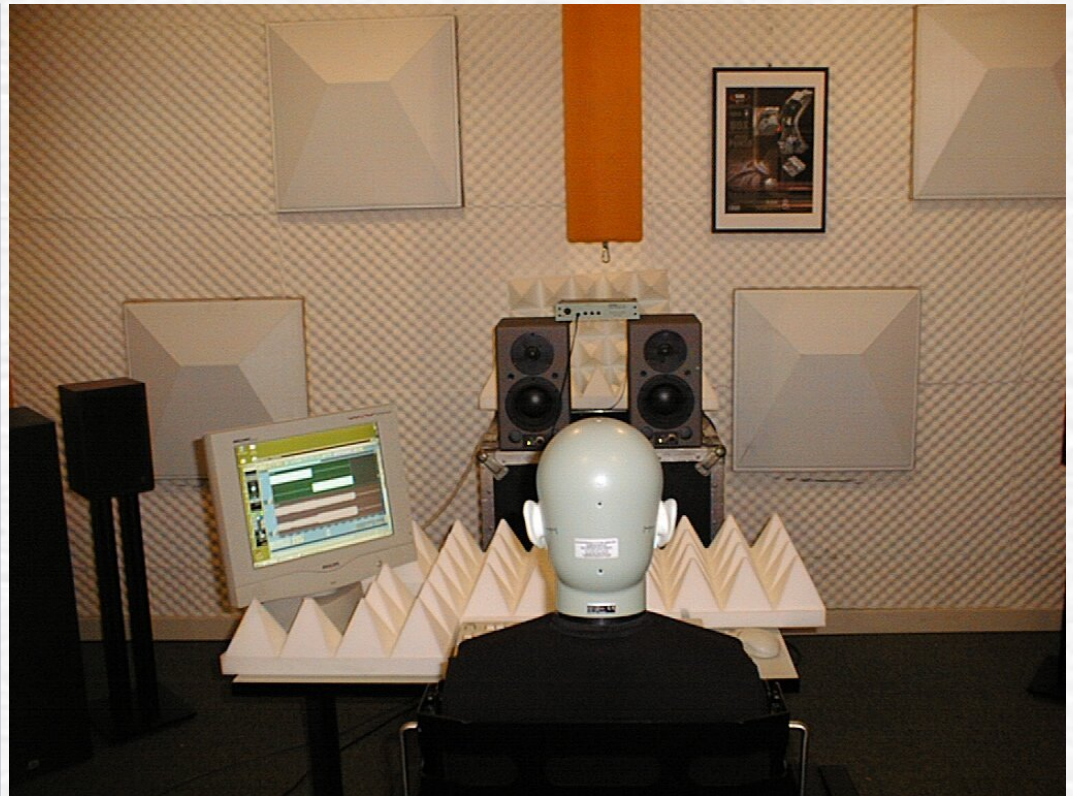
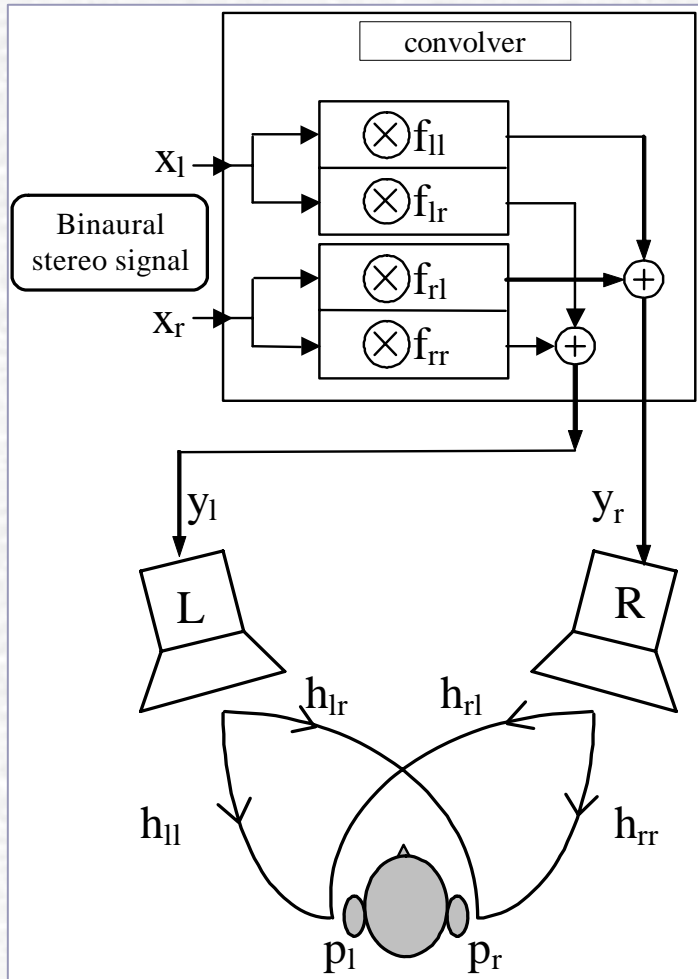
Binaural (Stereo Dipole)

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

Binaural (Stereo Dipole#2)

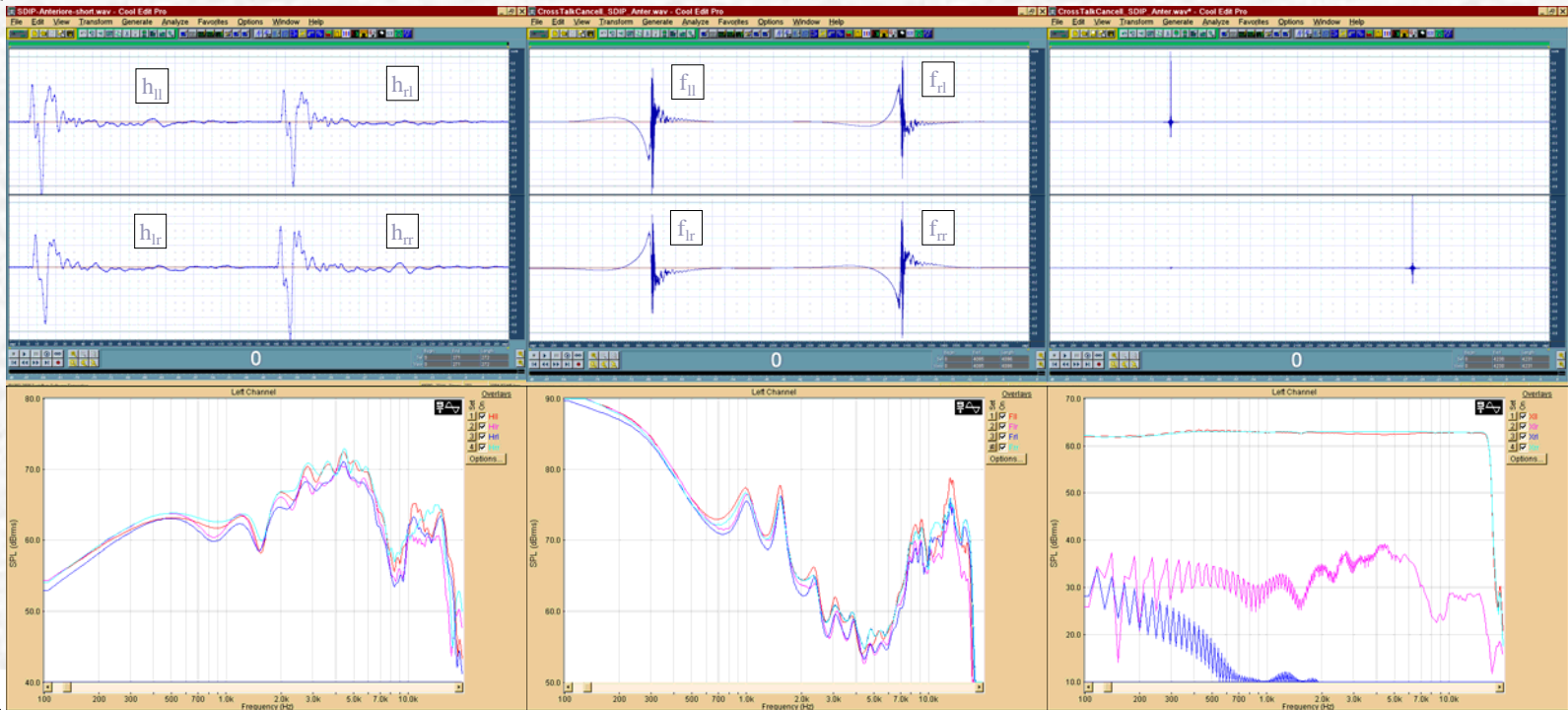


Binaural (Stereo Dipole#3)

$$\begin{cases} f_{ll} = (h_{rr}) \otimes \text{InvDen} \\ f_{lr} = (-h_{lr}) \otimes \text{InvDen} \\ f_{rl} = (-h_{rl}) \otimes \text{InvDen} \\ f_{rr} = (h_{ll}) \otimes \text{InvDen} \\ \text{InvDen} = \text{InvFilter}(h_{ll} \otimes h_{rr} - h_{lr} \otimes h_{rl}) \end{cases}$$

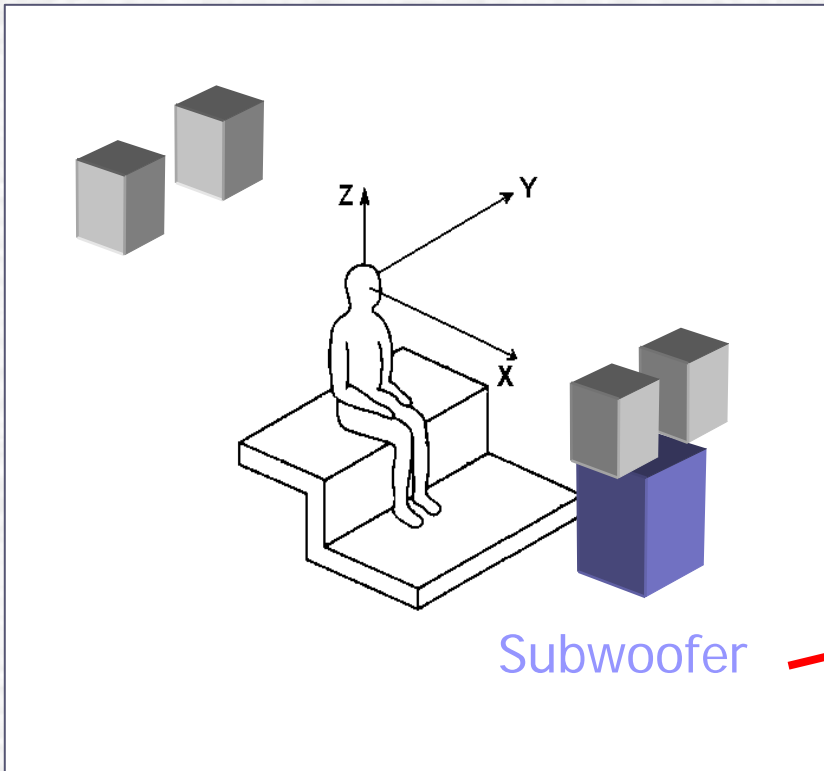
$$C(\omega) = \text{FFT}(h_{ll}) \cdot \text{FFT}(h_{rr}) - \text{FFT}(h_{lr}) \cdot \text{FFT}(h_{rl})$$

$$\text{InvDen}(\omega) = \frac{\text{Conj}[C(\omega)]}{\text{Conj}[C(\omega)] \cdot C(\omega) + \varepsilon(\omega)}$$



Binaural (Dual Stereo Dipole)

Scheme



advantages:

- 3D sound reproduction
- Rotating of the head
- The cross-talk filters could equalise also the loudspeakers

disadvantages:

- ~~Low frequencies~~
- ~~Coloration outside the "sweet spot"~~

Binaural (Dual Stereo Dipole#2)

Frontal



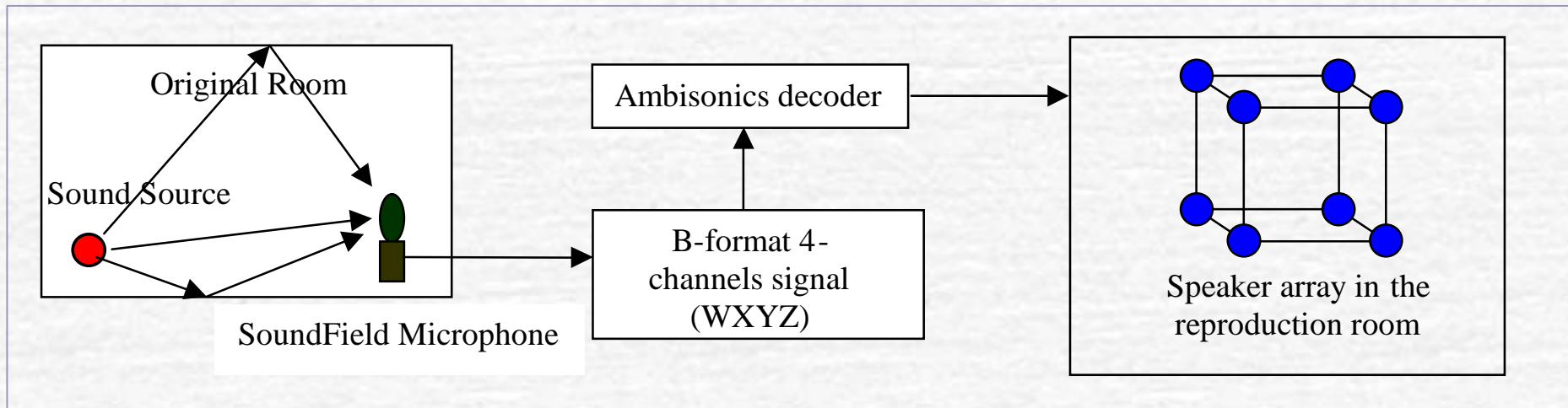
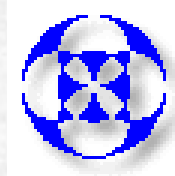
Quested 2108 monitors

Rear



Quested F11P monitors

Ambisonics 3D 1st order



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

Rooms for Ambisonics 3D 1st order



University of Parma

University of Bologna



ITU 5.1 surround

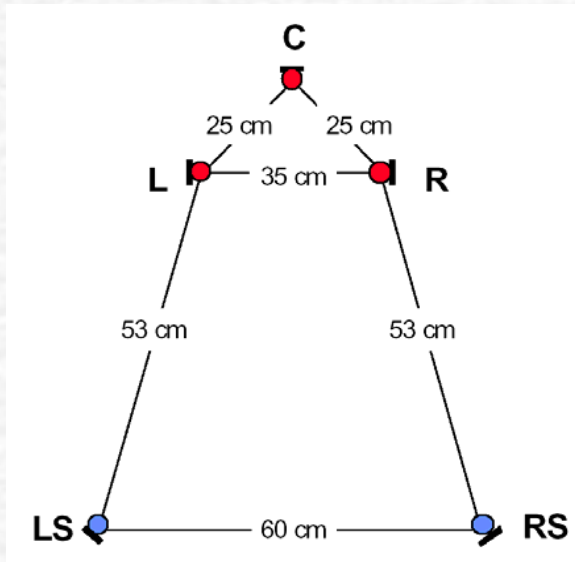
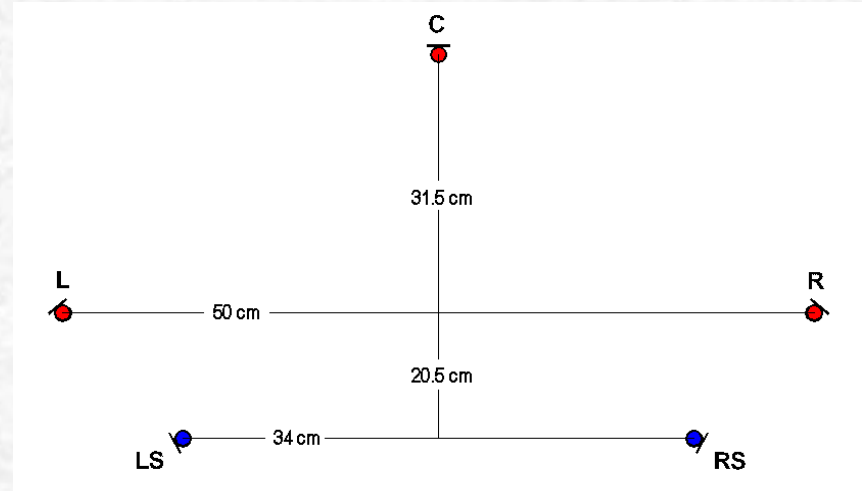
- Williams MMA

Schematic of the setup

C : Cardioid, 0°

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

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



- INA-5

Schematic of the setup

C : Cardioid, 0°

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

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



ITU 5.1 surround

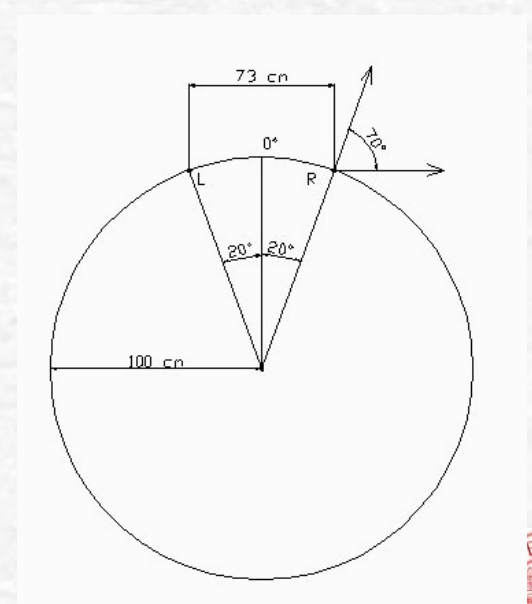
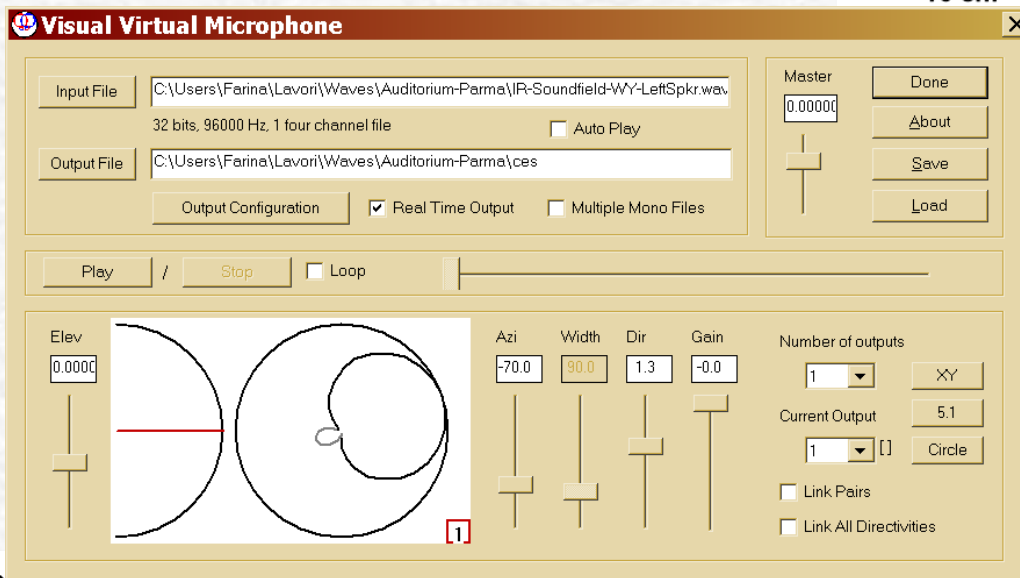
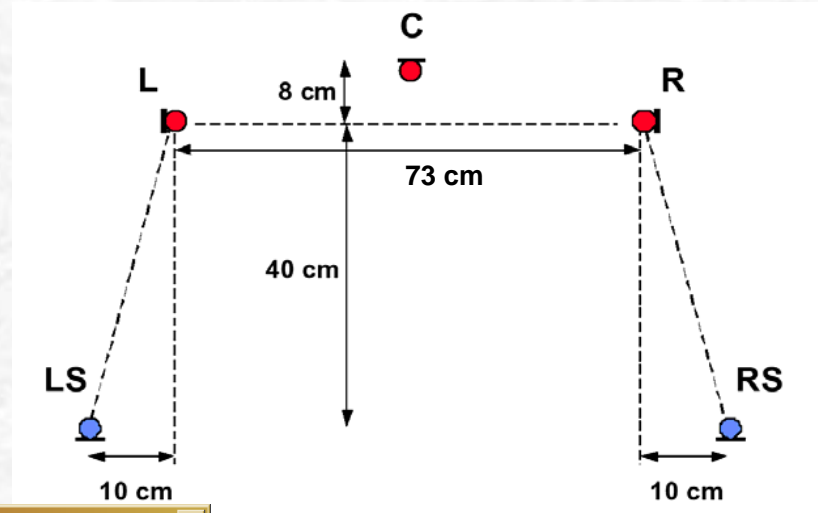
OCT

Schematic of the setup

C : Cardioid, 0°

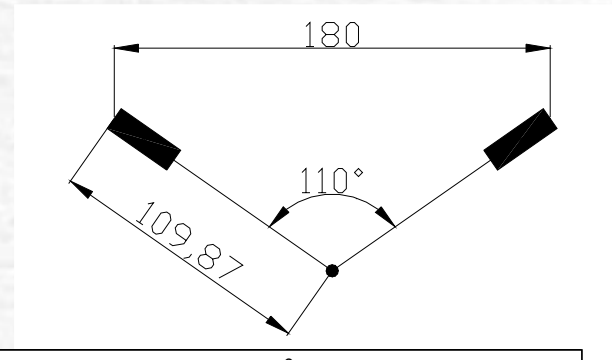
L, R : Super Cardioid, $\pm 90^\circ$

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

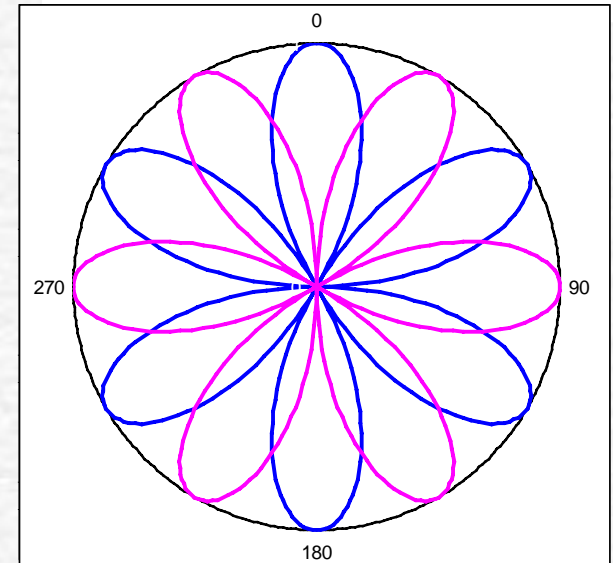


Virtual high-order microphones (M. Poletti)

One of the two ORTF cardioid is employed, which samples 36 positions along a 110 mm-radius circumference

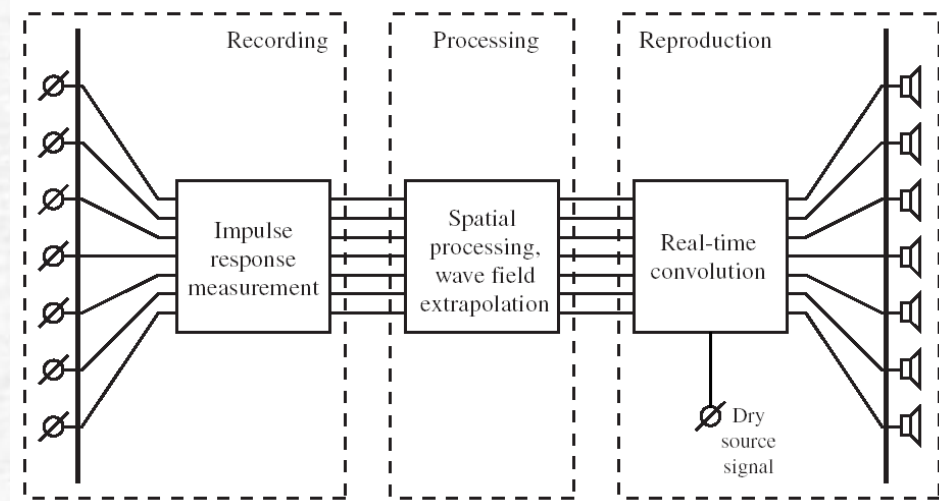
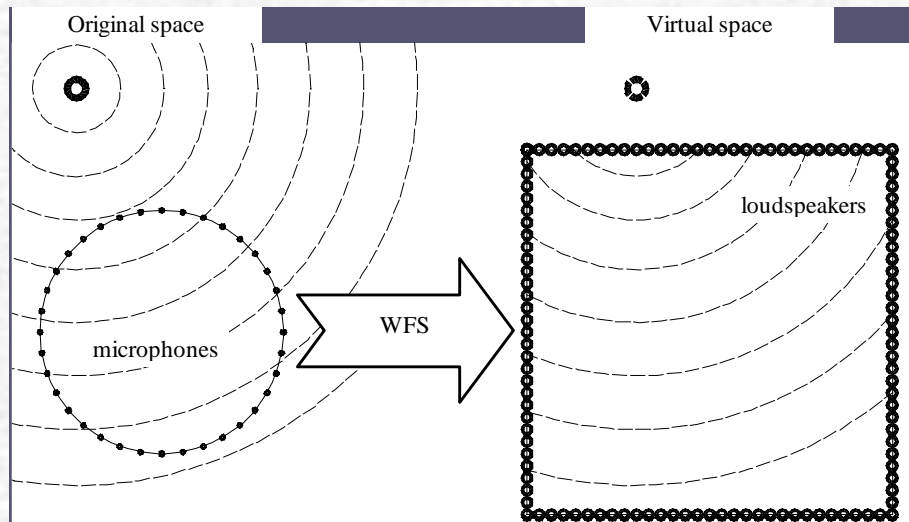


From these 36 impulse responses it is possible to derive the response of cylindrical harmonics microphones (2D Ambisonics) up to 5th order.



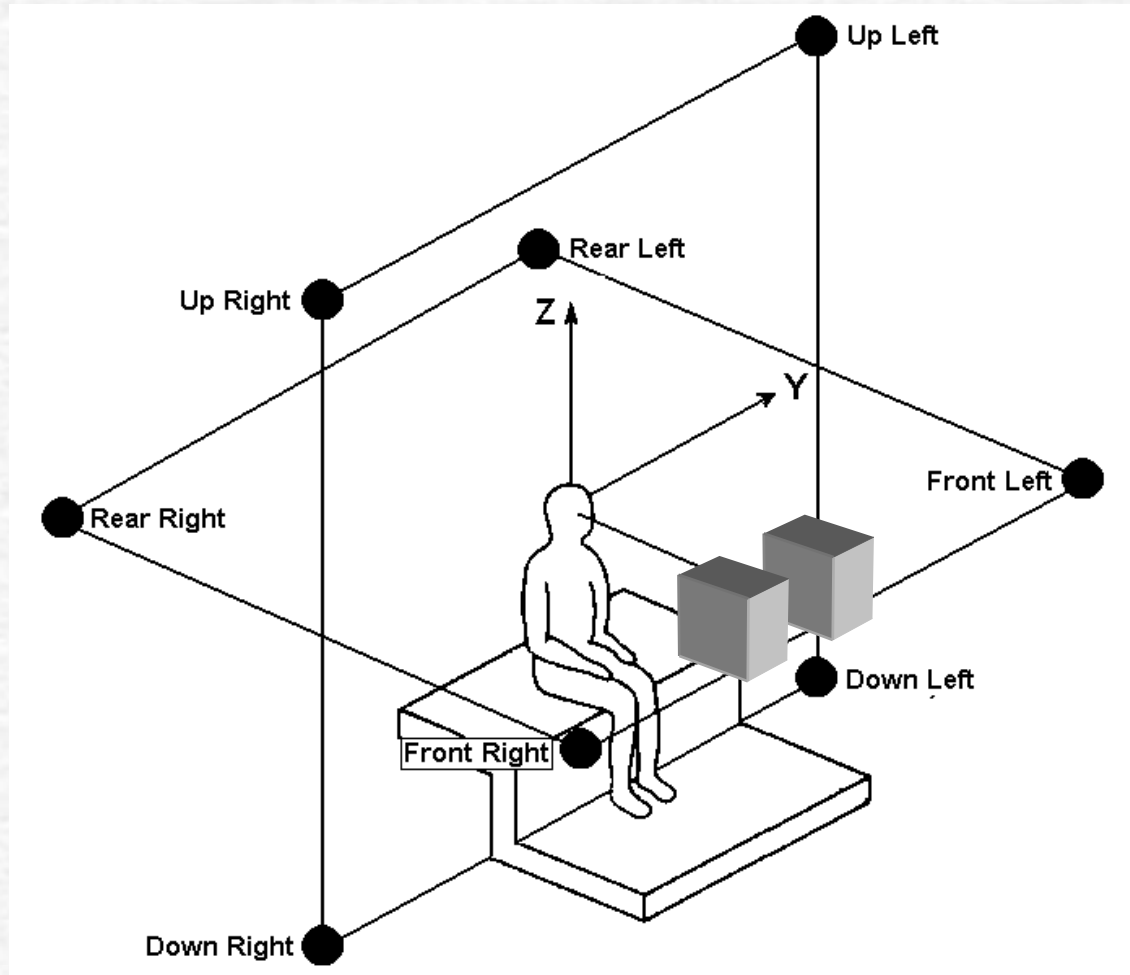
Wave Field Synthesis (WFS)

Flow diagram of the process

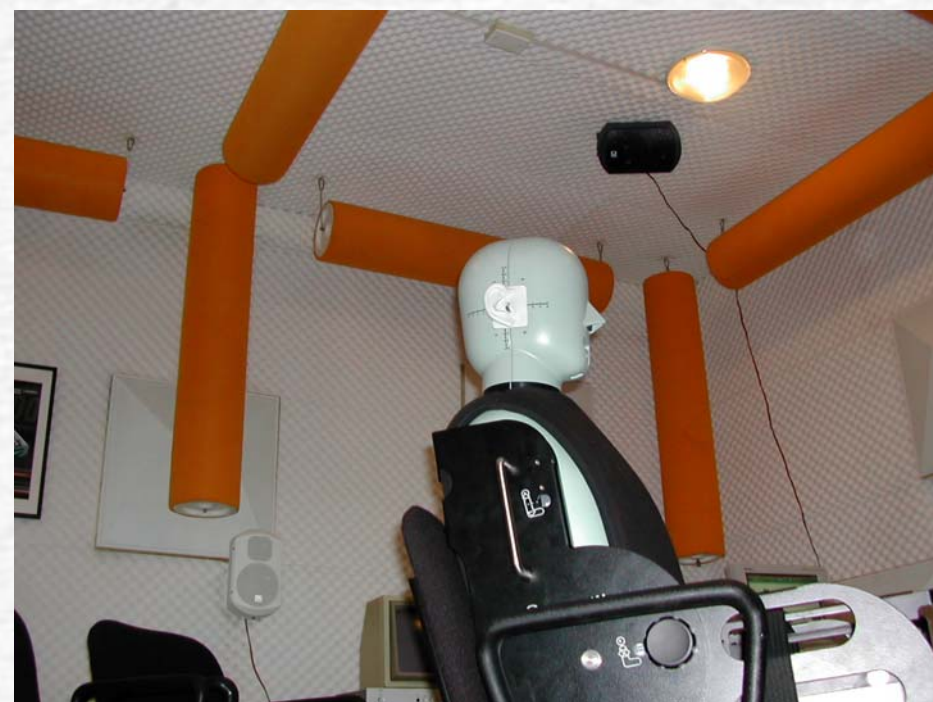


Hybrid methods (Ambiophonics)

Ambiophonics 3D (10 loudspeakers):

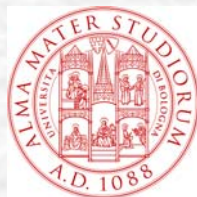


Hybrid methods (Ambiophonics#2)



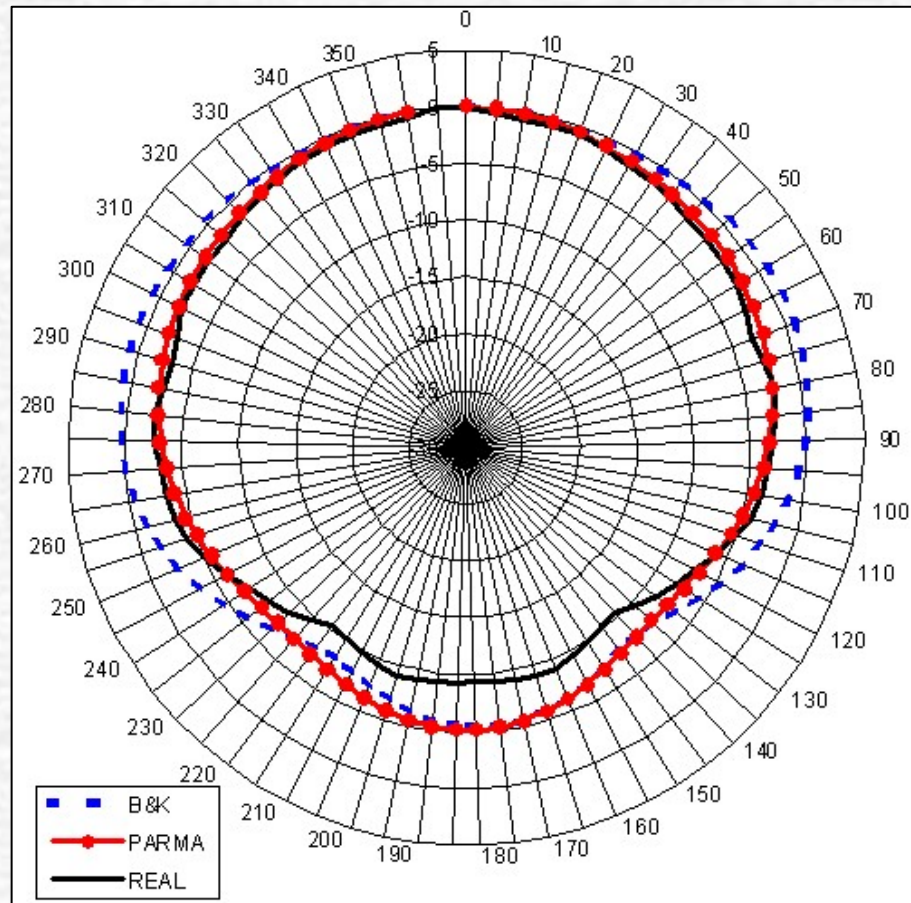
Conclusions

- The spatial informations have to be accurately sampled, making it possible to store, analyze and preserve these "3D acoustical photographs" of existing musical spaces for the posterity
- Many different kinds of impulse-response measurements are required for different 3D auralization methods: a proper set-up should include all different approaches
- Once the impulse responses are stored in suitable multichannel formats, they become available for surround productions with today technologies (ITU 5.1, 1st order Ambisonics) and future, more advanced methods (high order Ambisonics, WFS, Ambiophonics)
- The only point which requires substantial enhancements: sound sources used for IR measurements



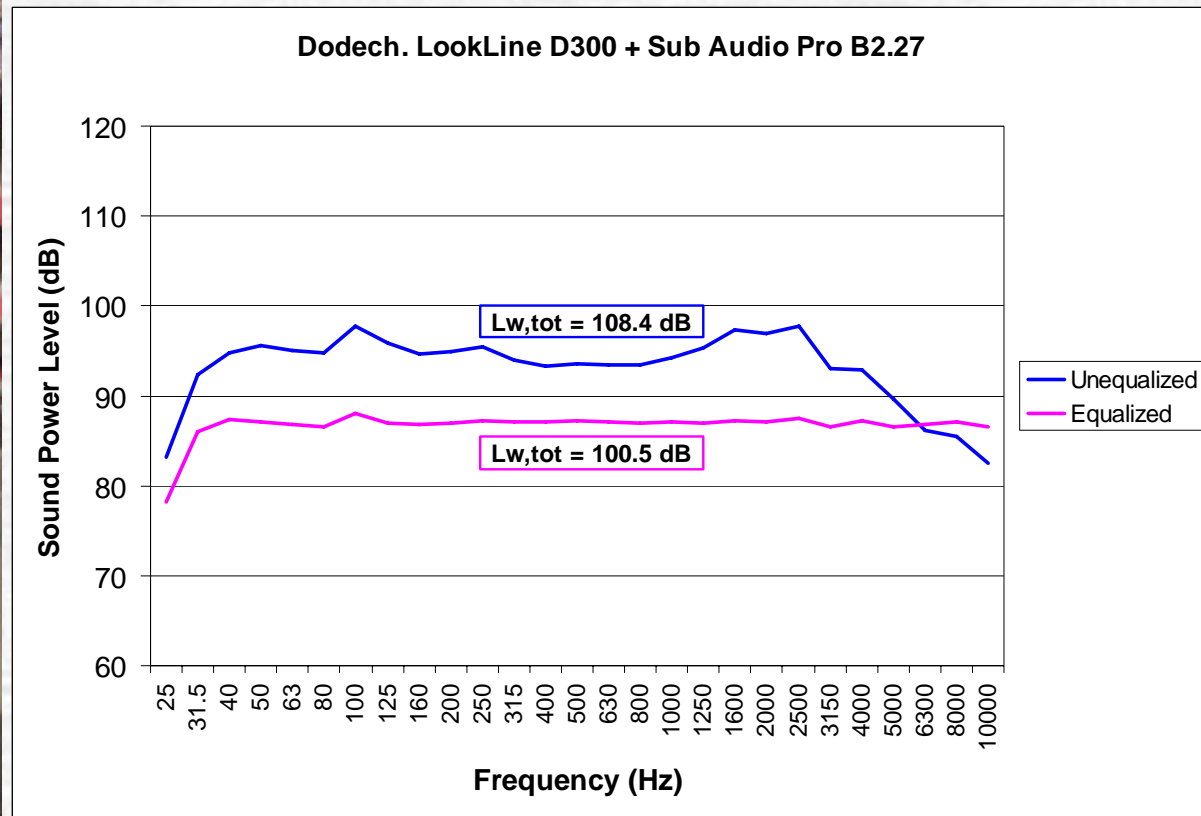
Future enhancements

- Sound source for realistic emulation of an human singer



Future enhancements

- Omnidirectional sound source with enhanced power frequency response



Acknowledgements

- This research was started thanks to the support of Waves, Tel Aviv, Israel (www.waves.com)
- For years 2004 and 2005 the research is also supported by the Italian Ministry for the University and Research (MIUR)
- During 2004 a new web site will be started:
www.acoustics.net

This will provide free access to the whole database, and it will be possible to add contributions.