



University of Parma

Industrial Engineering Department

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AcMus

3 a 5 de
novembro
2004

I Seminário

Música Ciência e Tecnologia

ACÚSTICA MUSICAL

“Listening Tests Performed Inside a Virtual Room Acoustic Simulator”

Authors: A. Farina, P. Martignon, A. Azzali, A. Capra

GOALS

This paper describes the processing required for obtaining a **realistic audible sound reproduction** and how to perform **good listening tests**.

The basic tool is a convolution procedure between “dry” music and the impulse responses measured (or numerically simulated) of a theatre

- The overall process is usually known as “**Auralization**”, and traditionally it was performed through the binaural technology (headphones reproduction)
- Here the process is generalized to many more loudspeaker-based reproduction systems: Stereo, Binaural, Stereo Dipole, Dual Stereo Dipole, Ambisonics, Ambiophonics.



Theatre la Fenice, Venice



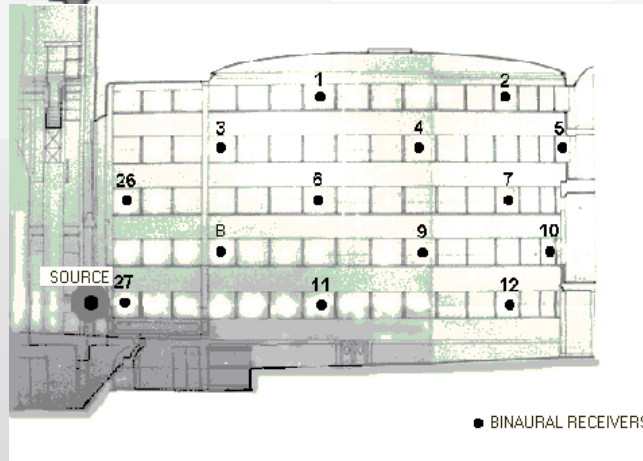
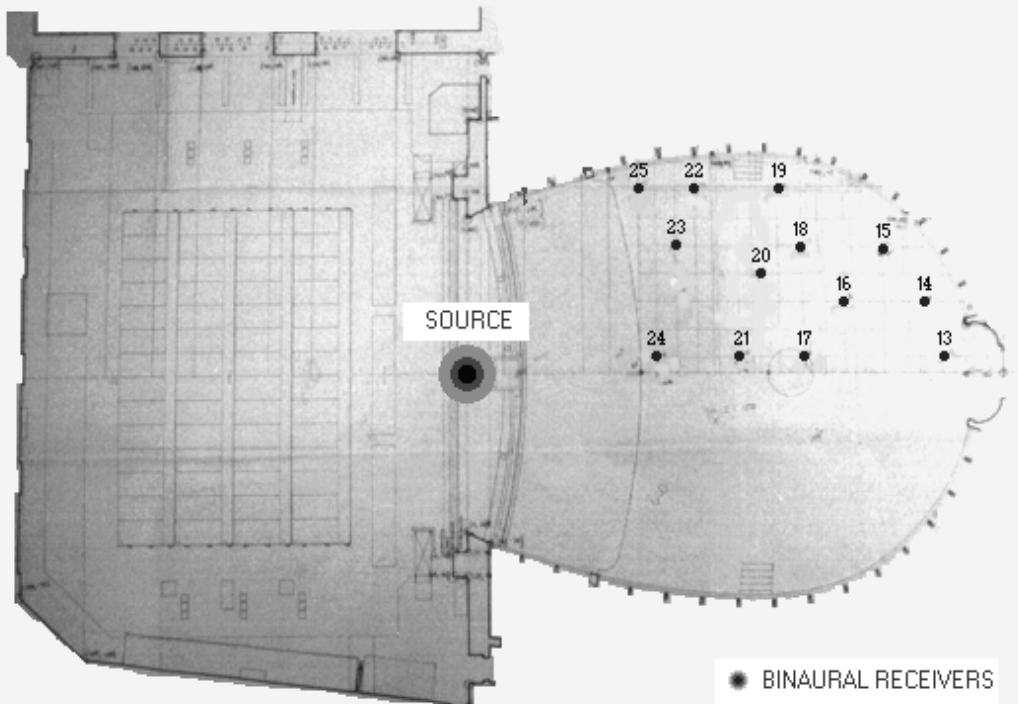
- The first theatre was realised in 1792 by Gian Antonio Selva, after the burning of Teatro San Benedetto
- In December 1836 the theatre burned down again and was rebuilt by G. and T. Meduna the year after
- The theatre was closed in 1995 for maintenance; it had to open again in February 1, 1996, but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

Acoustical measurements



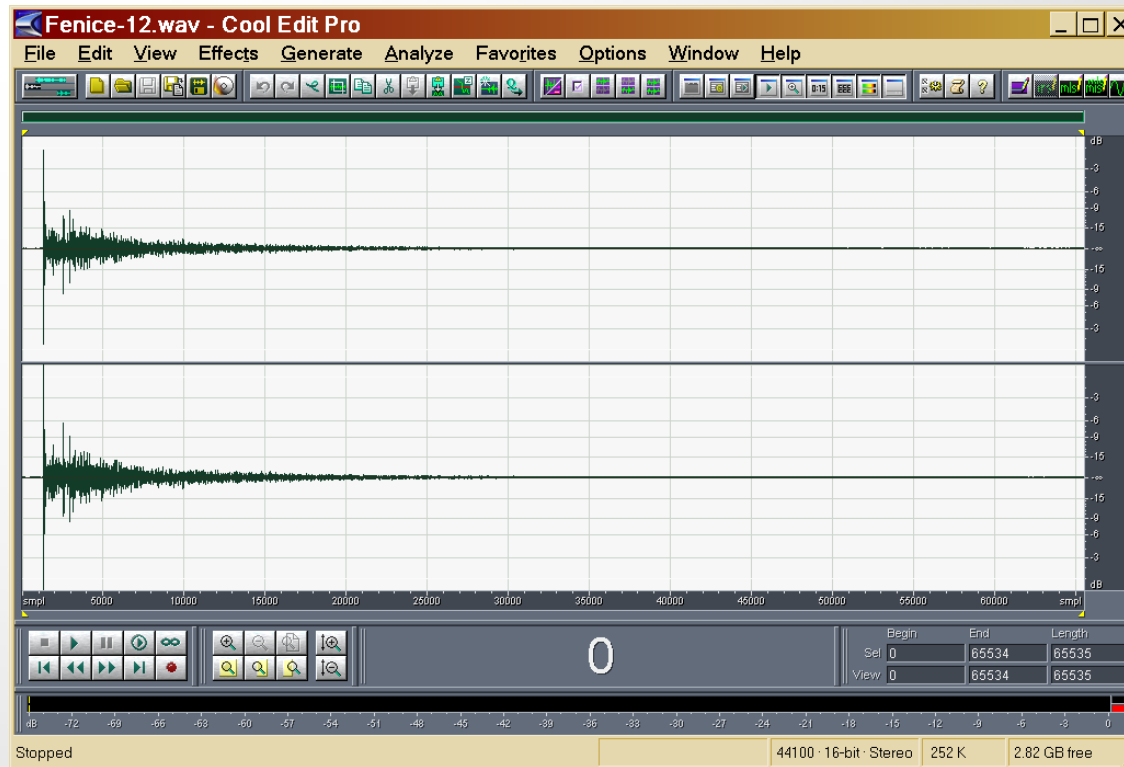
- Measurements were performed by L. Tronchin in November-December 1995 with a modified gun (with omnidirectional diffuser) and binaural microphones
- The goal was to analyze some acoustical problems about intercommunication between orchestra pit and stage and to gather information for designing the orchestra shell
- The data were processed with the software Aurora, which had been developed just 2 months before.

Acoustical measurements



- 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 the measurements the room was perfectly fitted, whilst the stage was empty (no scenery)

Impulse Responses of La Fenice



Point n. 12

Auralization Examples – La Fenice



Stop

Overture alle Nozze di Figaro di Mozart

- Dry music
- Convolution with experimental I.R. (pt. 12)
- Convolution with computer-simulated IR

Preludio al primo atto della Traviata di G.Verdi

- Dry music
- Convolution with experimental I.R. (pt. 12)
- Convolution with computer-simulated IR

The Past

A PRELIMINARY SUBJECTIVE TEST

Test: questionnaires compiled during headphones listening of several tracks played in 5 different theatres

Italian theatres choosen:

- Opera theatres


Teatro Regio (Parma)
Teatro Valli(Reggio E.)

- Auditoria

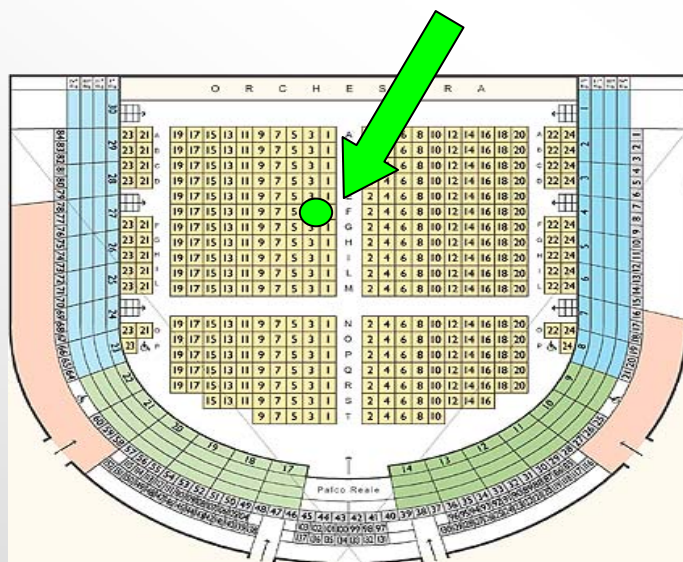

Paganini (Parma)
Sala 700 (Roma)

- Historical theatres

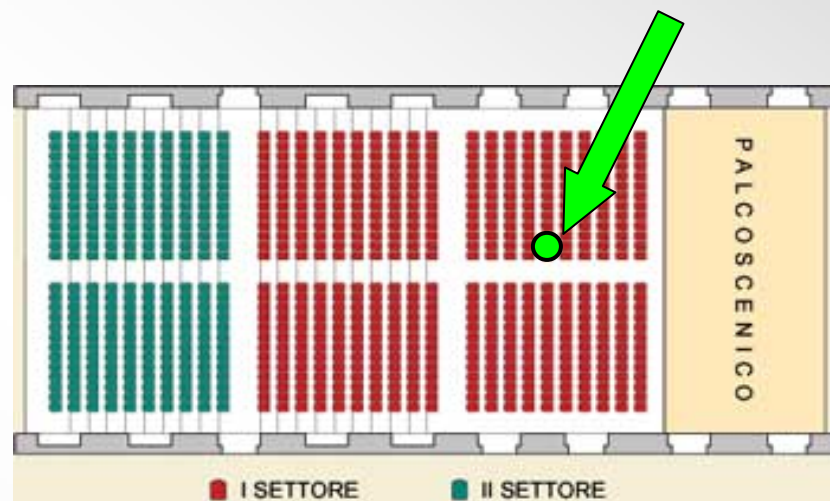

Teatro Olimpico (Vicenza)



Track used for the test: anechoic tracks auralized with binaural Impulse Response of 5 theatres. Acquisition of Impulse Response made with source on the stage and receiver placed between 5th and 6th of every room.



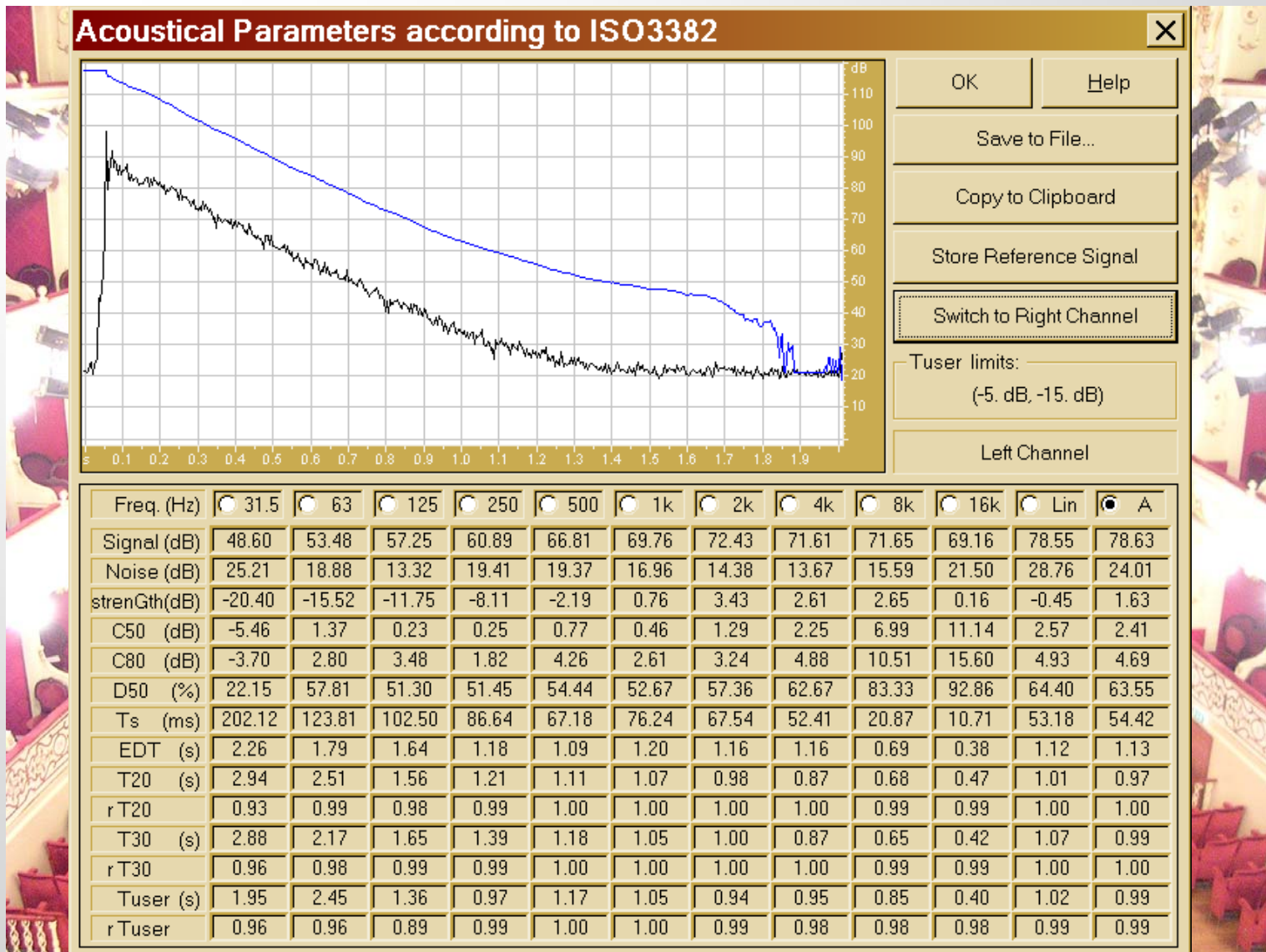
Teatro Regio (Parma)



Auditorium Paganini (Parma)

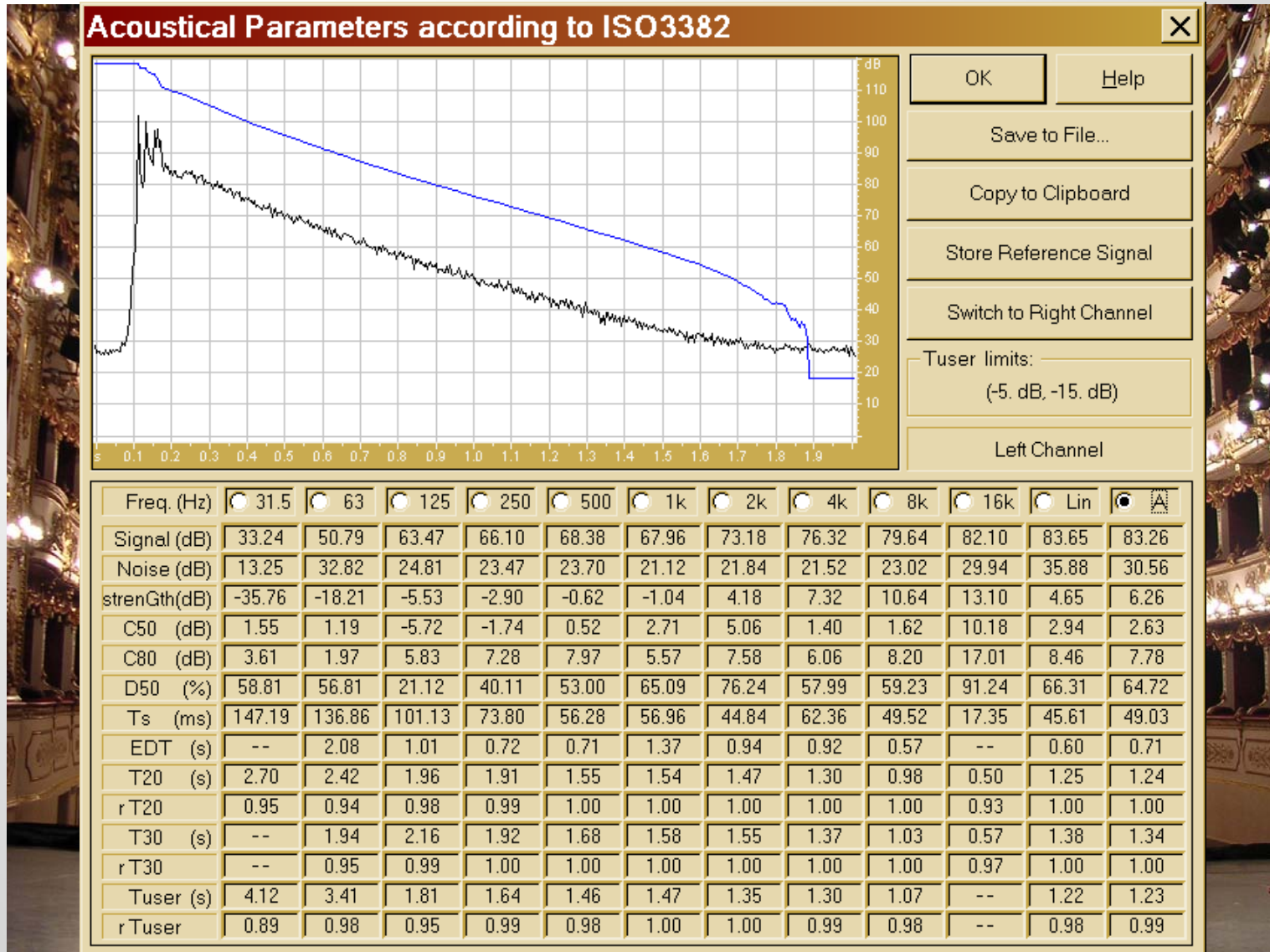


Teatro Regio in Parma (Italy)



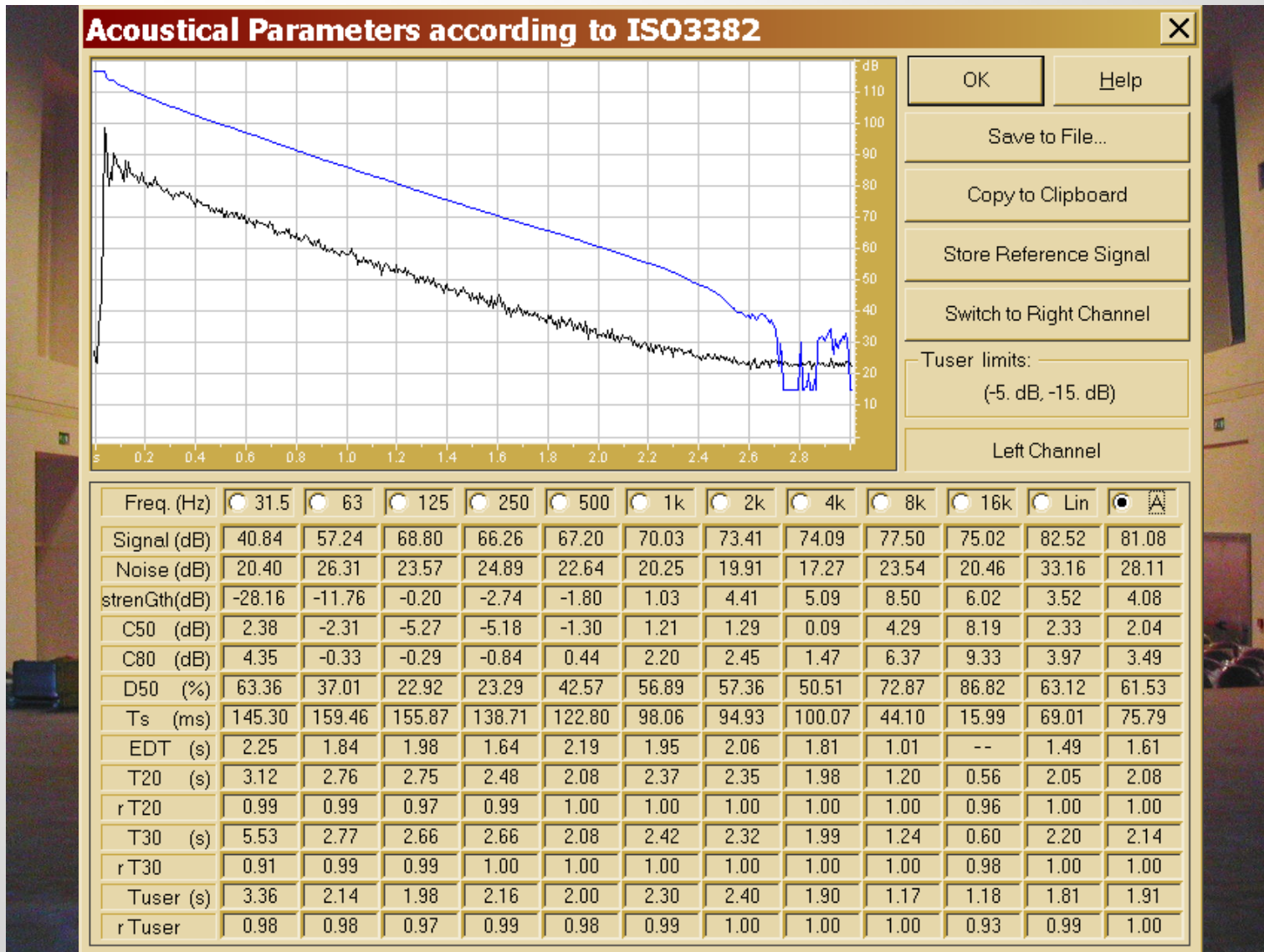
$T_{20} = 1.11 \text{ s}$

Teatro Valli, Reggio Emilia, Italy



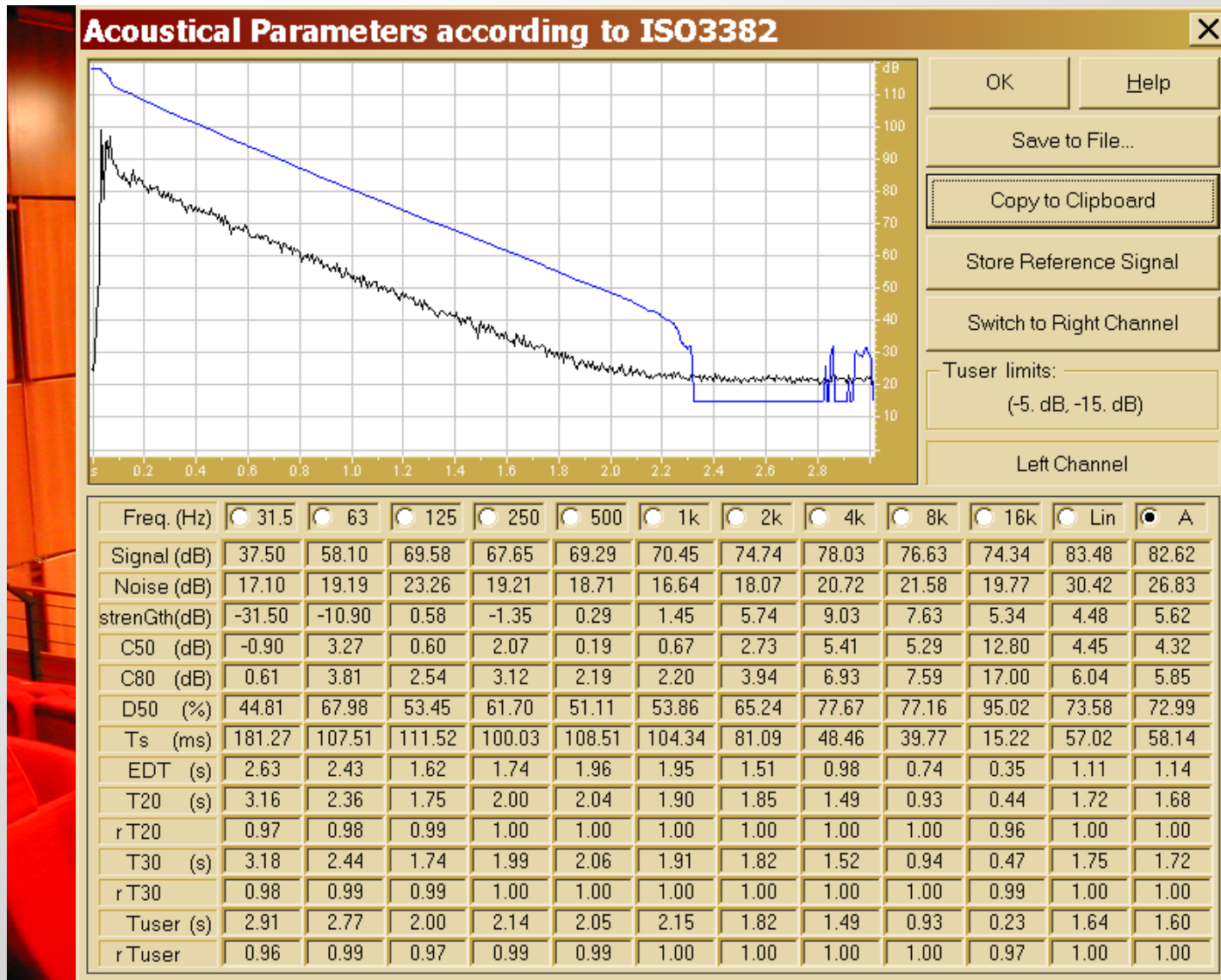
$T_{20} = 1.55 \text{ s}$

Paganini Auditorium, Parma, Italy



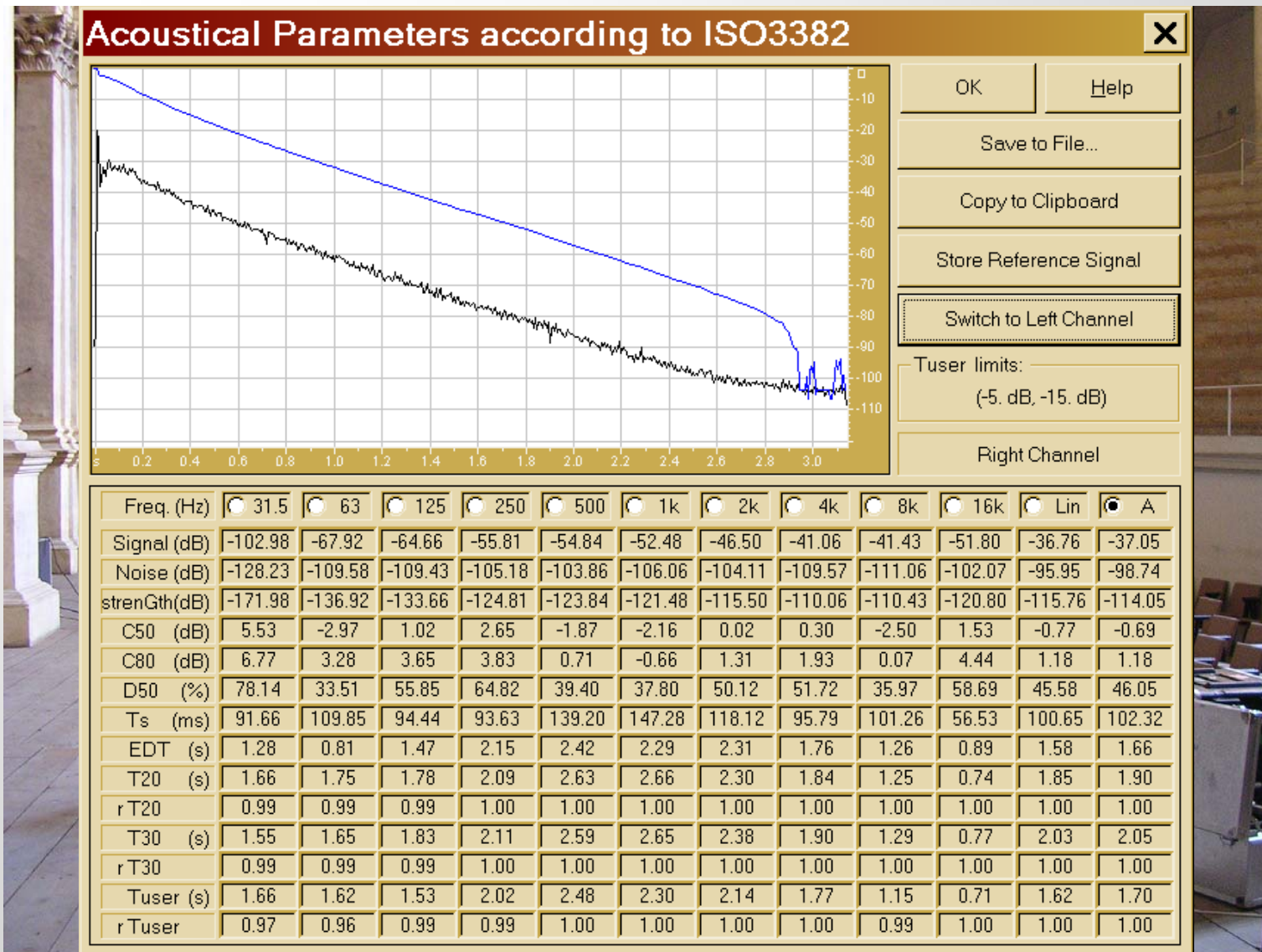
$T_{20} = 2.08 \text{ s}$

S.Cecilia Auditorium 700, Rome, Italy



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

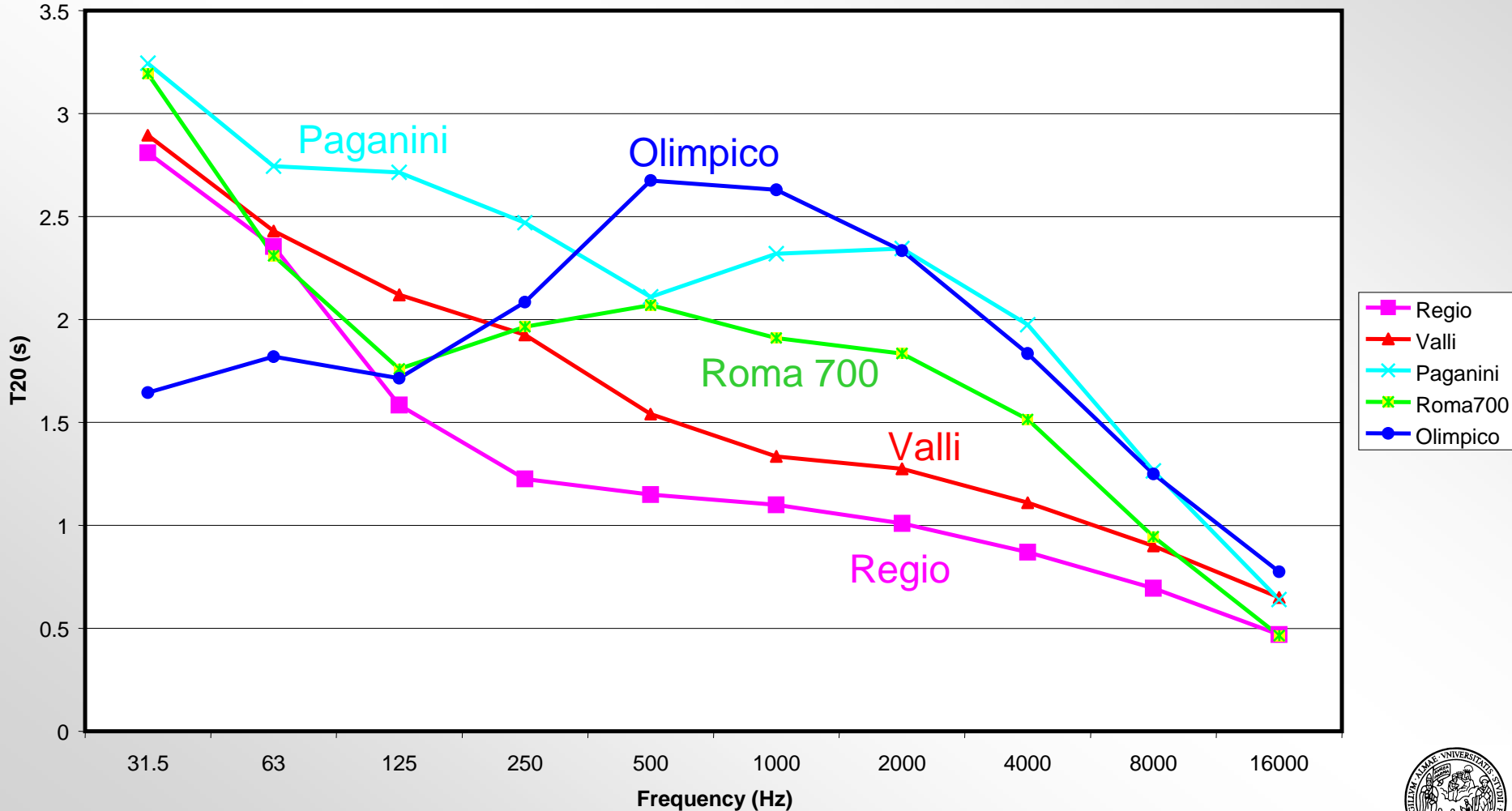
Teatro Olimpico, Vicenza, Italy



$T_{20} = 2.63 \text{ s}$

Acoustical properties of the theatres

Reverberation Time T20 of 5 Italian theatres



Acoustical properties of the theatres

According to ISO 3382 standard

Param.	Regio	Valli	Paganini	Roma-700	Olimpico
C50 [dB]	1.82	7.48	-1.47	-2.45	0.03
C80 [dB]	4.93	9.28	0.81	0.83	1.20
D50 [%]	60	84	42	37	50
Ts [ms]	48	28	115	144	110
EDT [s]	1.08	1.26	2.09	1.98	2.43
T20 [s]	1.10	1.44	2.22	1.99	2.65
T30 [s]	1.11	1.55	2.24	1.99	2.64
LF	0.10	0.12	0.22	0.12	0.18
IACC (Early)	0.71	0.88	0.60	0.70	0.81
TB	1.47	1.70	1.20	1.11	0.91
BR	1.27	1.41	1.17	0.94	0.72



Sound Samples

Chosen in order to evidence the difference between rooms designed for opera and rooms for purely orchestral music:

ORCHESTRAL:

- G.Verdi, Preludio al Primo Atto de "*la Traviata*"
- W.A. Mozart, Overture de "*le Nozze di Figaro*"
- Strauss, "*Pizzicate Polka*"

VOCAL:

- Mozart, "*Così fan tutte*" (voice and piano)
- Tosti, "*Non t'amo più*" (voice and piano)
- "*My Funny Valentine*" (jazz, voice solo)



Listening seat in **Casa della Musica** (Parma-Italy):

Instrumentation:

- A liquid cooled Computer (*Futureclient*)
- Open dynamic headphones
Sennheiser HD 580 Precision



- Audio-pro Subwoofer for very low frequencies (18-50 Hz)



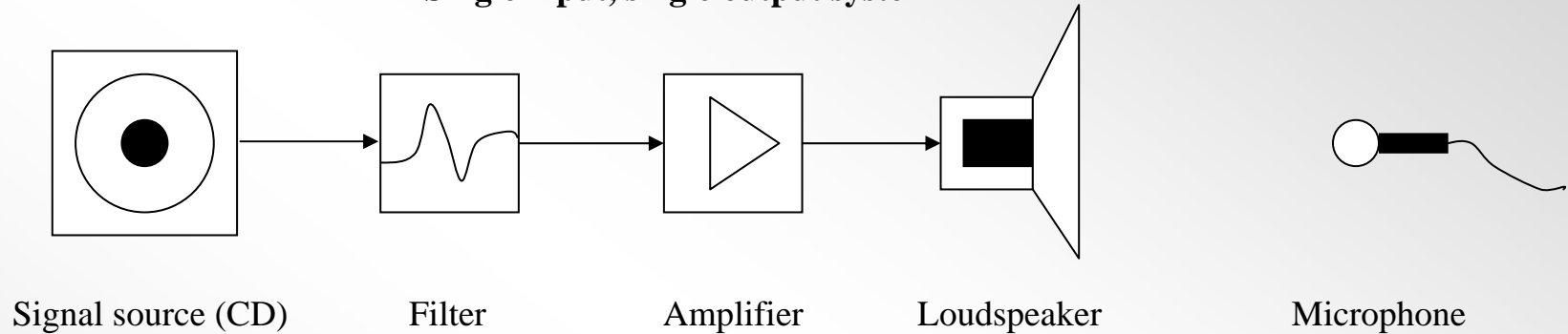
Headphones

- Open dynamic headphones: *Sennheiser HD 580 Precision*
- Digital equalizing filters are employed for compensating the frequency response of the headphone+dummy head
- The computation of these inverse filters revealed to be very important for the “transparency” of the reproduction chain

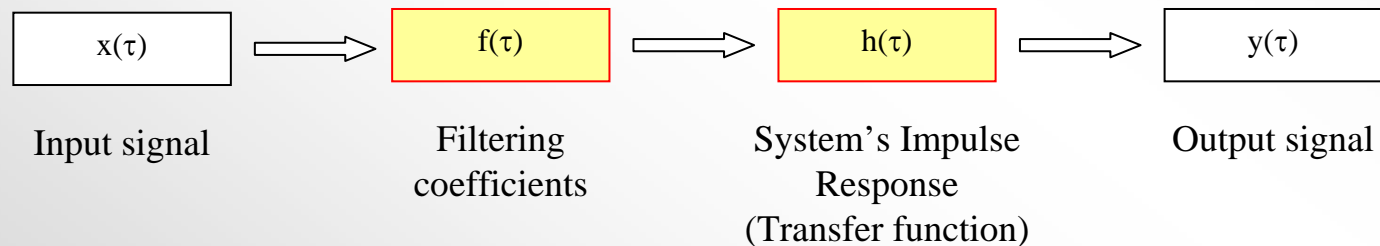


Theory of inverse filters

Single input, single output system



Block diagram



Combined transfer function

As all the stages are linear:

$$y(i) = x(i) \otimes f(j) \otimes h(l)$$

The goal of the filter is to “equalize” – this means to make the output $y(i)$ to be equal to the input $x(i)$. This is obtained if:

$$f(j) \otimes h(l) \Rightarrow \delta(i)$$

In which $\delta(i)$ is the Dirac’s Delta function (a single sample having unit value, preceded and followed by thousands of zeroes) – this way the total effect of the filter+system is simply a delay of a few milliseconds, with no other evident alteration

Possible design strategies for an equalizing filter

- Mourjopoulos – Least-squares recursive method in time domain – the whole frequency range is always completely inverted.
- Neely & Allen – the filter is designed in the frequency domain – only the magnitude of the transfer function is inverted, so the equalized system will have flat frequency response, but it will still be “smeared” in time.
- Nelson & Kirkeby – again in the frequency domain, but the whole complex value is inverted, adding a small regularization quantity at the denominator for avoiding instabilities and ensuring a finite length of the inverse filter

Theory of Kirkeby inversion

- **Step 1 – pass to frequency domain through FFT**

$$H(\omega) = \text{FFT}[h(\tau)]$$

- **Step 2 – make the complex reciprocal at each frequency:**

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

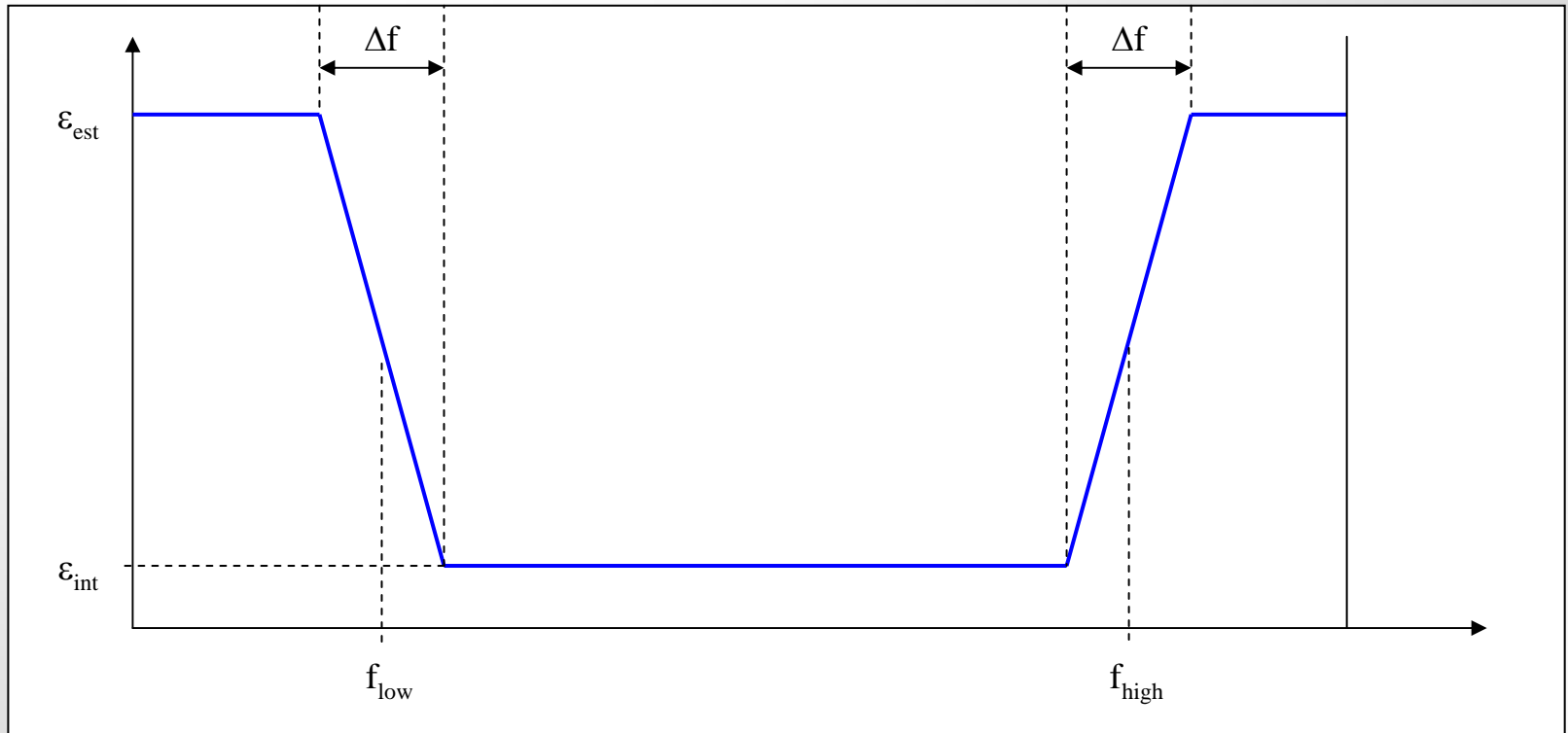
- **Step 3 – go back to time domain through an IFFT:**

$$f(\tau) = \text{IFFT}[F(\omega)]$$

Parametro di regolarizzazione



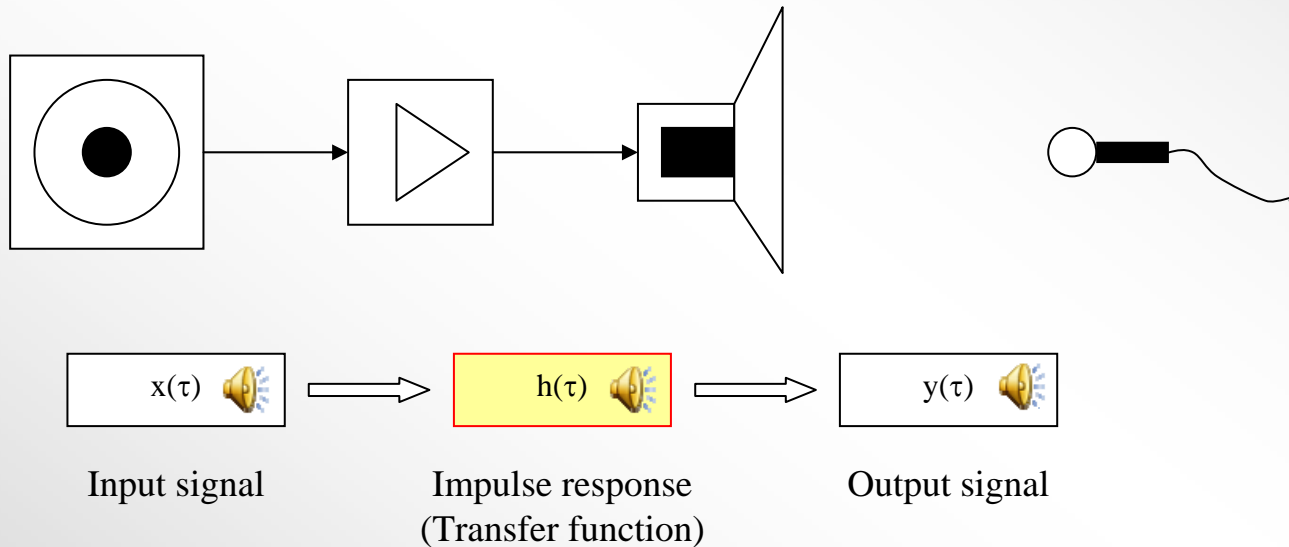
Regularization parameter $\varepsilon(\omega)$ variable with frequency



Changing the value of the regularization parameter allows for a very accurate filter at central frequency, and progressively a less aggressive filtering at very low or very high frequencies.

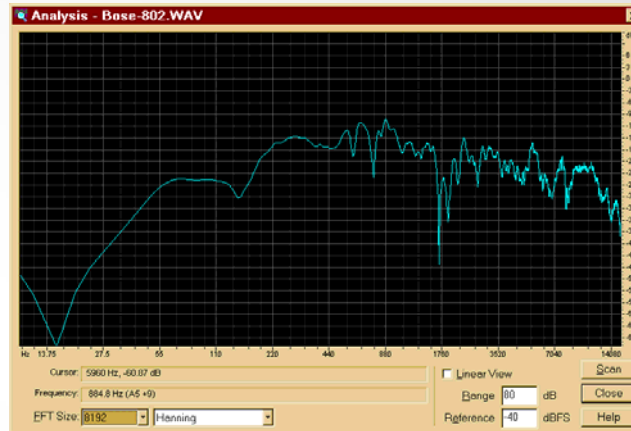
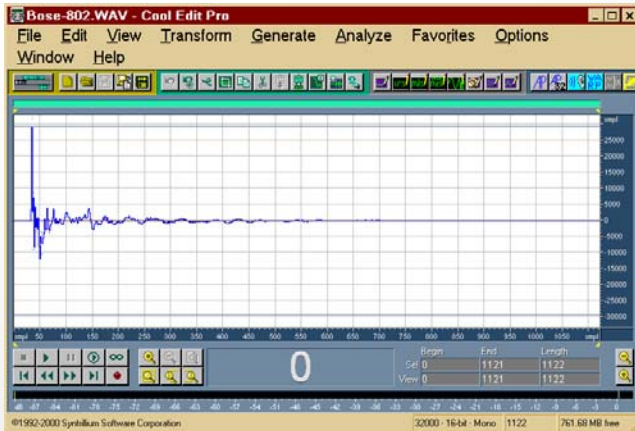
Example

- A loudspeaker+microphone system was measured:

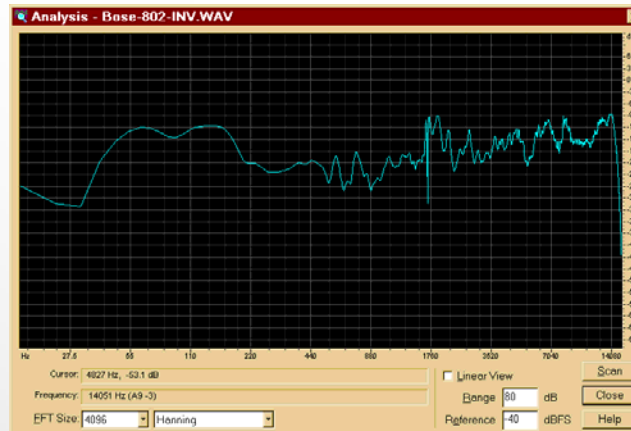


Inverse filter example

- System's impulse response

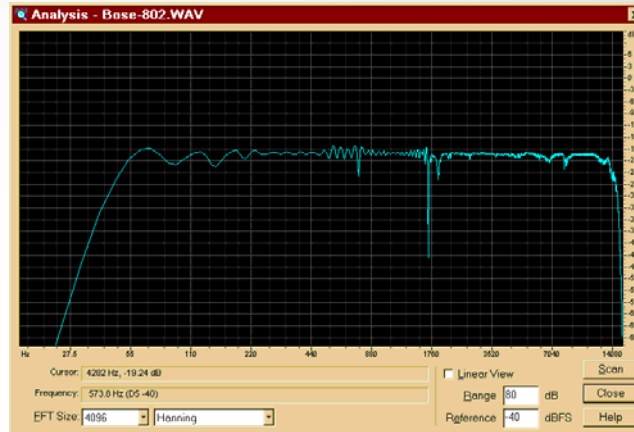
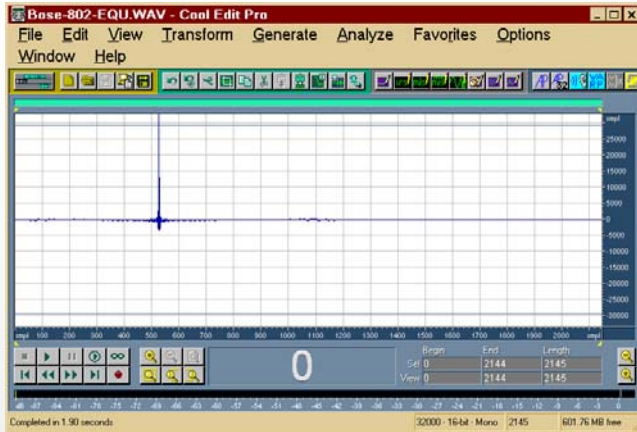


□ Filtro inverso

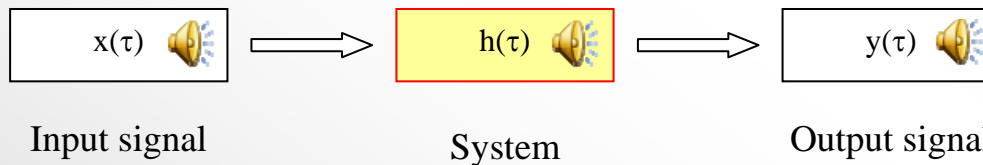


Inverse filter example

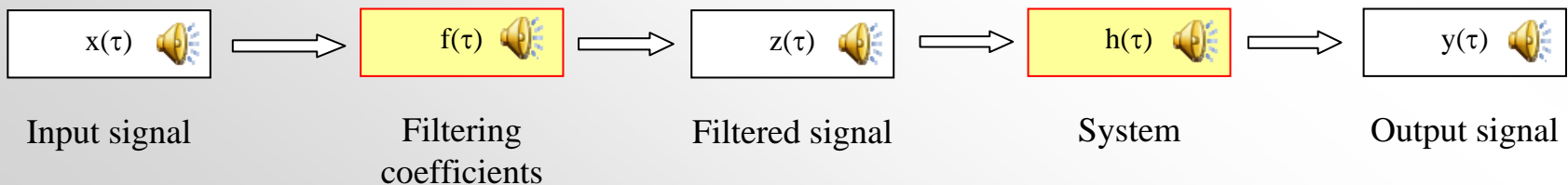
- Convolution of inverse filter with the system's impulse response



Not filtered system



System with filter



The software

- Real-time control over the theatre (ABCDE) and the music piece (123456)
- Collection of the questionnaires with a graphical user's interface
- The user is allowed to switch at will the sound samples and theatres, and to change the responses



Statistical Analysis of the results

Aim: possible correlation between objective parameters measured inside the theatres and subjective descriptors.

Objective parameters chosen:

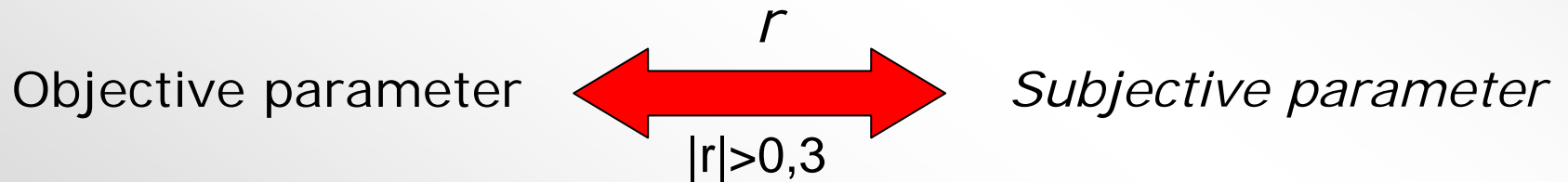
- **Monophonic parameters:** reverberation times **T10, T20, T30, EDT**; clarity **C50** e **C80**; center time **Ts**; **D50**.
- **Spatial parameters:** *Lateral Fraction(Lf)* and *Inter Aural Cross Correlation (IACC)*
- **Tonal parameters:** *Tonal Balance (TB)* and *Bass Ratio (BR)*



Method of analysis: *linear regression* (multiple regression to manage the whole matrix of subjective/objective data).

Defining r as coefficient of correlation:

$$r = \frac{\sigma_{xy}}{\sigma_x \sigma_y}$$



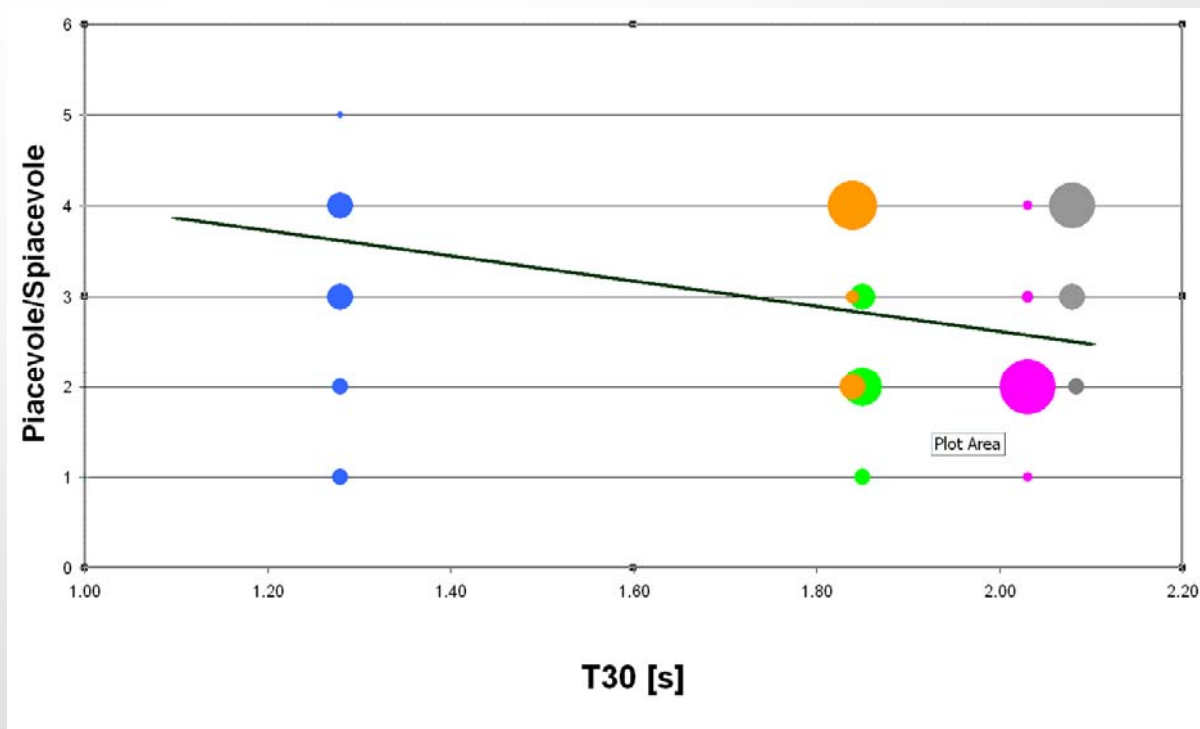
Matrix 9x11 of “r”: objective parameters on the abscissa and subjective parameters on the ordinate.

Coeff. Di Regressione Lineare ORCHESTRALI												
	C50	C80	D50	Ts	EDT	T20	T30	LF	IACC	TB	BR	
Piacevole-Spiacevole	0,14	-0,19	0,19	-0,24	-0,19	-0,20	-0,12	-0,21	0,16	-0,07	0,09	
Rotondo-Spigoloso	0,35	0,42	0,46	-0,51	-0,35	-0,37	-0,22	-0,51	0,32	-0,22	0,15	
Morbido-Duro	0,17	0,33	0,28	-0,45	-0,40	-0,39	-0,19	-0,35	0,28	-0,11	0,13	
Diffuso- Localizzabile	0,17	0,30	0,22	-0,36	-0,35	-0,37	-0,26	-0,25	0,26	0,02	0,21	
Distaccato- Avvolgente	-0,17	-0,24	-0,19	0,23	0,16	0,19	0,09	0,21	-0,21	-0,02	-0,17	
Secco-Rimbombante	-0,24	-0,42	-0,32	0,50	0,48	0,50	0,34	0,37	-0,36	0,01	-0,26	
Acuti Accentuati- Acuti Ridotti	-0,19	-0,28	-0,35	0,45	0,31	0,26	0,04	0,44	-0,22	0,31	0,02	
Bassi Accentuati- Bassi Ridotti	0,22	0,29	0,38	-0,48	-0,36	-0,32	-0,18	-0,46	0,20	-0,34	-0,04	
Sommesso-Sonoro	-0,08	-0,01	0,00	-0,11	-0,15	-0,09	-0,02	-0,05	-0,01	-0,12	-0,11	



Results

Bad correlation between objective and subjective parameters.
Example: pleasant vs. T30 gives a correlation $r=-0.20$



● T.Regio ● Roma 700 ● T.Olimpico ● A.Paganini ● T.Valli



Results

IACC \longleftrightarrow **diffuse-localisable**
0.42

Tonal Balance $\begin{cases} \nearrow 0.31 & \text{treble boosted-} \\ \searrow -0.34 & \text{treble reduced} \\ \nearrow & \text{bass boosted-bass} \\ \searrow & \text{reduced} \end{cases}$

Lateral Fraction \longleftrightarrow **round-sharp**
-0.51

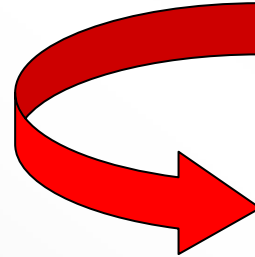


Although the results were inherently bad, they were useful for improving the methodology, along these findings:

- Future listening tests



Only one music sample for test



Short explanation of the attributes

- Statistical analysis of the results



Advanced statistical methods: *factor analysis* or *multivariate regression*.



The Present

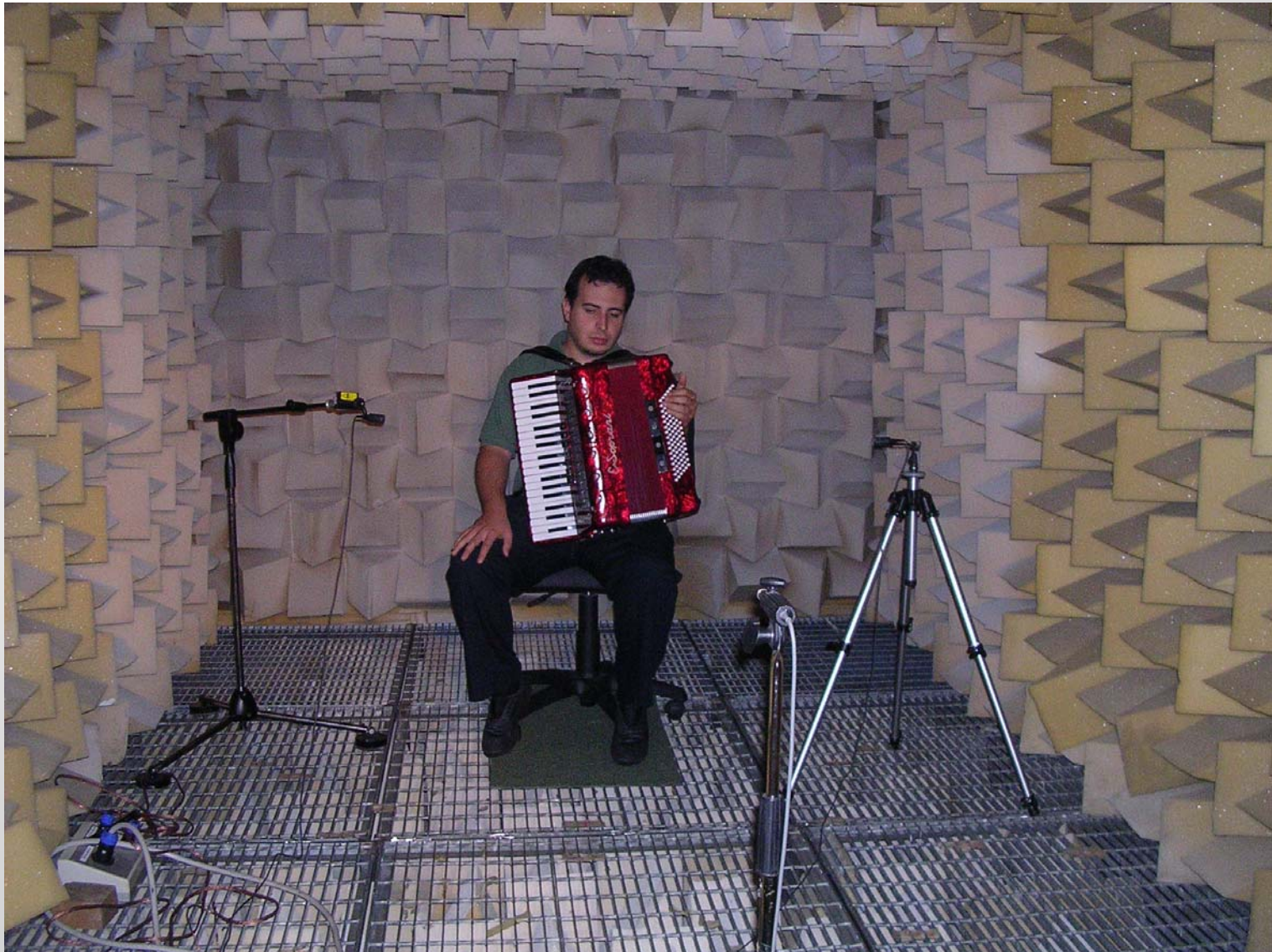
THE NEW LISTENING TEST WITH 4 DIFFERENT SOUND SYSTEMS

- 4 reproduction sound systems being compared:
 1. Headphones
 2. Traditional normal stereo
 3. Stereo Dipole
 4. Dual Stereo Dipole
- Monophonic sound source (accordion) – no multiple soundtracks
- The same 5 virtual rooms as in the preliminary experiment
- Different source-receiver distances in each room



Recording of the anechoic music piece

(ASK Industries, Reggio Emilia)



THE NEW LISTENING ROOM

(“*Casa della Musica*” – Parma)

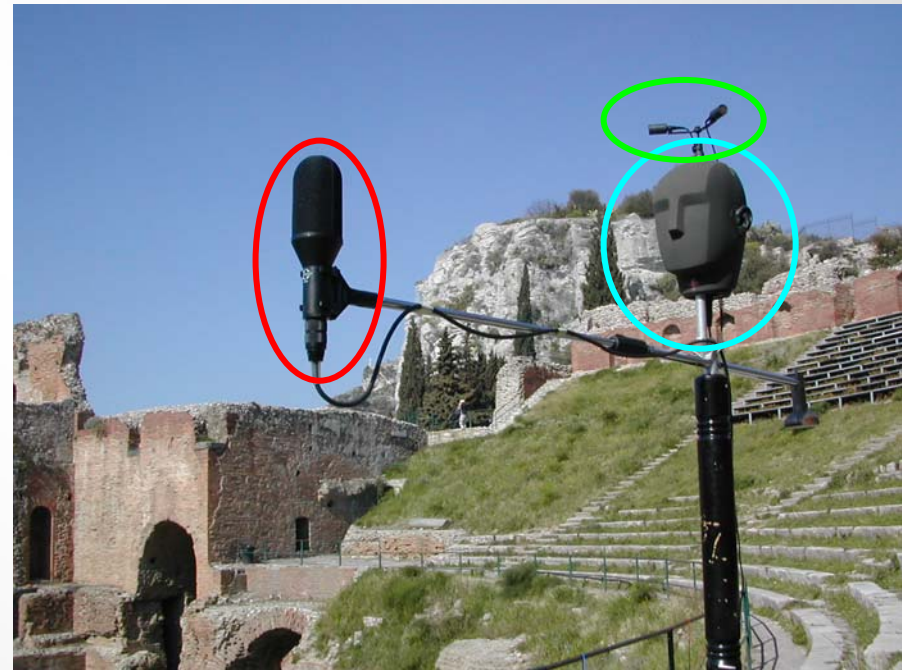
To reduce the reverberation:

- High Frequencies: traditional absorbing panel (glass wool, pyramid, ecc...)
- Low Frequencies: resonant open cavities (cartoon boxes and tube traps), double side rigid and vertical absorbing panels.



Modern multiple-format microphonic systems

- Microphones *Neumann KS-140* in configuration ORTF (angle of 110°)
 - Dummy head *Neumann KU-100* with binaural microphones
 - Soundfield ST-250 microphone probe
- Turn table for recordings to different angles



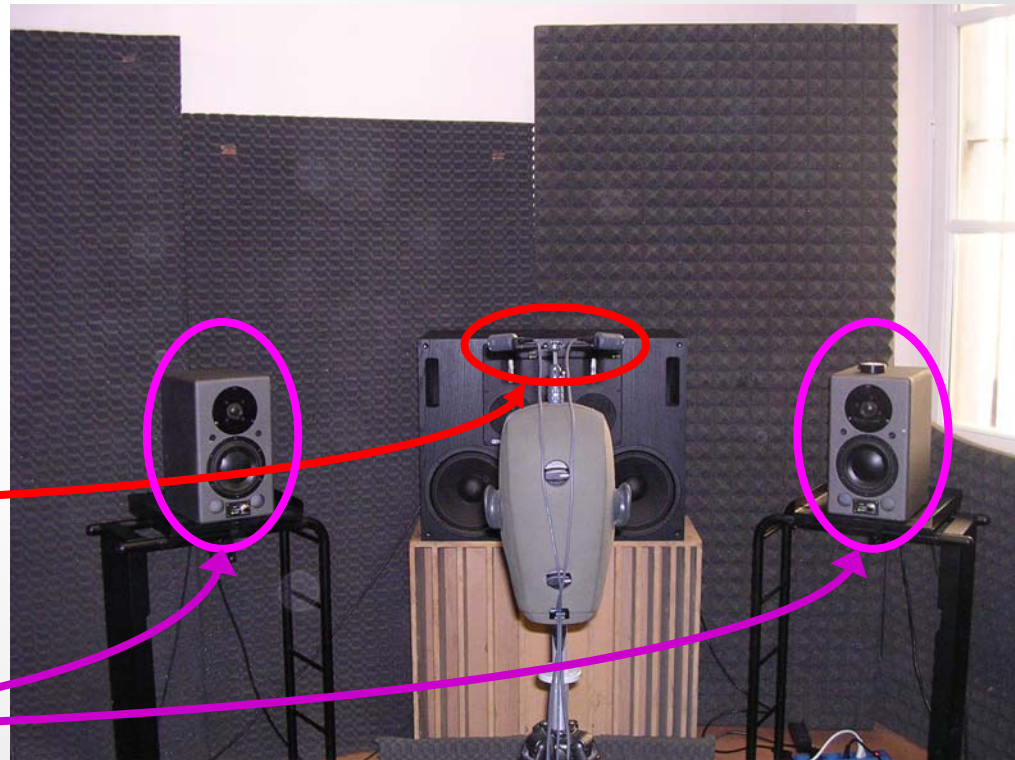
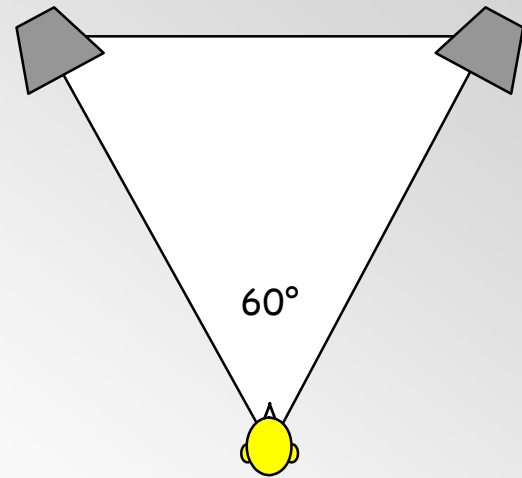
Binaural / HEADPHONES

- *Sennheiser HD 580 Precision*
- In principle, they should put the right pressure exactly where it was recorded, at the ears, maintaining perfect separation and not being affected by the room response.
- Each signal is passed through an inverse FIR filter, which is computed after a measurement performed with the headphones over the dummy head

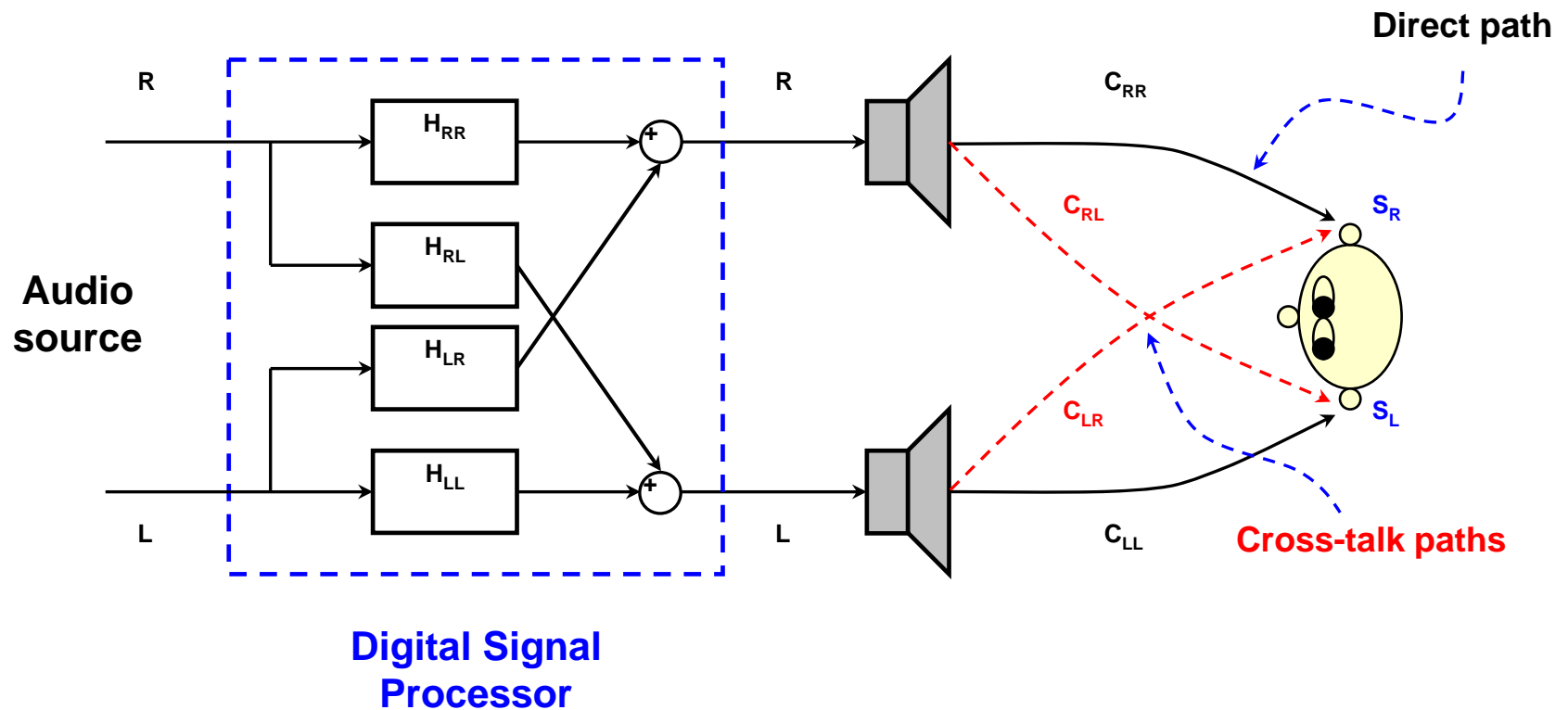


THE NORMAL STEREO

- *Dynaudio* self-powered studio monitors
- The sound is picked up by the two cardioid microphones
- Each of the two signals is passed through a proper inverse filter, computed after a measurement performed placing the **ORTF microphones** at the listening position, in front of the **loudspeakers**



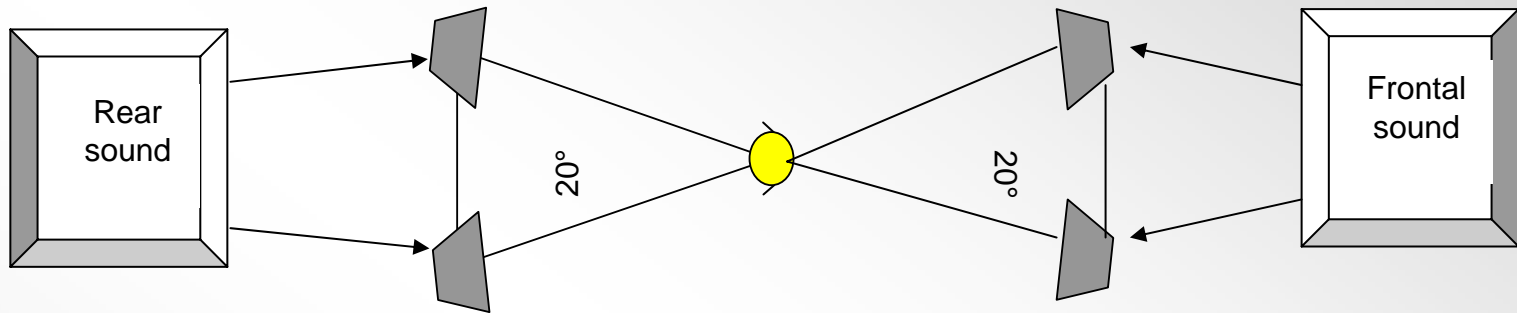
THE STEREO DIPOLE



The cross-talk cancellation allows for the replica of the recorded signals at the ears of the listener



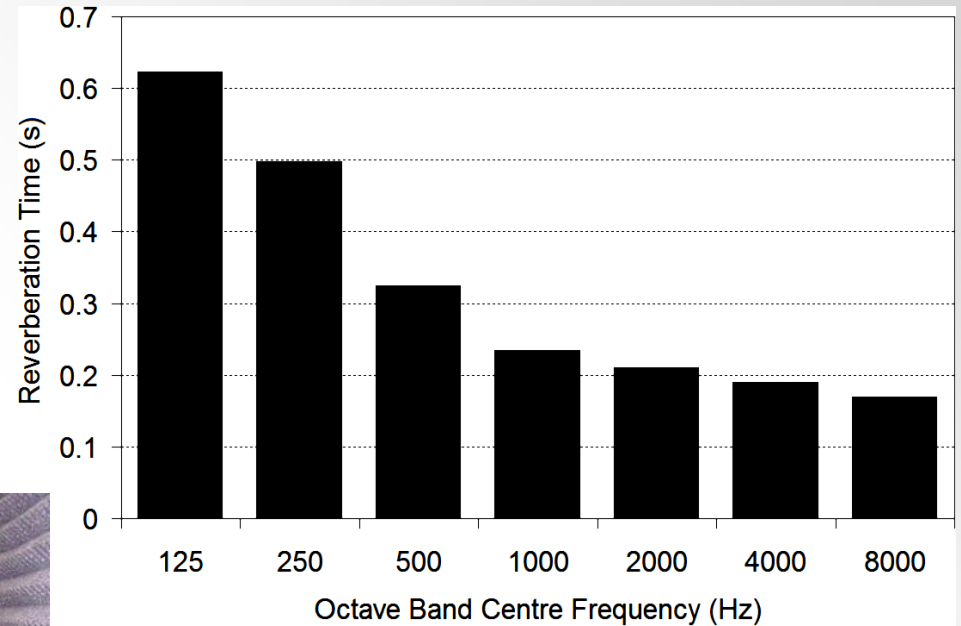
THE DUAL STEREO DIPOLE



Reproduction over a Dual-Stereo-Dipole loudspeaker rig

It is a four-channel system, in which a frontal stereo dipole is employed for reproducing the sound coming from directions located in the frontal hemispace, and the rear stereo dipole reproduces the sound coming from the rear hemispace.

Reverberation Time of the listening room

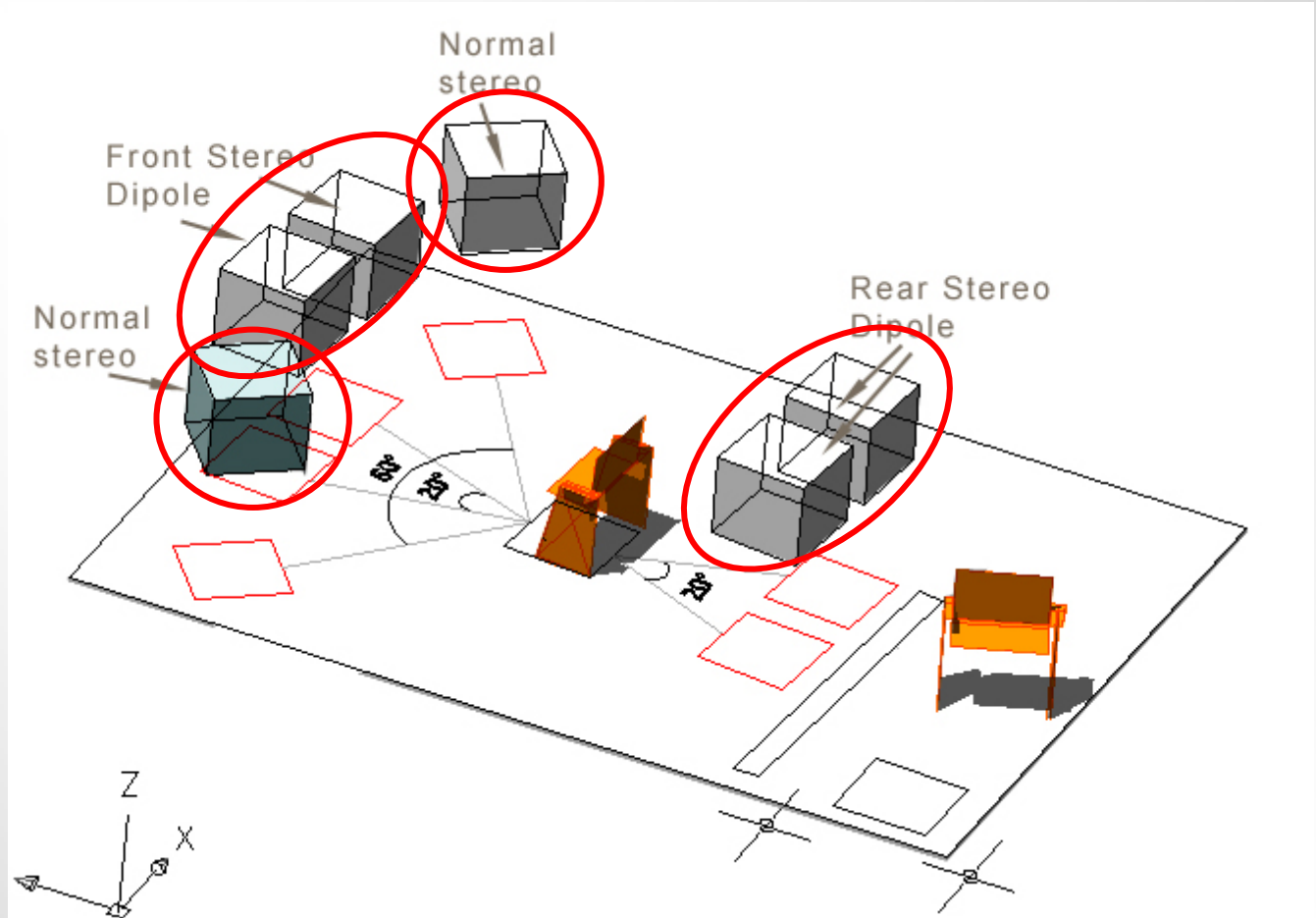


Speakers positioning

Genelec S30D

Dynaudio

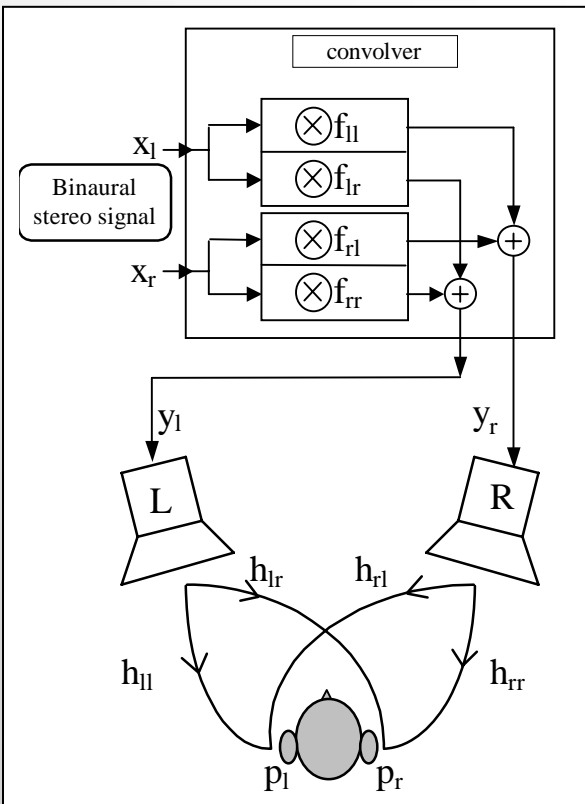
QSC AD-S82H



Design of cross-talk canceling filters



- First, a binaural measurement is made in front of the Stereo Dipole loudspeakers



- Then, the cross-talk cancelling filters are computed, so that their convolution with the measured impulse responses reduces to the identity matrix

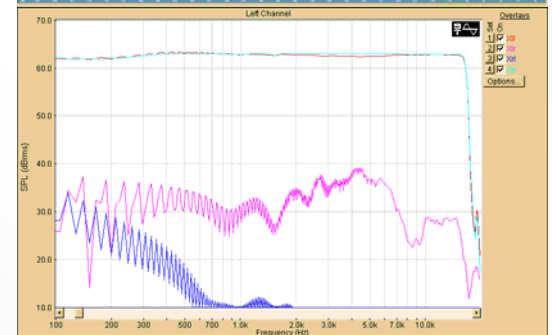
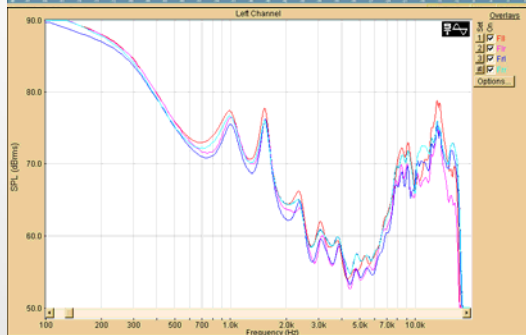
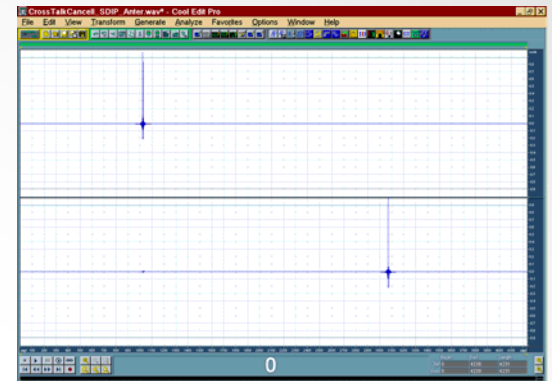
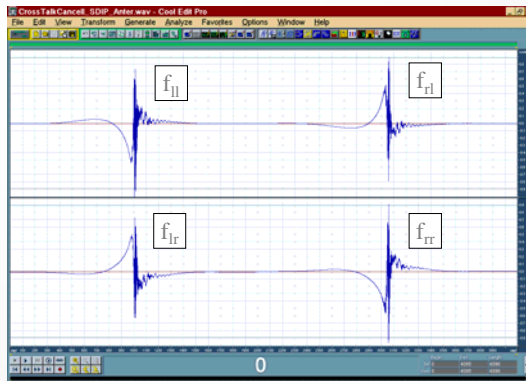
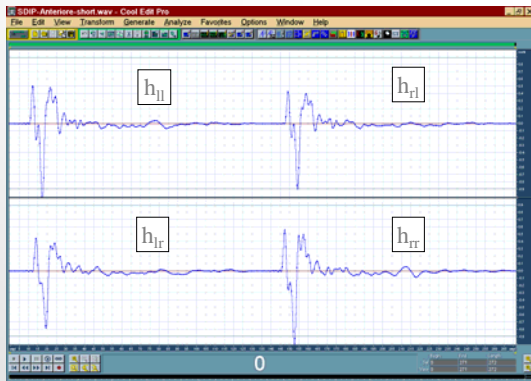


Stereo Dipole inverse filters

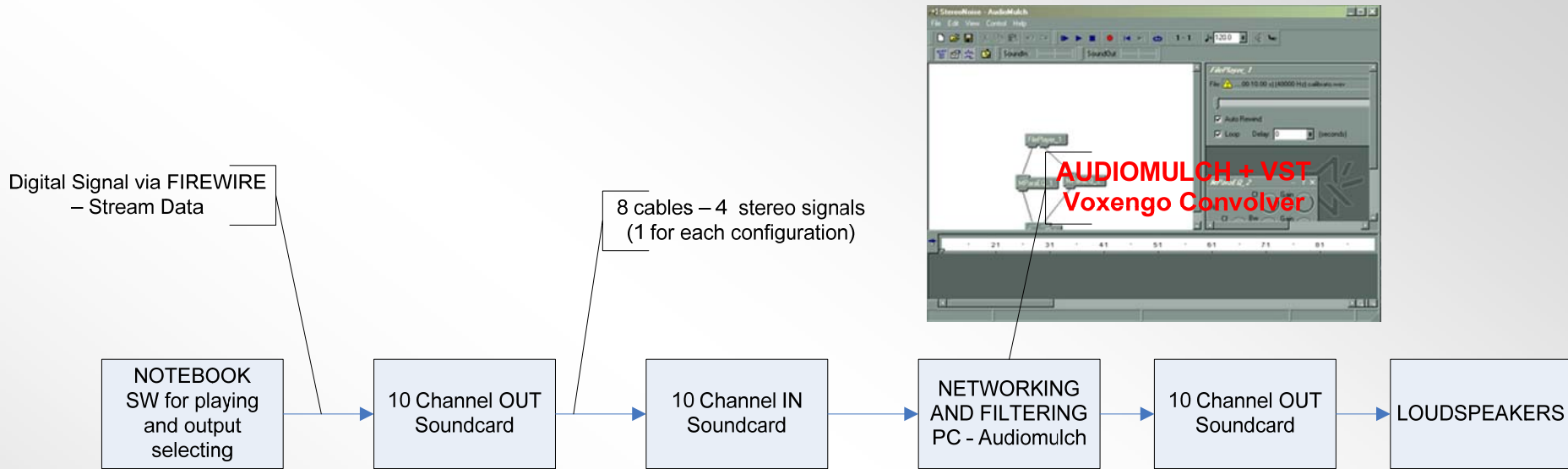
$$\begin{cases} f_{ll} = (h_{rr}) \otimes \text{InvDen} \\ f_{lr} = (-h_{lr}) \otimes \text{InvDen} \\ f_{rl} = (-h_{rl}) \otimes \text{InvDen} \\ f_{rr} = (h_{ll}) \otimes \text{InvDen} \\ \text{InvDen} = \text{InvFilter}(h_{ll} \otimes h_{rr} - h_{lr} \otimes h_{rl}) \end{cases}$$

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

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



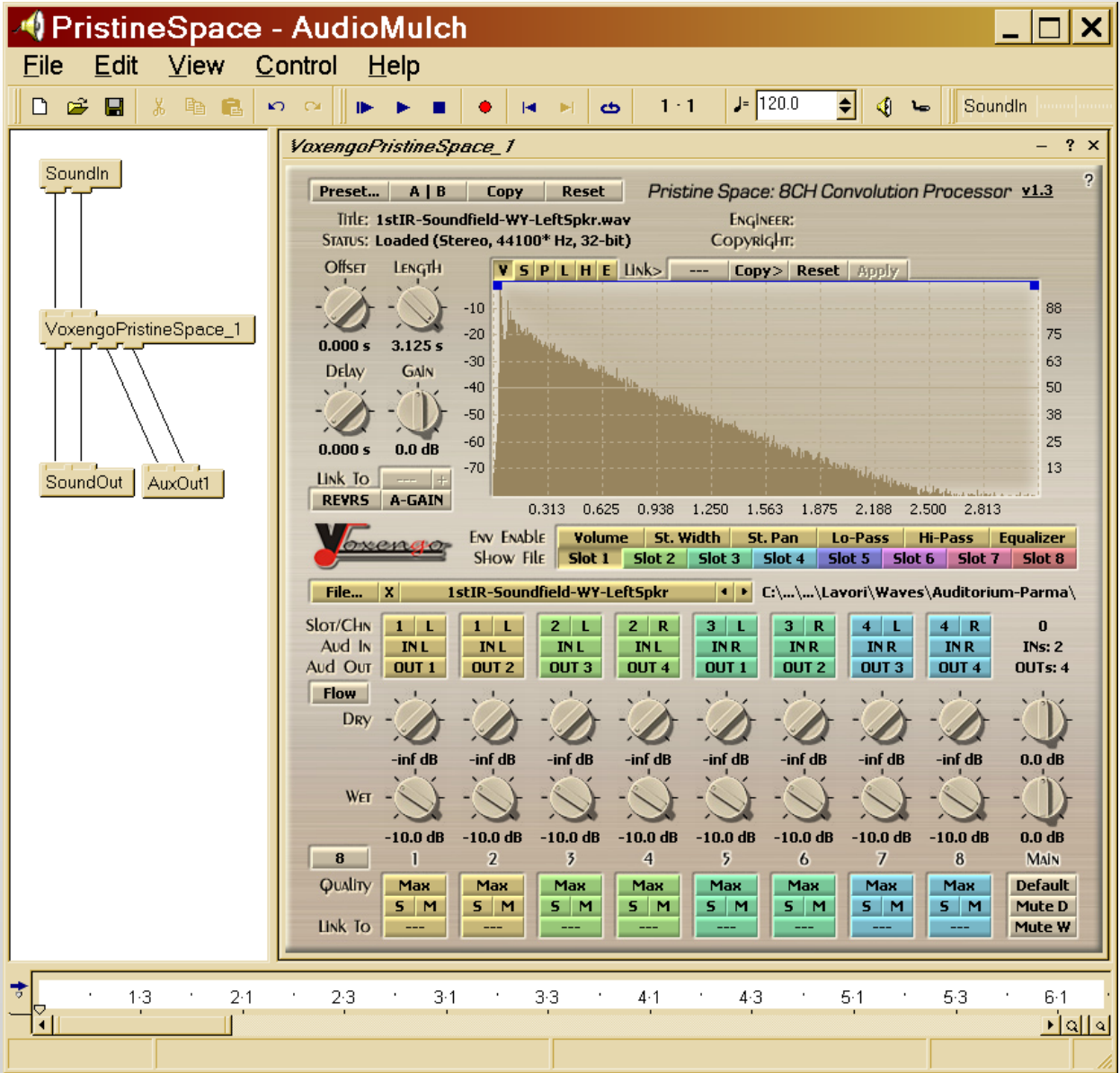
Hardware and Software setup



- Anechoic tracks preconvolved with IRs of the theatres at the beginning of the chain
- Filtering through Audiomulch in real time
- Voxengo Pristine Space VST plugin employed as multichannel convolver



The Voxengo Pristine Space multichannel convolver



- Filtering through Audiomulch in real time

- Voxengo Pristine Space VST plugin employed as multichannel convolver



Software for collecting the answers

VALUTAZIONE INDICE QUALITA' ACUSTICA STEREO

Situazione d'ascolto **1** 2 3 4 5 6 7 8 9 10

Fine

Stop

Qual'è la dimensione dell'ambiente che stai ascoltando?

Piccola Media Grande Molto Grande

0

Quanto è realistico il suono?

artificiale un poco artefatto realistico molto realistico

0

Quanto lontano senti l'artista in metri?

0 metri 25 75 100 metri

0

Label1

- Listening and simultaneous evaluation of 10 sounds, differing for position inside the theatre and system of reproduction



GOALS of the new subjective experiment

- Evaluation of spatiality and distance in auditoria through auralized tracks and different reproduction systems
- Evaluation of the best system for a realistic reproduction
- The subjective tests are undergoing in these months, the results will be published in 2005



Conclusions (preliminary yet)

- The loudspeakers are generally judged more natural than the headphones
- The frontal (single) stereo dipole is the system providing the better result/effort ratio: it works with a normal stereo system, it does not require a lot of computational power for running the inverse filters in real time, and is generally very natural and provides good localization
- Adding the second stereo dipole gives some advantages, but generally they are not worth the extra effort required (4-channels processing, etc.)
- The normal stereo gives very poor localization, and the perceived spectrum changes with the direction of the sound source, so the “colour” of the sound is not preserved
- It is very important to provide a comfortable user’s interface: large LCD screen, wireless mouse, comfortable seat, proper lighting, thermohygrometric confort.

The Future

Future improvement of the listening room

- Multichannel (surround) sound systems, including:
 - ITU 5.1 horizontal surround (2D) through Dolby AC3 or DTS
 - Advanced 7.1 horizontal surround (2D) through DTS-ES or Microsoft WMA
 - 1°-order periphonic (3D) Ambisonics (8 loudspeakers)
 - Hybrid Stereo Dipole + Ambisonics = Ambiophonics (3D)
- A single computer running both the playback tool and the filtering tool
- Large LCD widescreen display (also good for films!)
- Fanless, completely silent liquid-cooled computer
- 16 playback channels through an Apogee DAC 16 converter and an RME Hammerfall soundcard with 2 ADAT optical outputs

ITU 5.1 surround

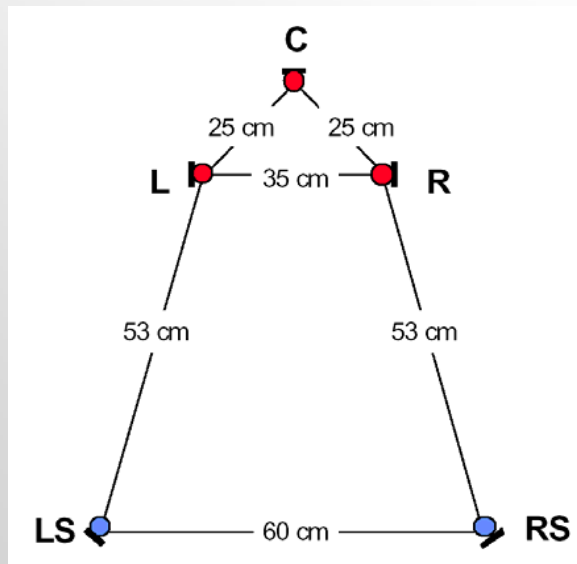
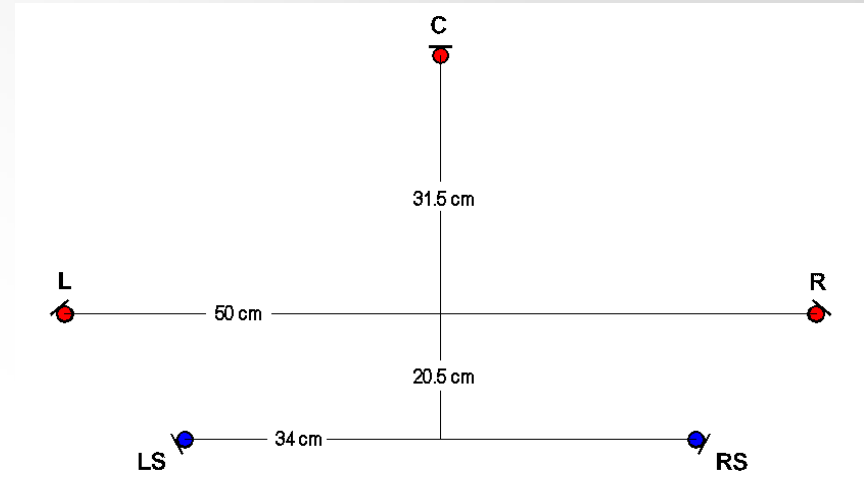
- Williams MMA

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

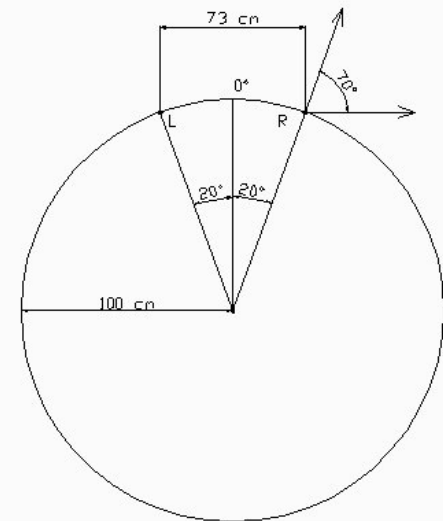
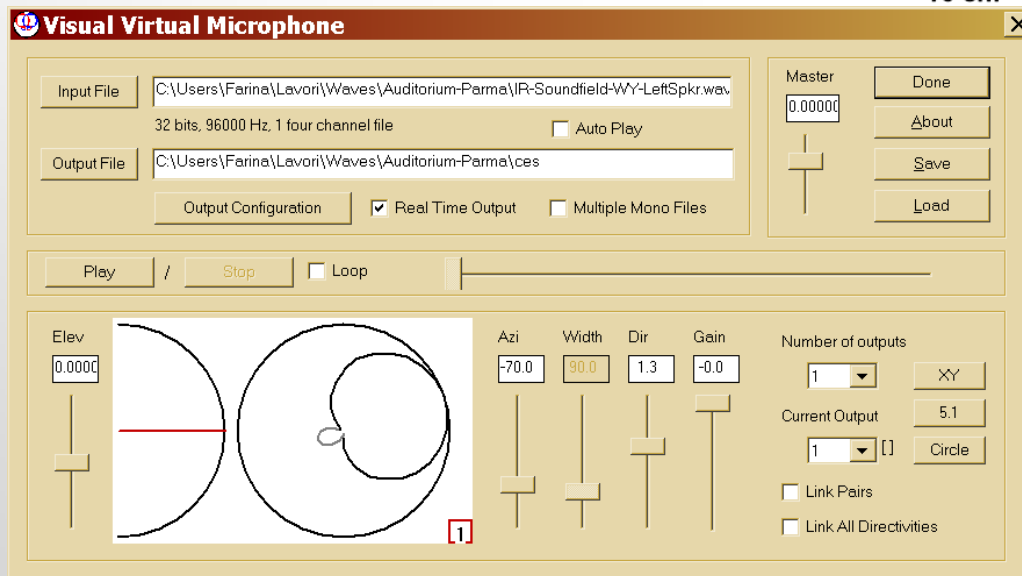
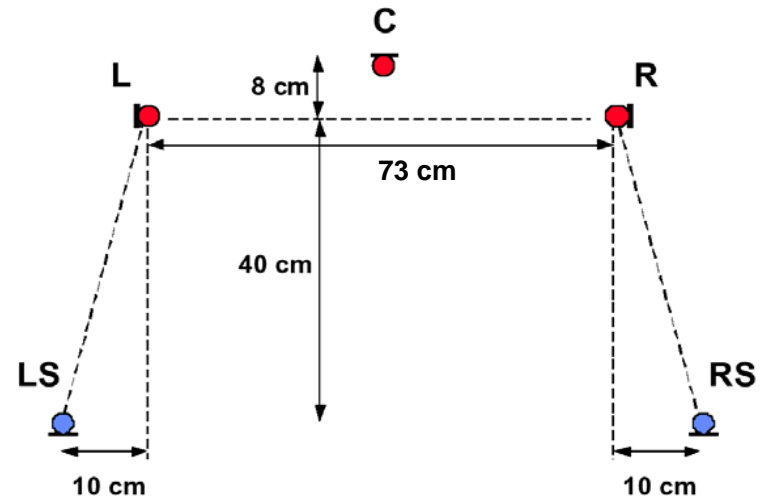
OCT

Schematic of the setup

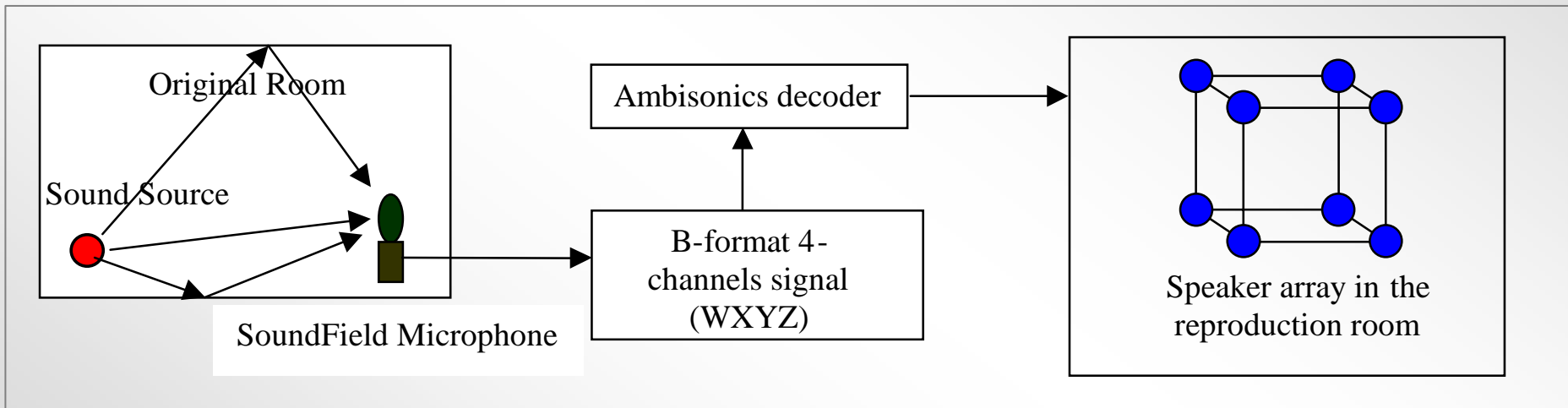
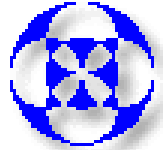
C : Cardioid, 0°

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

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



Ambisonics 3D 1st order



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

The Soundfield microphone

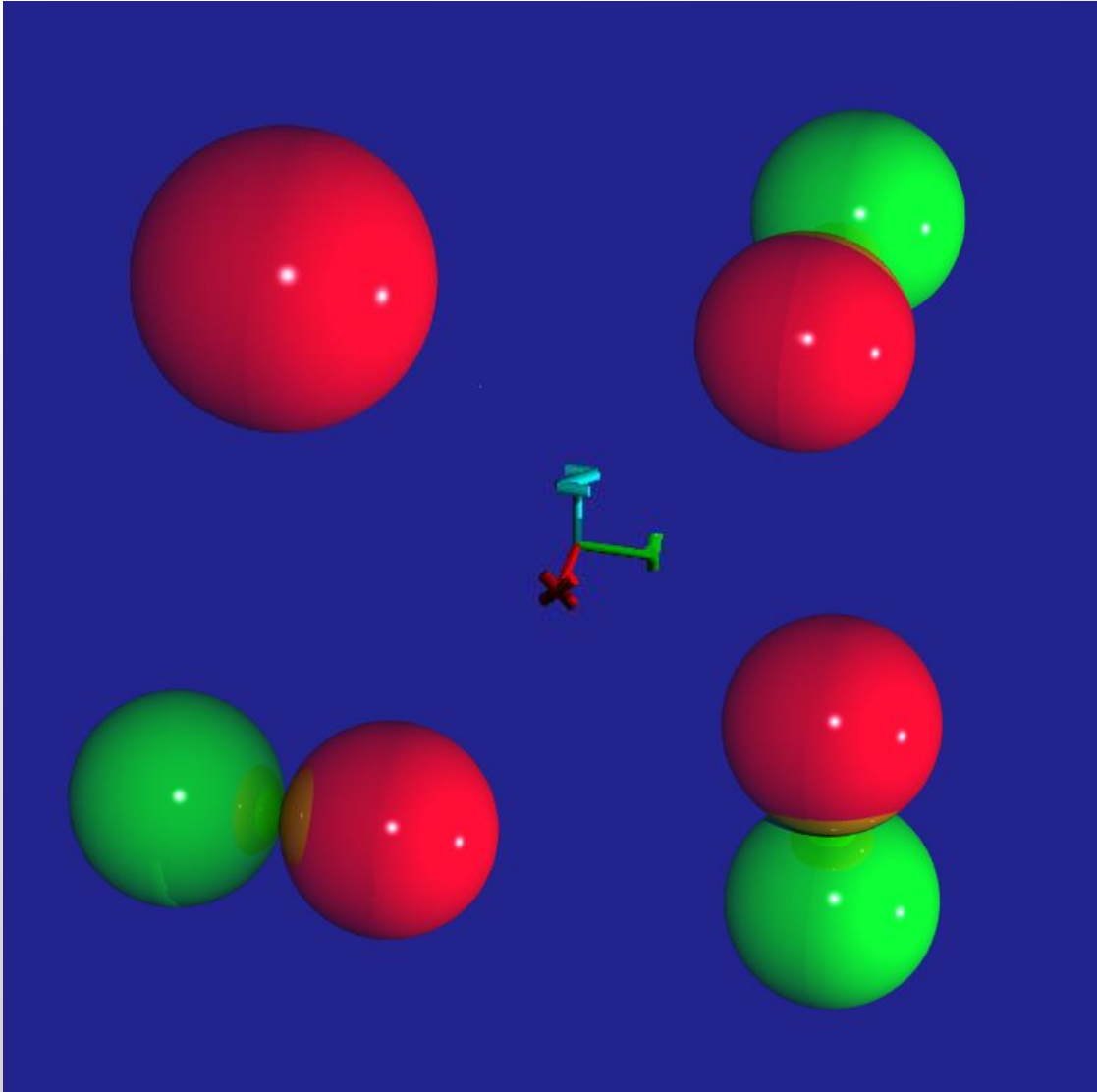


- This microphone is equipped with 4 subcardioid capsules, placed on the faces of a tetrahedron
- The signals are analogically processed in its own special control box, which derives 4 “B-format” signals
- These signals are:

- W : omnidirectional
- X,Y,Z : the three figure-of-eight microphones aligned with the ISO cartesian reference system

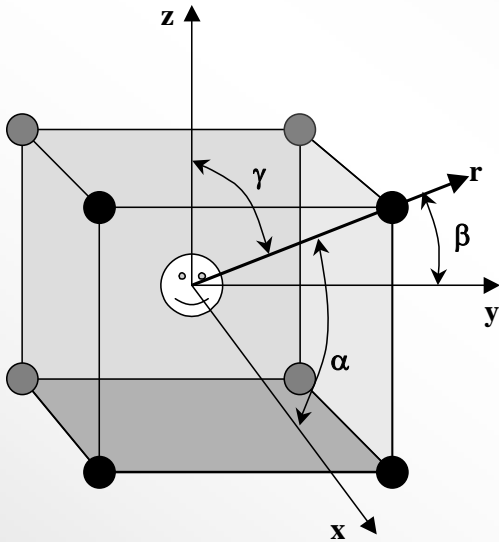


The B-format components



- Physically, W is a signal proportional to the pressure, XYZ are signals proportional to the three Cartesian components of the particle velocity
- when a sound wave impinges over the microphone from the “negative” direction of the x -axis, the signal on the X output will have polarity reversed with respect to the W signal

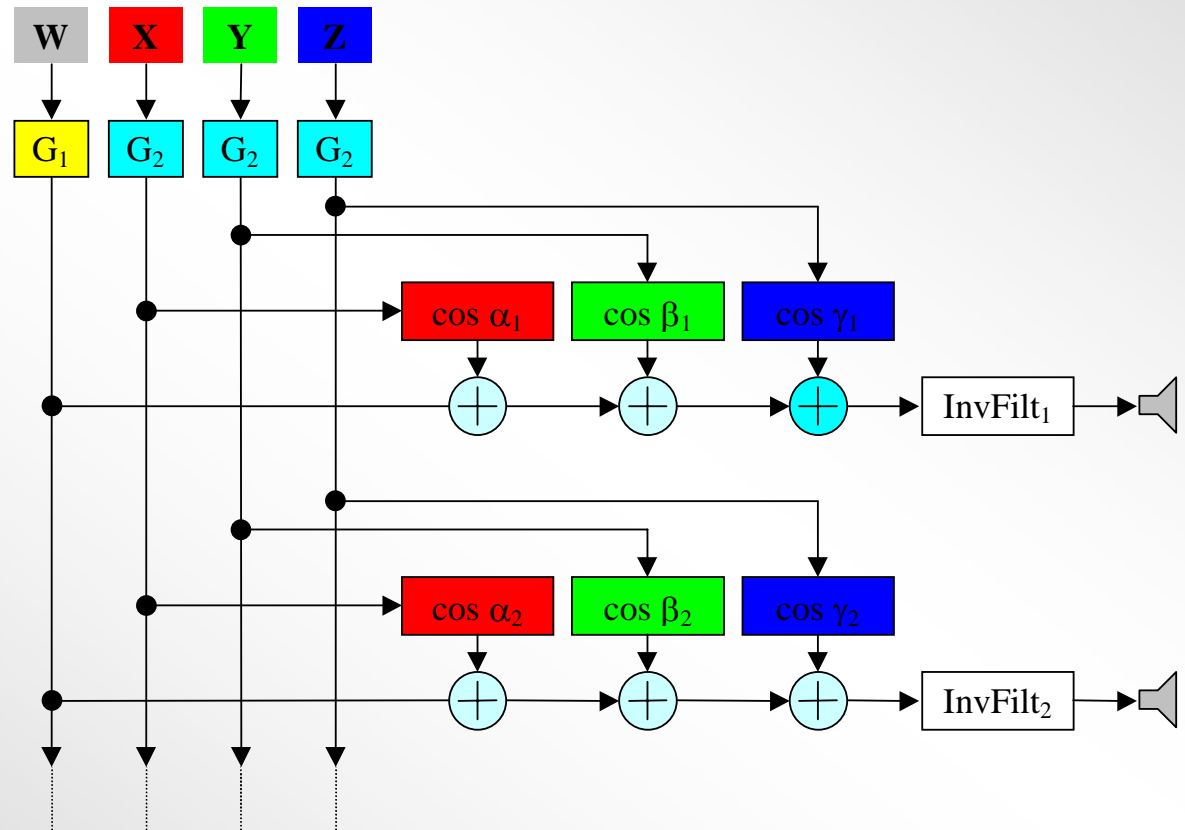
Ambisonics decoding



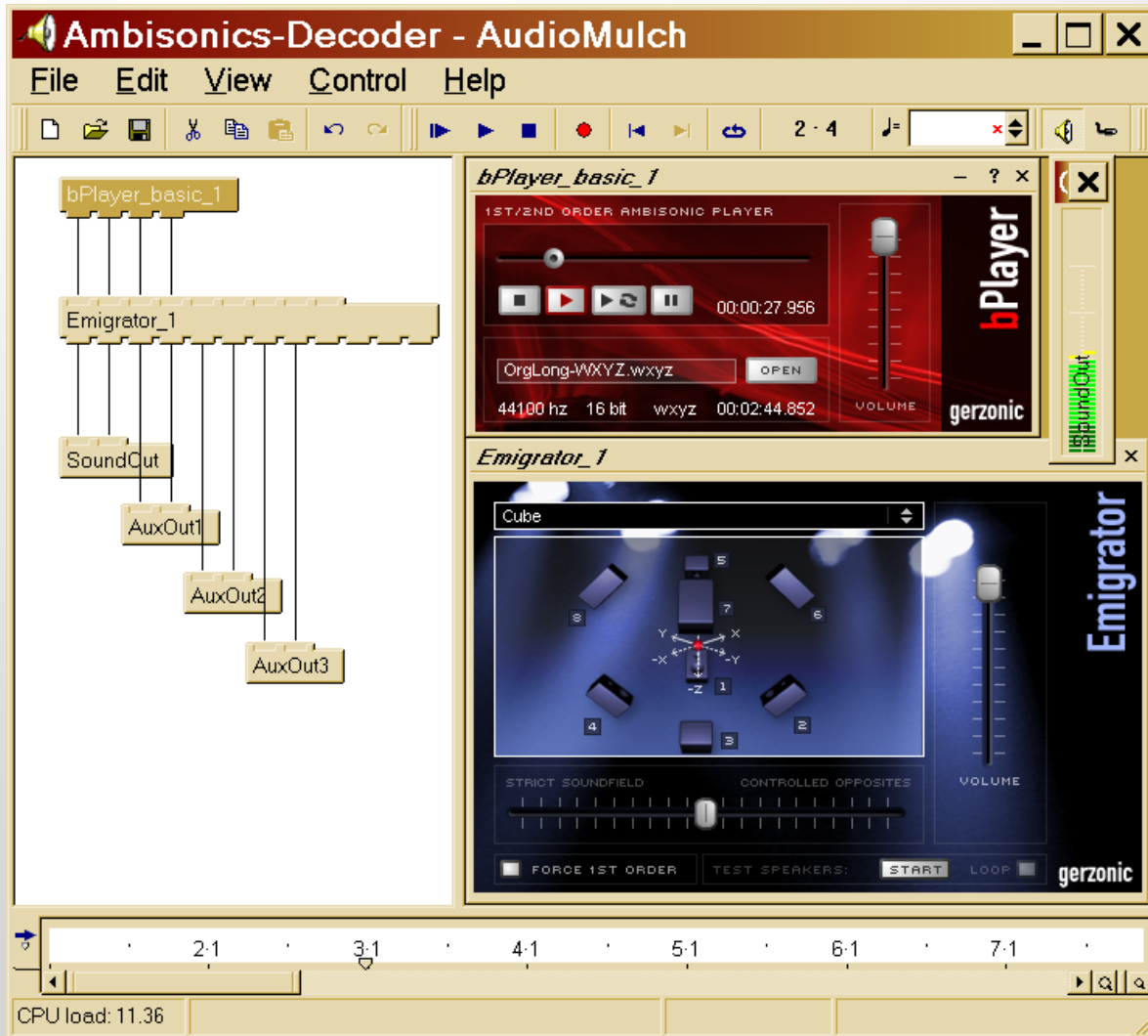
$$F_i = \frac{1}{2} \cdot [G_1 \cdot W + G_2 \cdot (X \cdot \cos(\alpha) + Y \cdot \cos(\beta) + Z \cdot \cos(\gamma))]$$

Each speaker feed is simply a weighted sum of the 4 B-format signals.

The weighting coefficients are computed by the cosines of the angles between the loudspeaker and the three Cartesian axes



A software Ambisonics decoder



Audiomulch VST
host

Gerzonic bPlayer

Gerzonic Emigrator

Rooms for Ambisonics 3D 1st order

ASK – Reggio Emilia



University of Ferrara

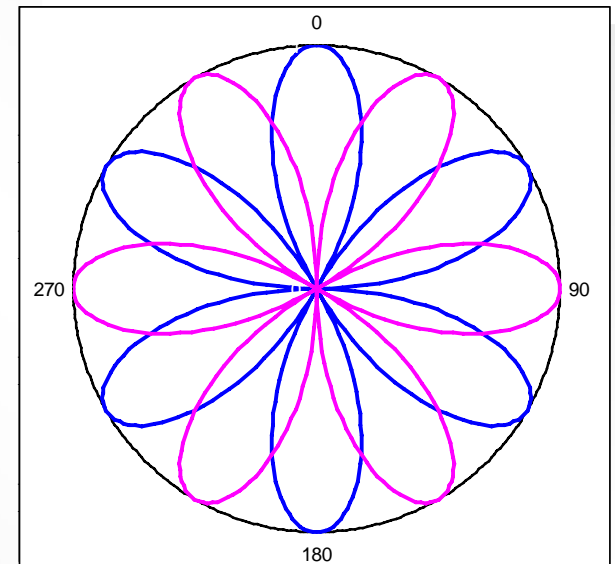
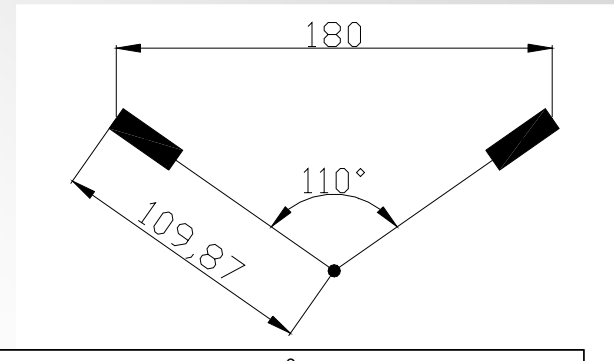


University of Bologna

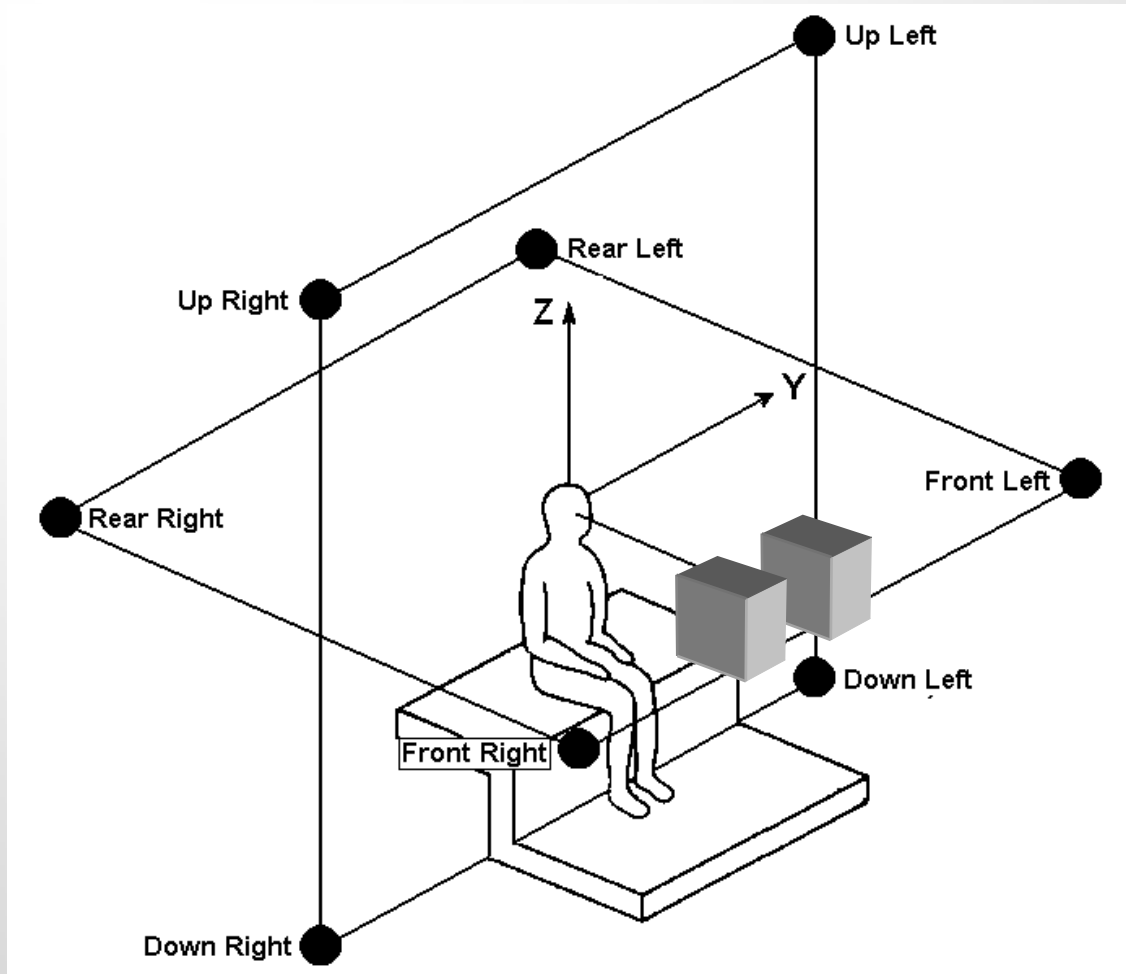
Virtual high-order microphones (M. Poletti)

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

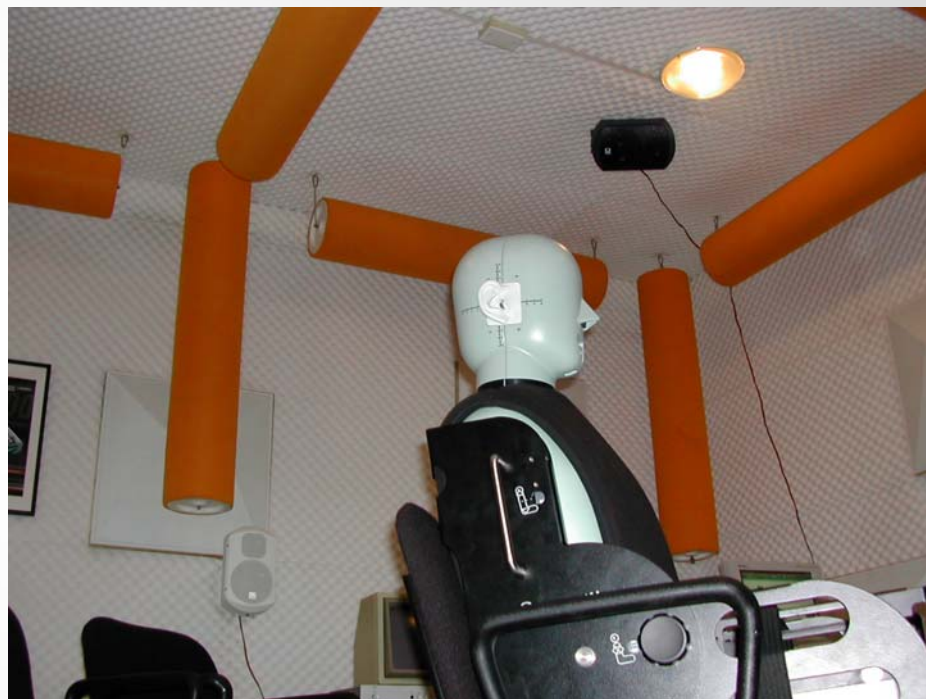
From these 36 impulse responses it is possible to derive the response of cylindrical harmonics microphones (2D Ambisonics) up to 5th order.



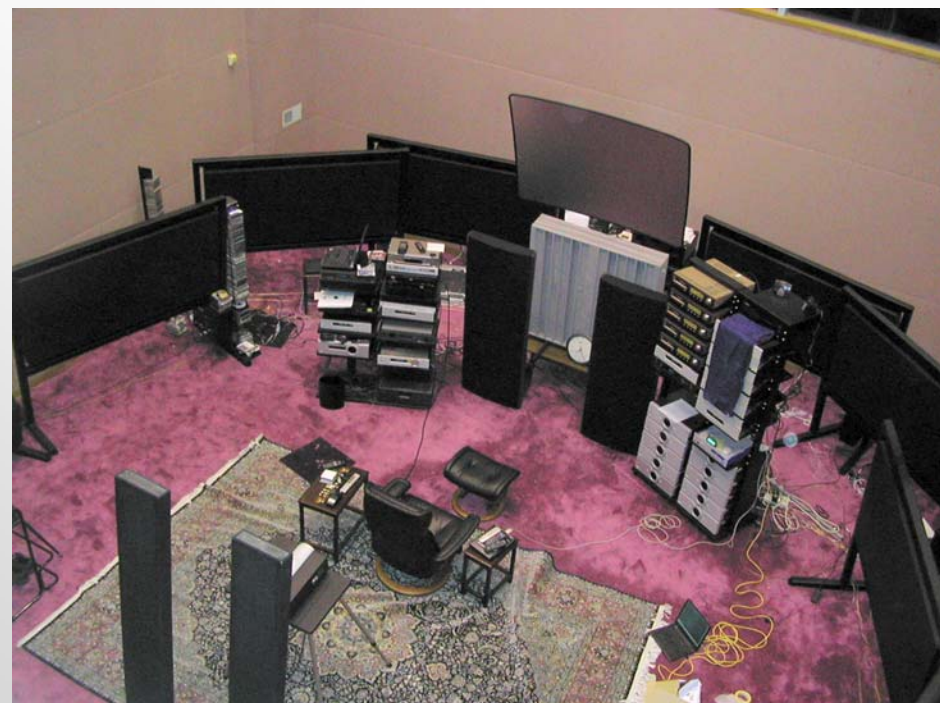
Ambiophonics 3D (10 loudspeakers):



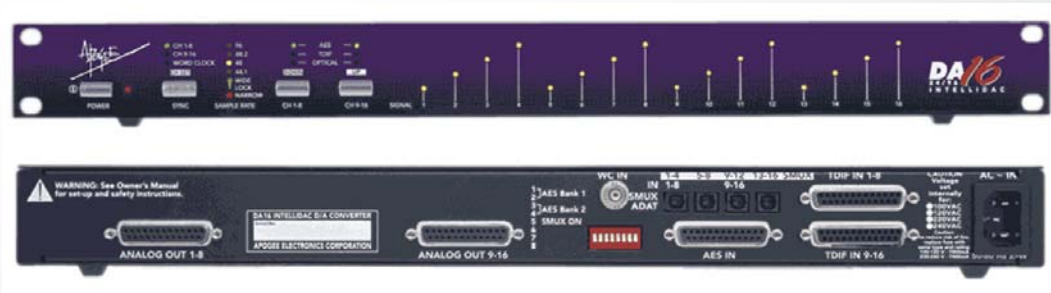
Ambiophonics Room at ASK



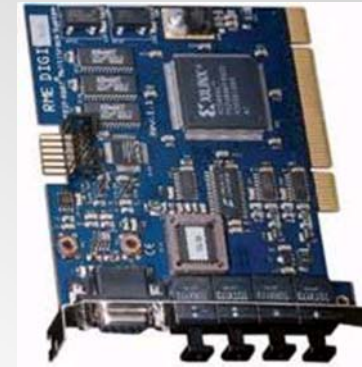
Ralph Glasgal's Ambiophonics Room at the Ambiophonics Institute



New hardware tools



Apogee DAC 16



RME Hammerfall soundcard



Liquid-cooled FutureClient PC



Toshiba 23" LCD TV/Monitor

Internet resources

All the papers previously published by Angelo Farina can be downloaded from his personal web site:

www.angelifarina.it

The software system employed for this research is based on the following modules:

Adobe Audition (**www.adobe.com**)

Aurora Plugins (**www.aurora-plugins.com**)

Audio Mulch (**www.audiomulch.com**)

Voxengo Pristine Space (**www.voxengo.com**)

