



116th AES Convention
Berlin, 8-11 May 2004



Measurement and 5.0 rendering of spatial impulse responses of rooms



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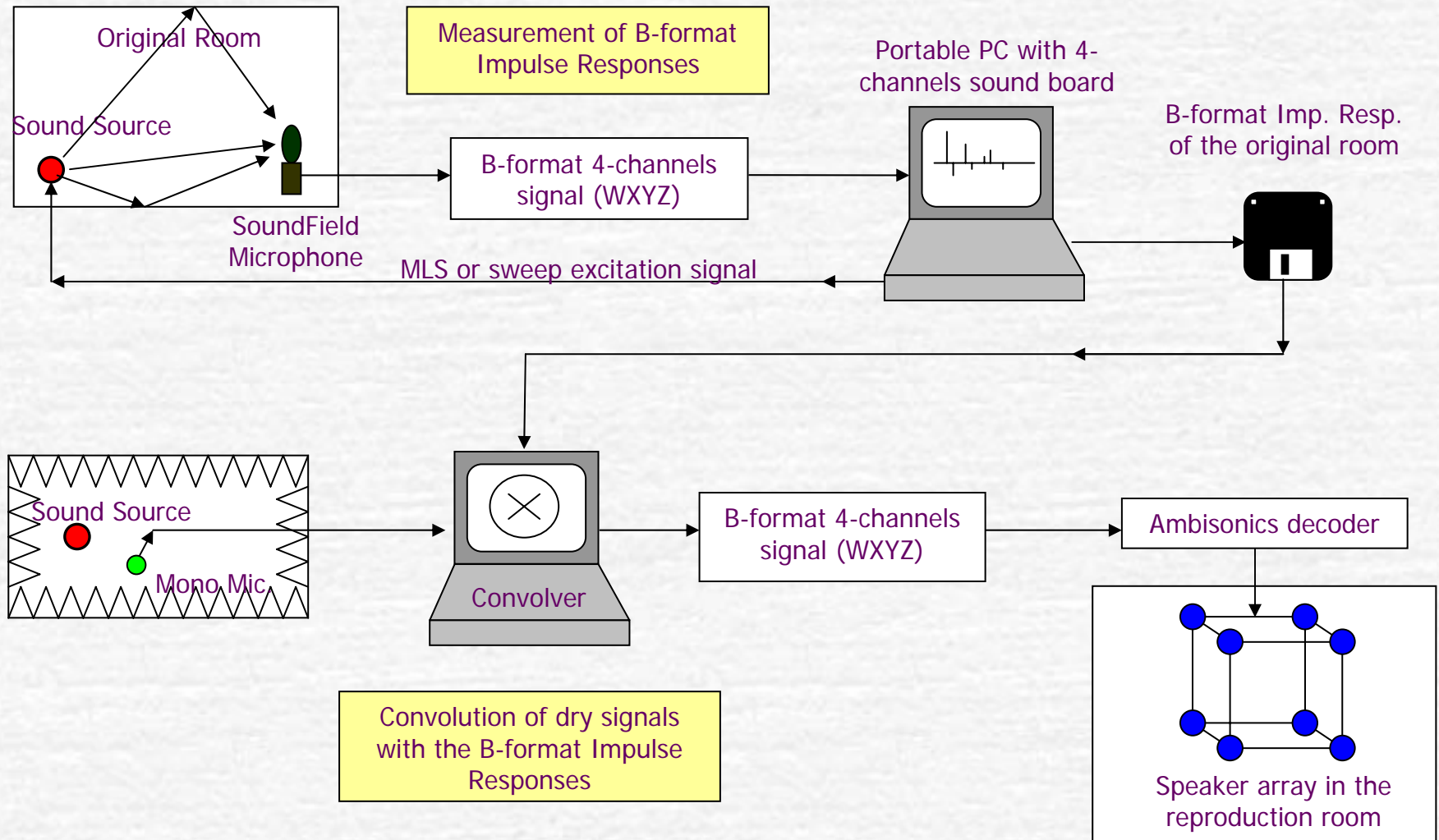
[HTTP://pcfarina.eng.unipr.it](http://pcfarina.eng.unipr.it) - mail: angelo.farina@unipr.it

Topics

- This presentation is a tribute to [M. Gerzon](#), who had foreseen 3D impulse response measurements and 3D Auralization obtained by convolution.
- Comparison between Auralizations based on calculated and measured IRs (e.g. Theatre "La Fenice", Venice)
- The advantages (and disadvantages) of employing measured IRs
- Possible approaches to Auralization over ITU 5.0 "surround" systems



Concept (Gerzon, 1975)



Why do we measure these 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 (5.1) and tomorrow’s musical productions
3. Auralization in special listening rooms can be performed 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, Tronchin measured binaural impulse responses

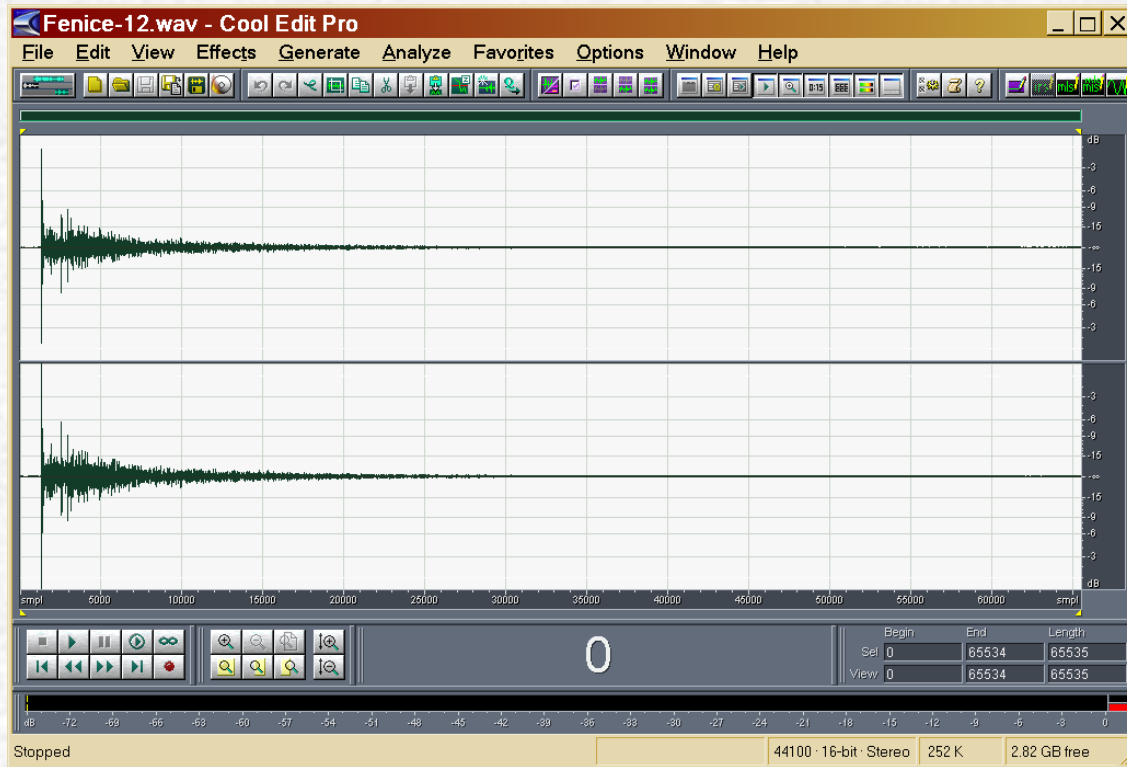


Impulse Responses of La Fenice

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

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Numerical simulation vs. measurement



Prelude 1° act “La Traviata”
by G.Verdi

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

Stop

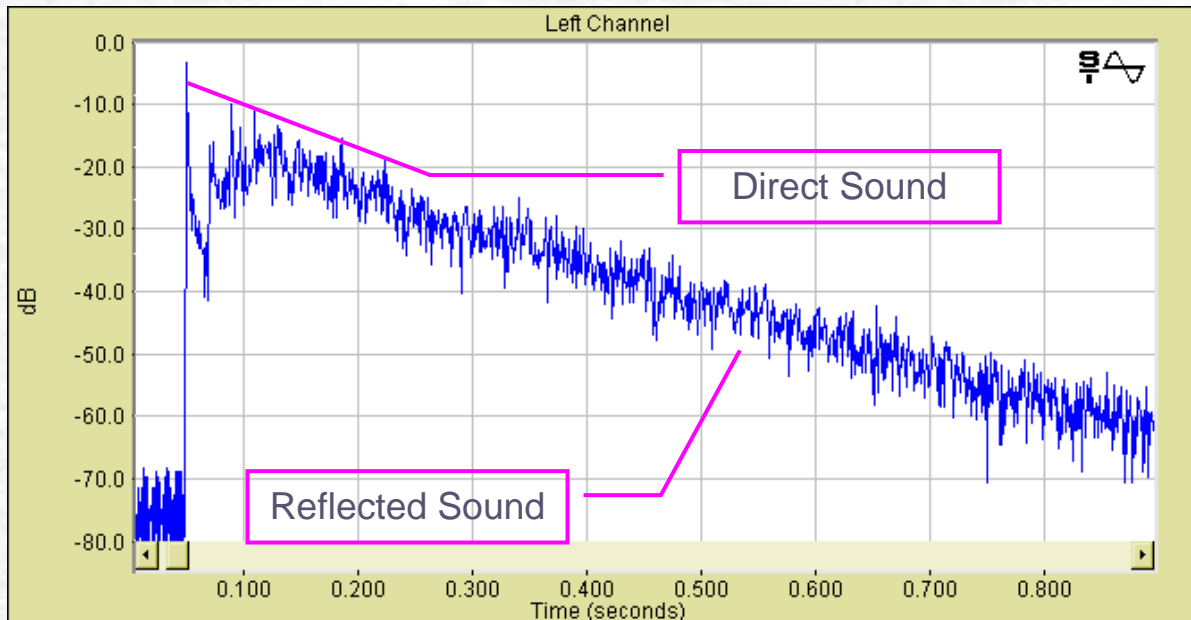
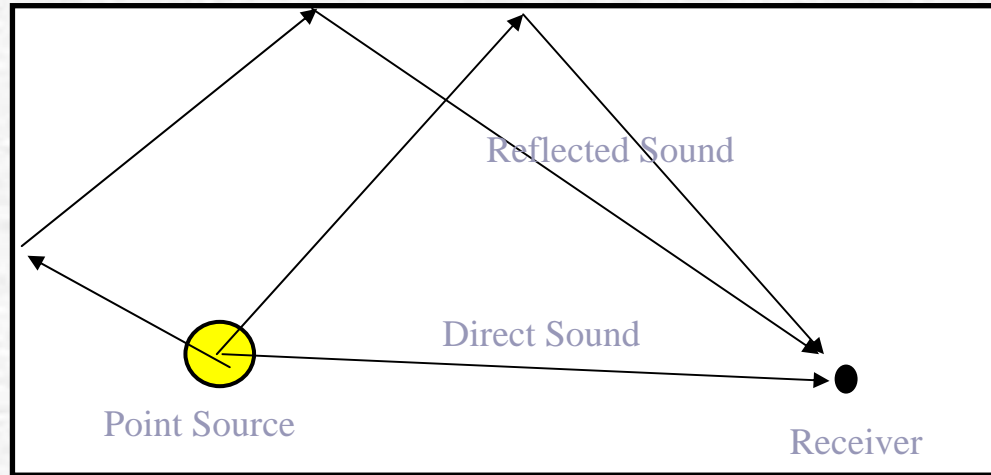


Advanced IR capture and rendering (project)

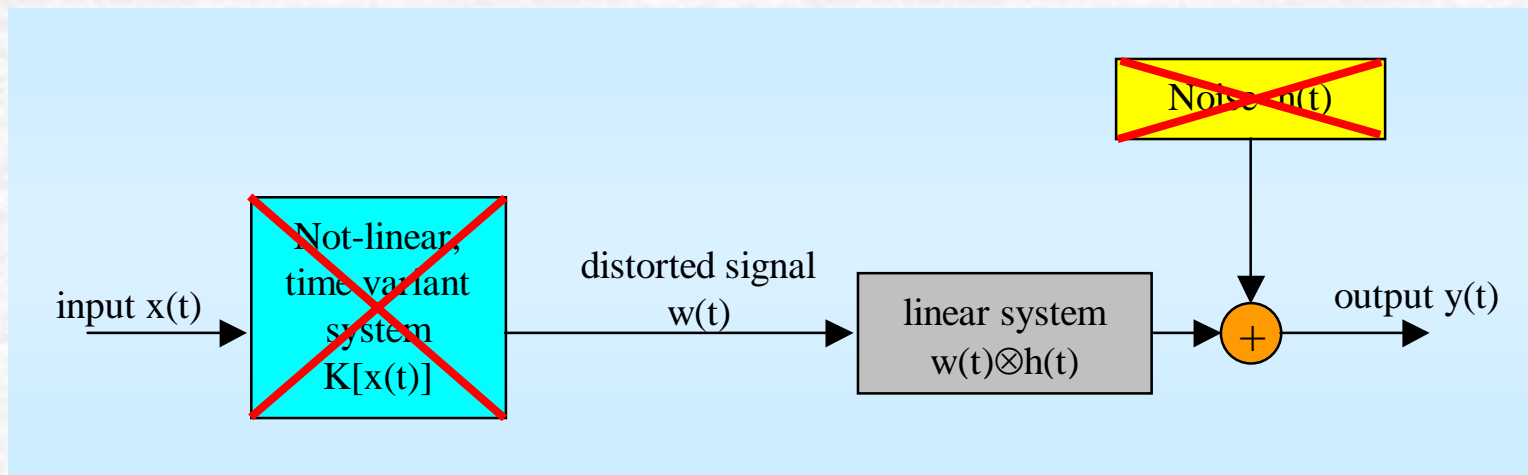
- Description of the measurement technique
- ~~• Analysis of some acoustical parameters of some theaters measured~~
- Description of the processing methods to be employed for transforming the measured data in audible reconstructions of the original spaces
- ~~• Description of the usage of the measured 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 $n(t)$.



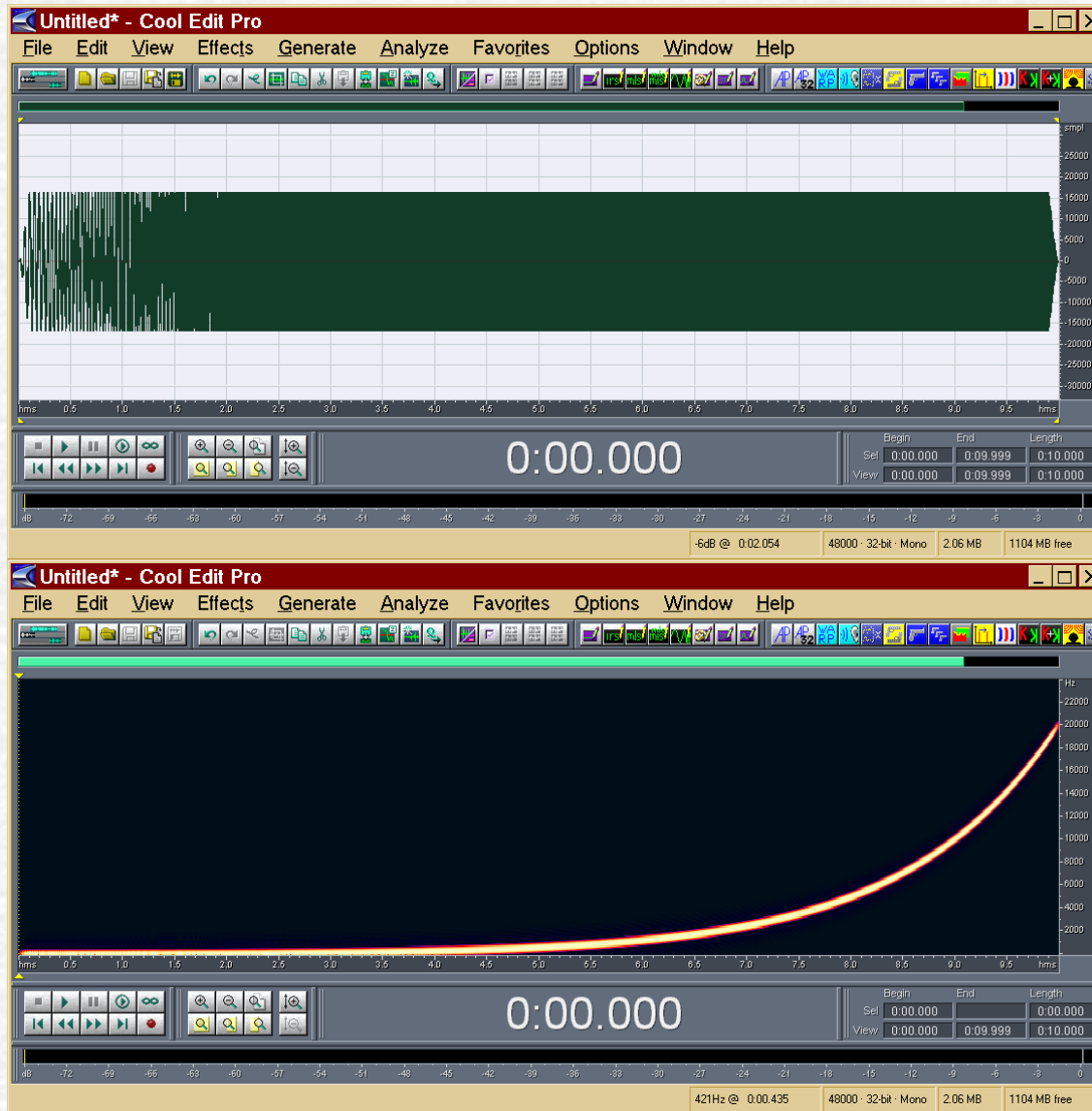
Test signal: Log Sine Sweep

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

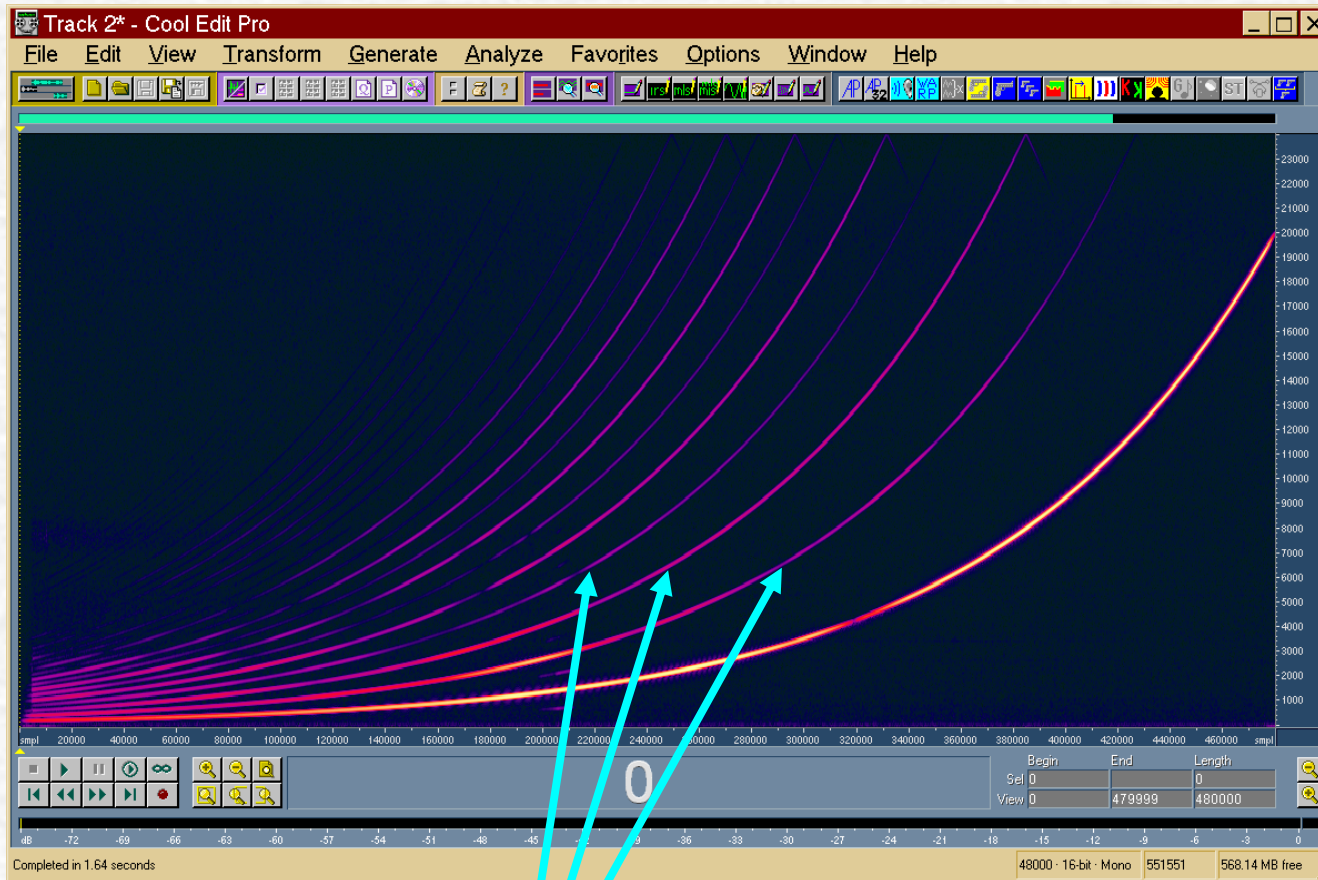
$$x(t) = \sin \left[\frac{2 \cdot \pi \cdot f_1 \cdot T}{\ln \left(\frac{f_2}{f_1} \right)} \cdot \left(e^{\frac{t}{T} \cdot \ln \left(\frac{f_2}{f_1} \right)} - 1 \right) \right]$$



Test Signal – $x(t)$



Measured signal - $y(t)$

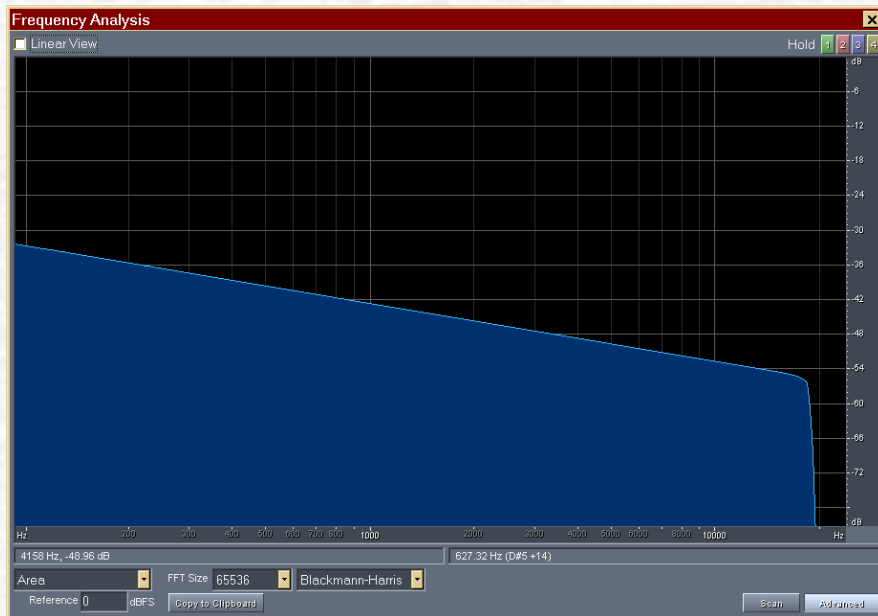


- ❖ The not-linear behaviour of the loudspeaker causes many harmonics to appear

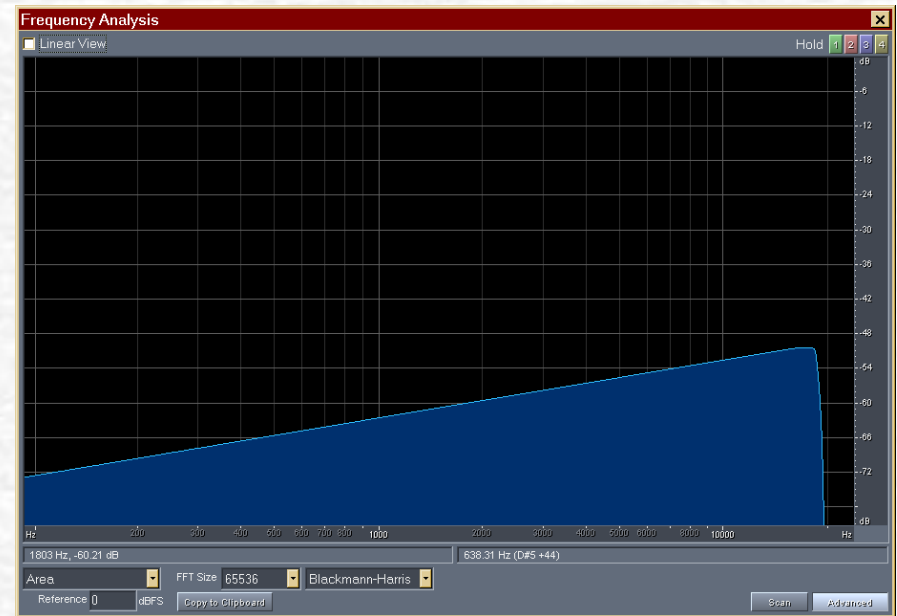


Deconvolution of Log Sine Sweep

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



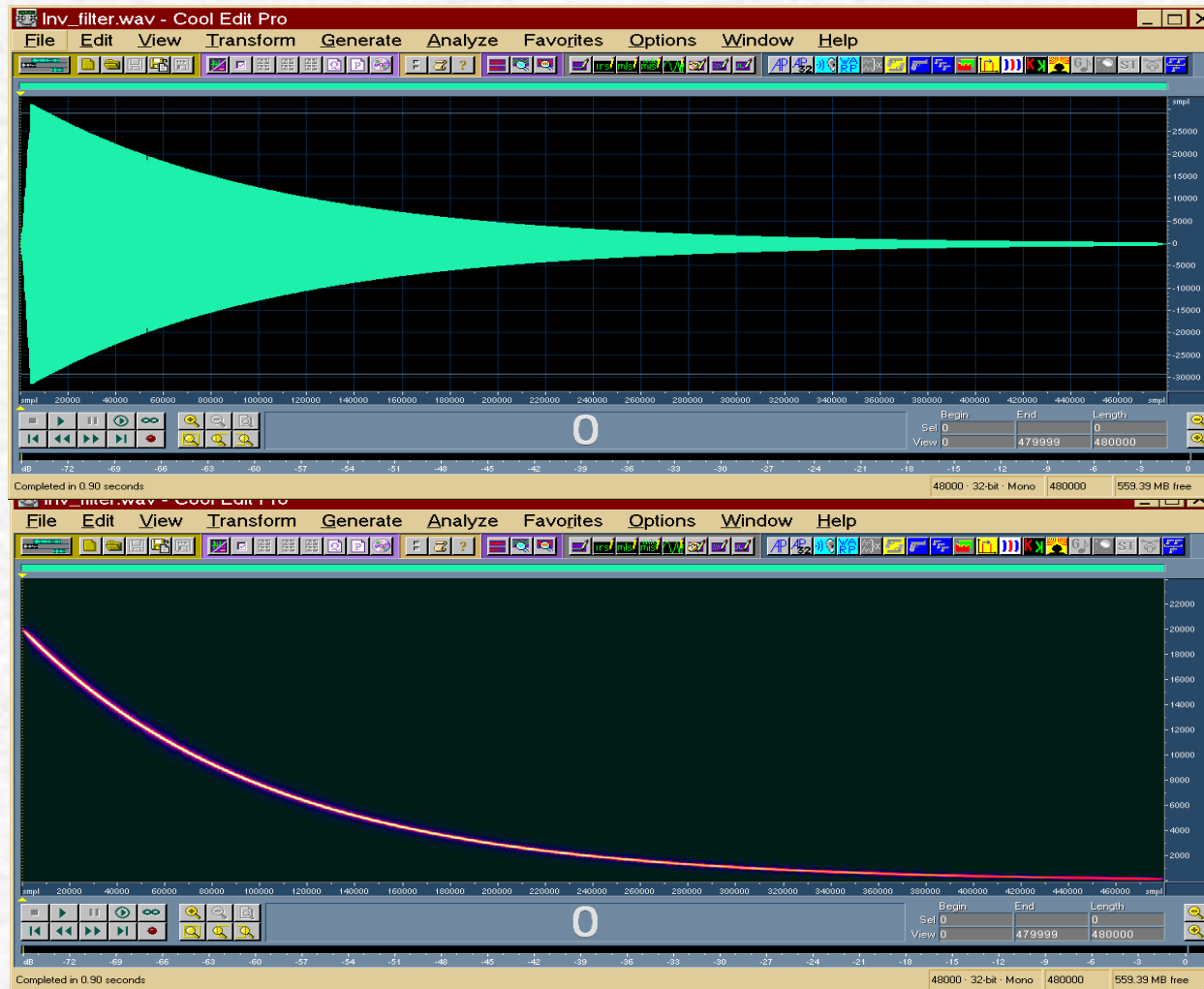
Test Signal $x(t)$



Inverse Filter $z(t)$



Inverse Filter – $z(t)$



The deconvolution of the IR is obtained convolving the measured signal $y(t)$ with the inverse filter $z(t)$



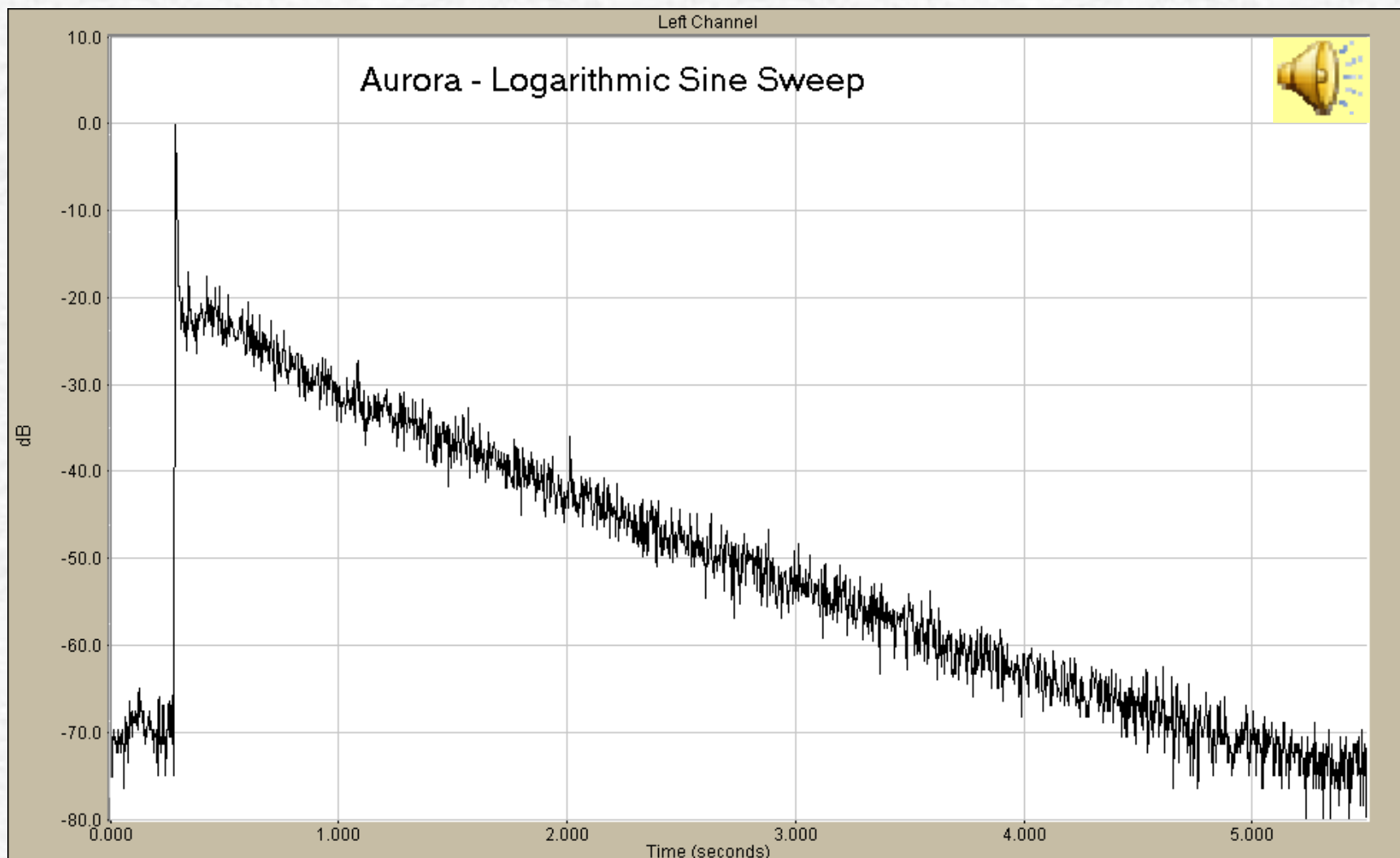
Result of the deconvolution



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



Maximum Length Sequence vs. Sweep



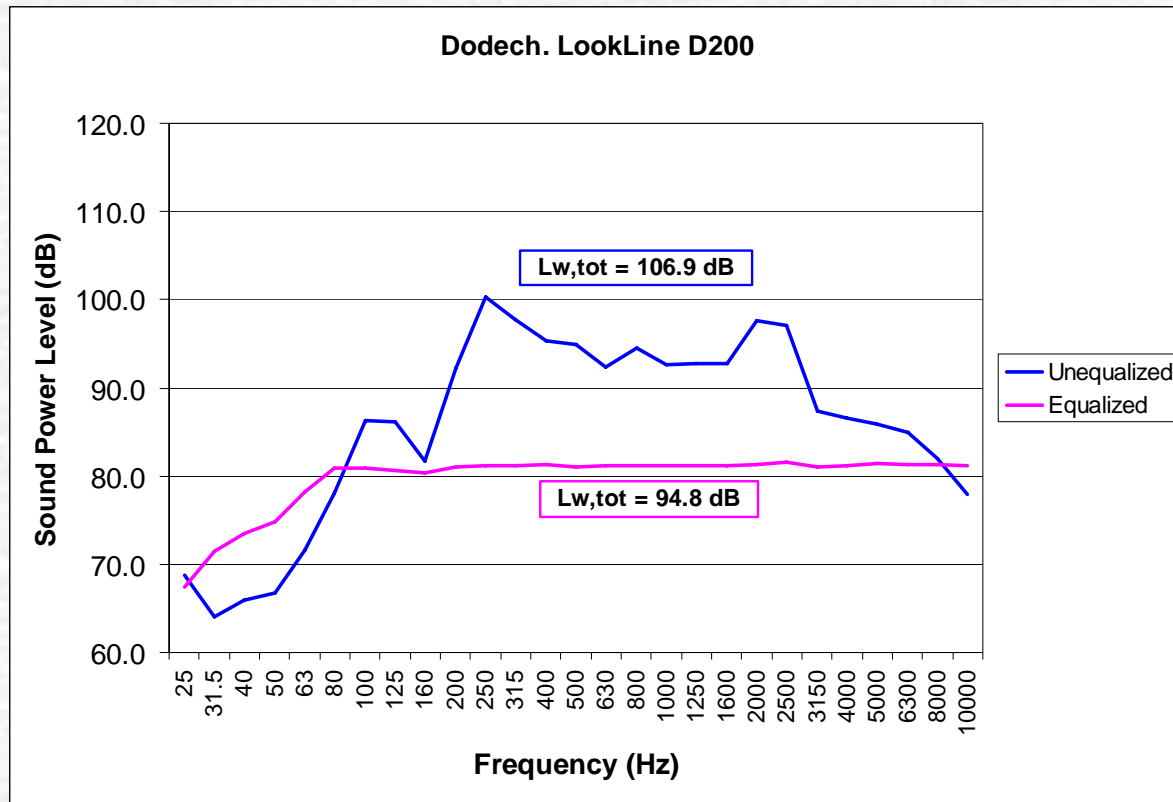
Measurement Setup

- 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



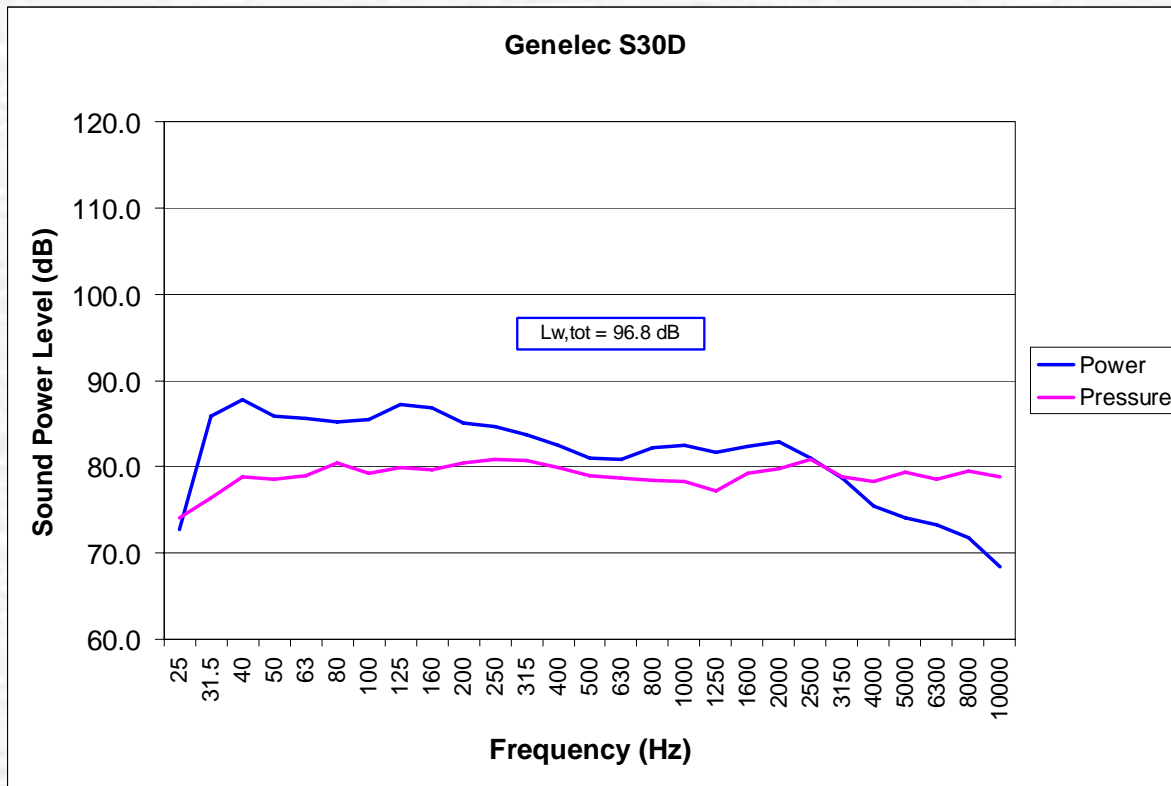
Transducers (sound source #1)

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



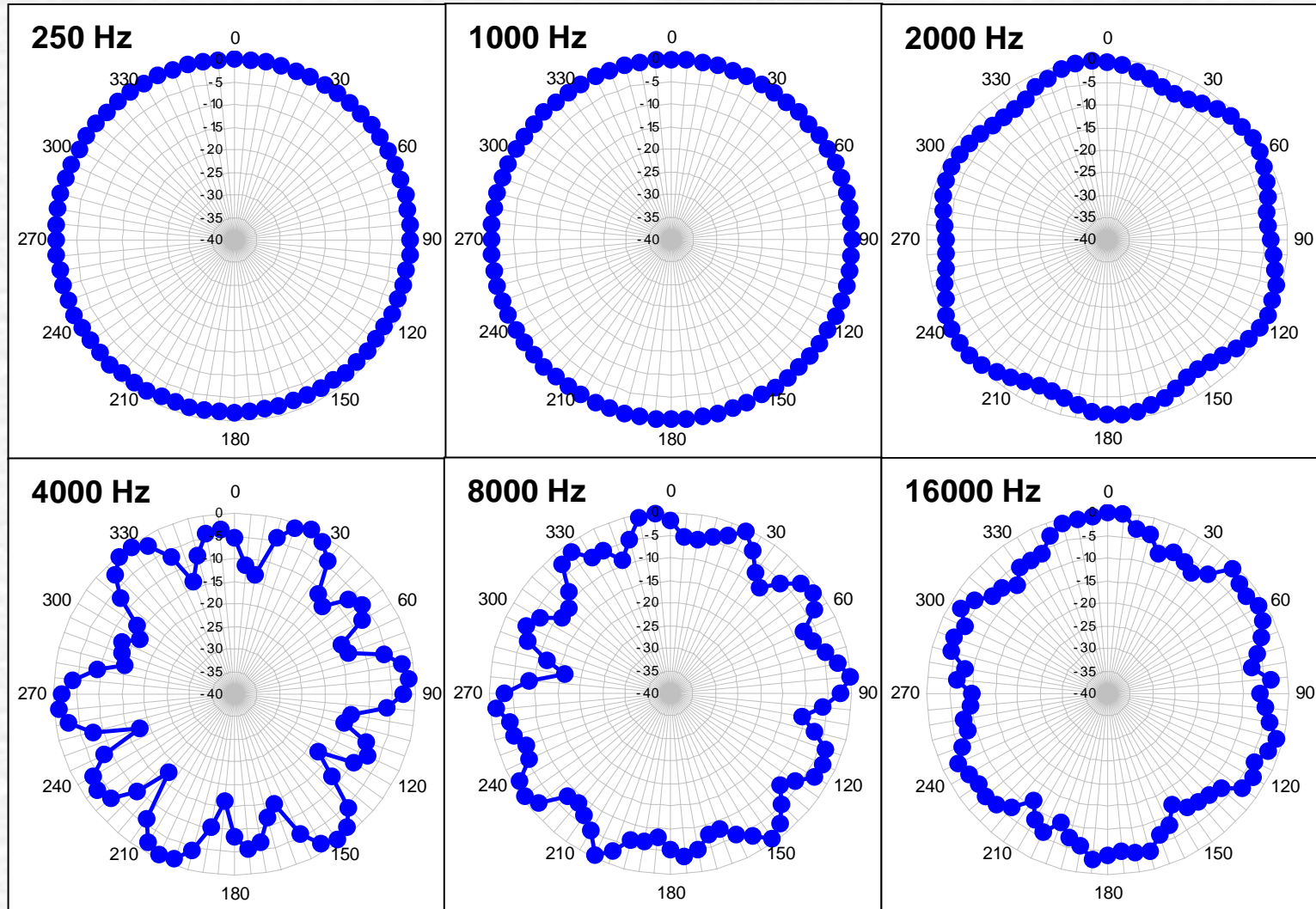
Transducers (sound source #2)

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



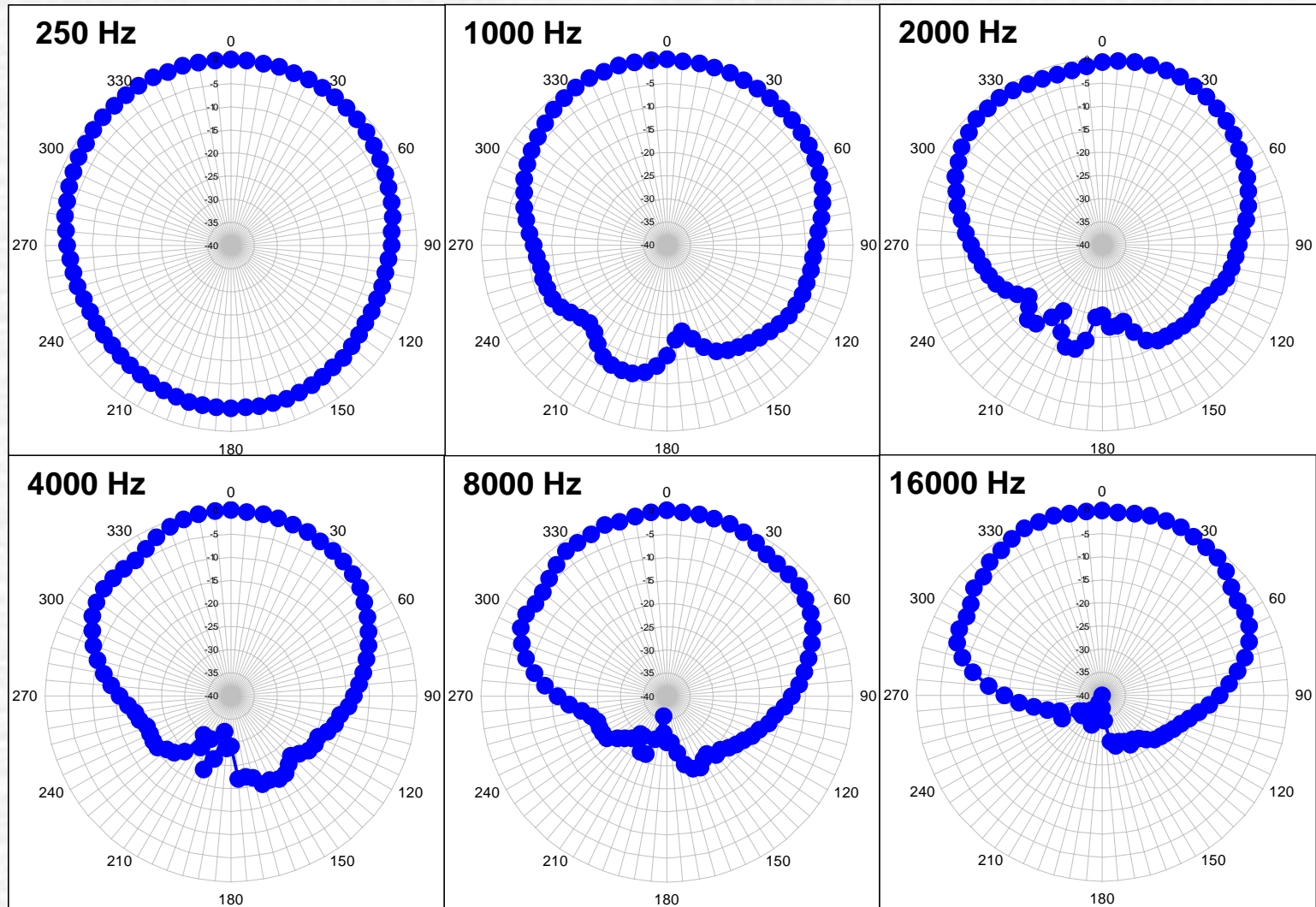
Directivity of transducers

LookLine D200 dodechaedron



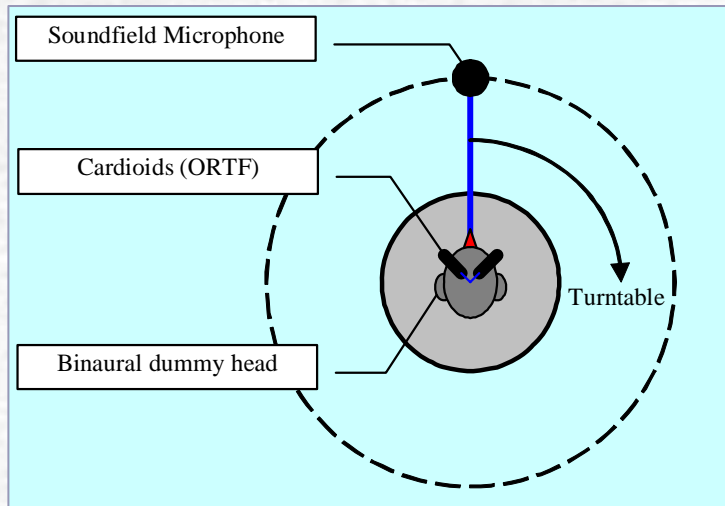
Directivity of transducers

Genelec S30D reference studio monitor



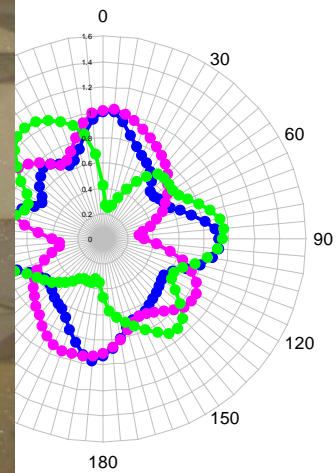
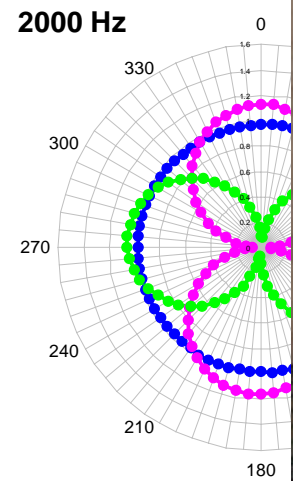
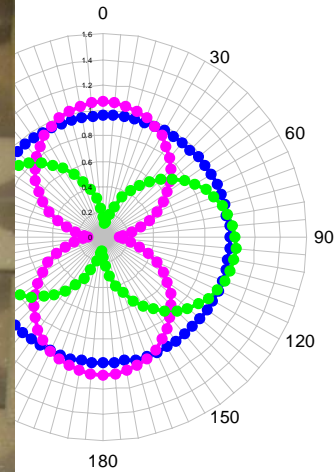
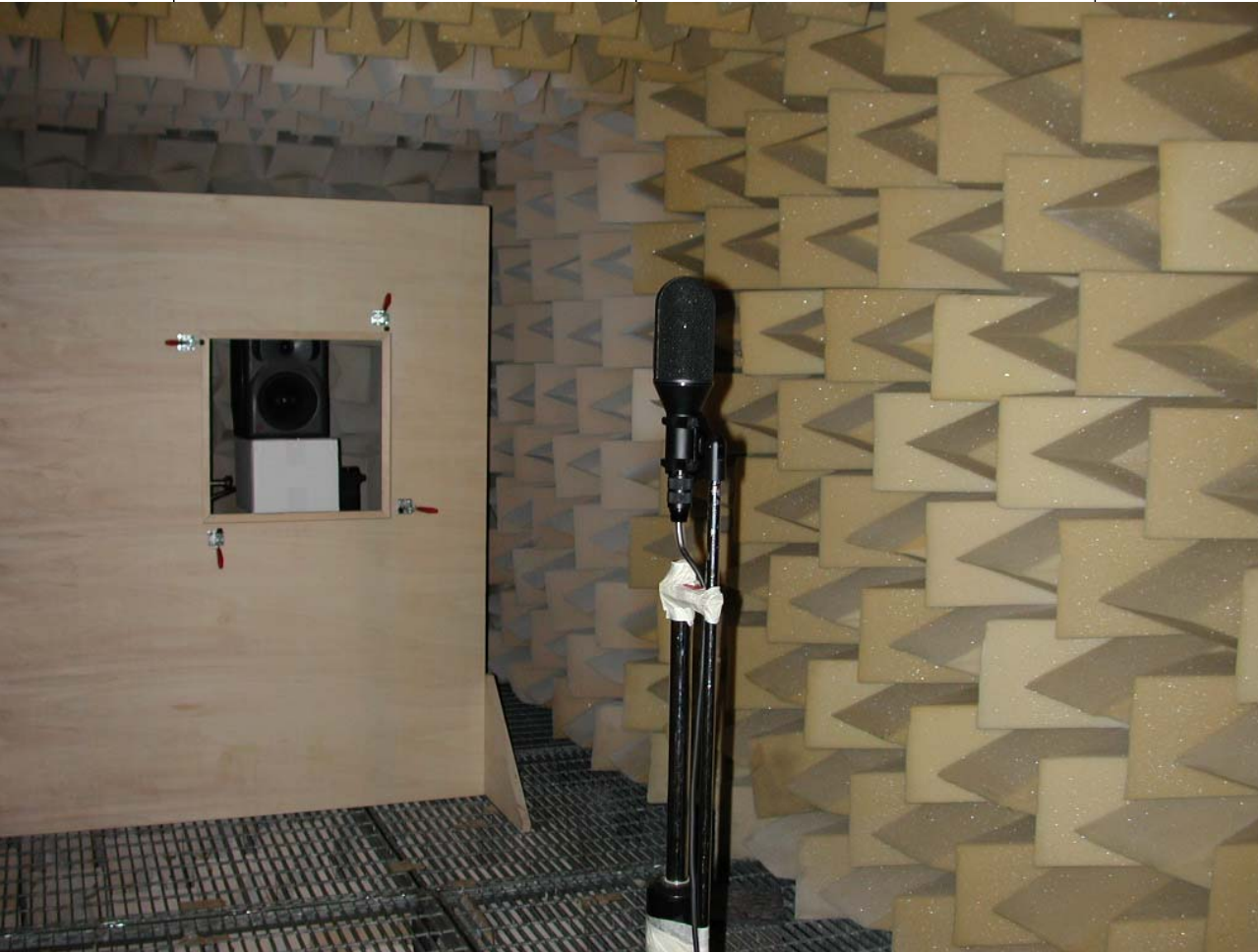
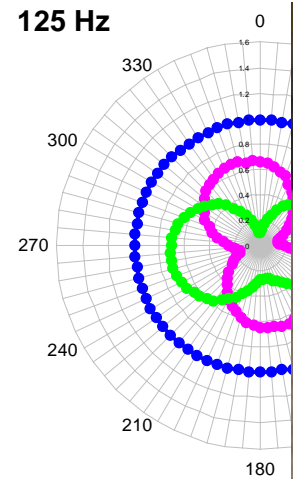
Transducers (microphones)

- 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)



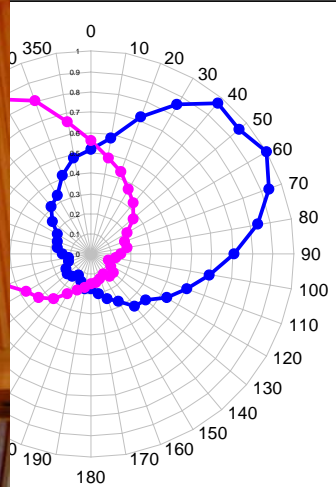
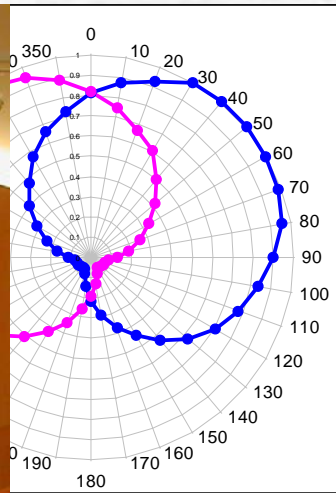
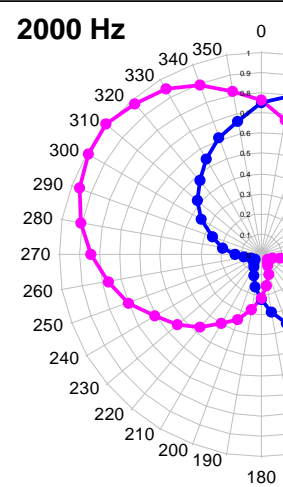
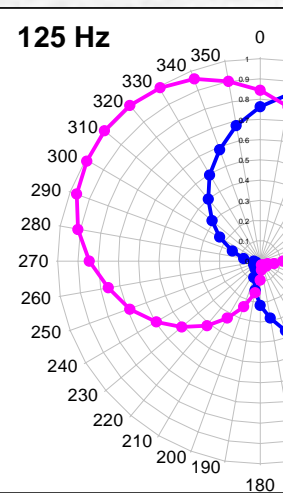
Directivity of transducers

Soundfield ST-250 microphone



Directivity of transducers

Neumann K-140 (ORTF Cardioids)



Other hardware equipment

- Rotating Table:
 - Outline ET-1

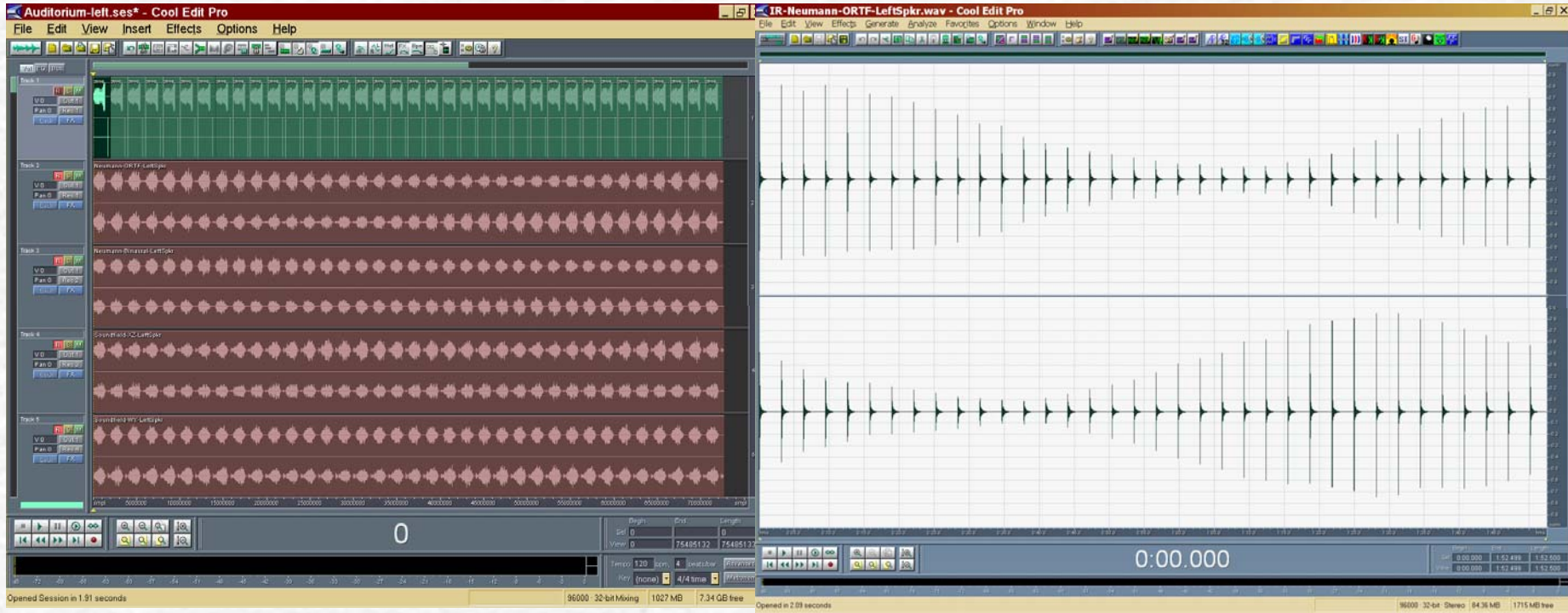


- Computer and sound card:
 - Signum Data Futureclient P-IV 1.8 GHz
 - Aardvark Pro Q-10 (8 ch., 96 kHz, 24 bits)

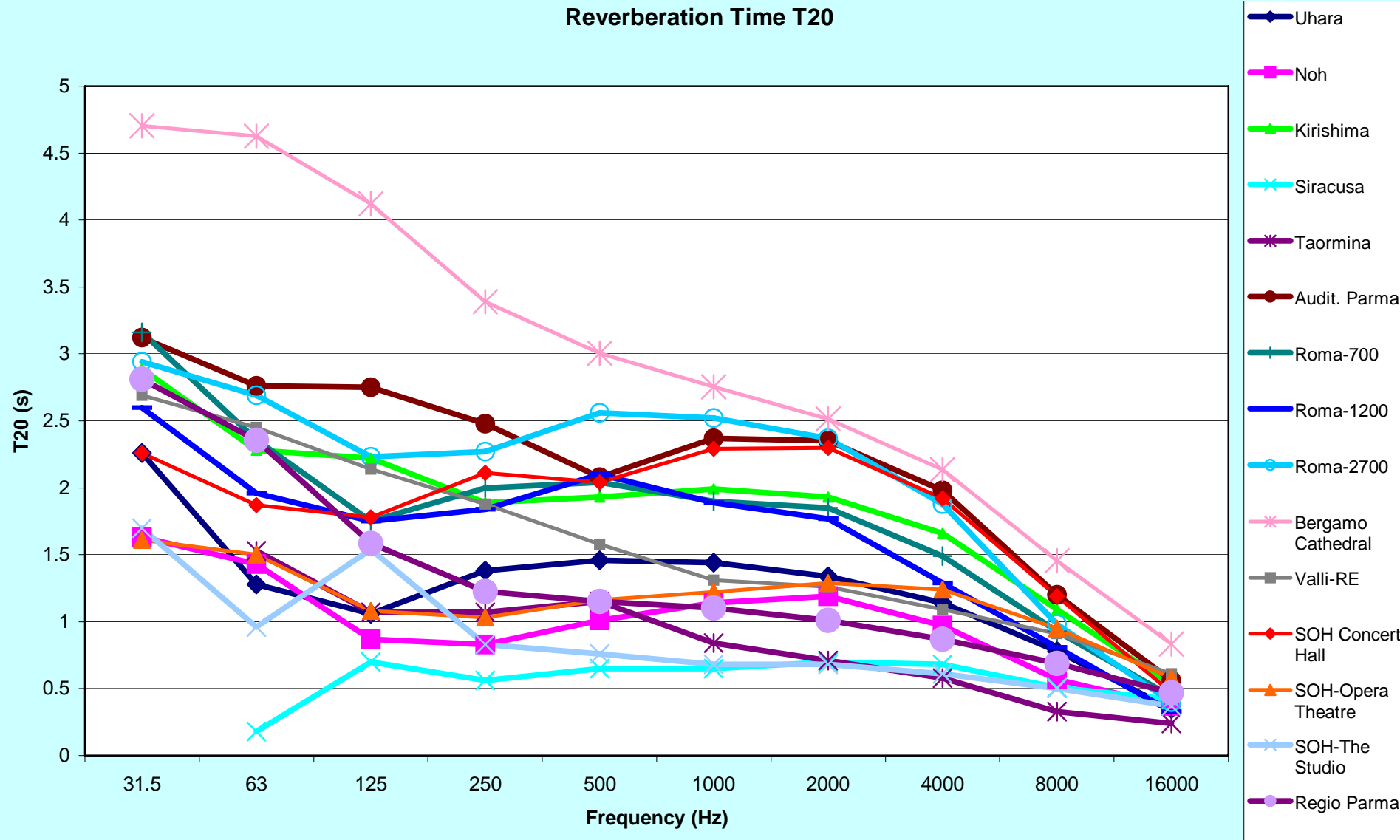


Measurement procedure

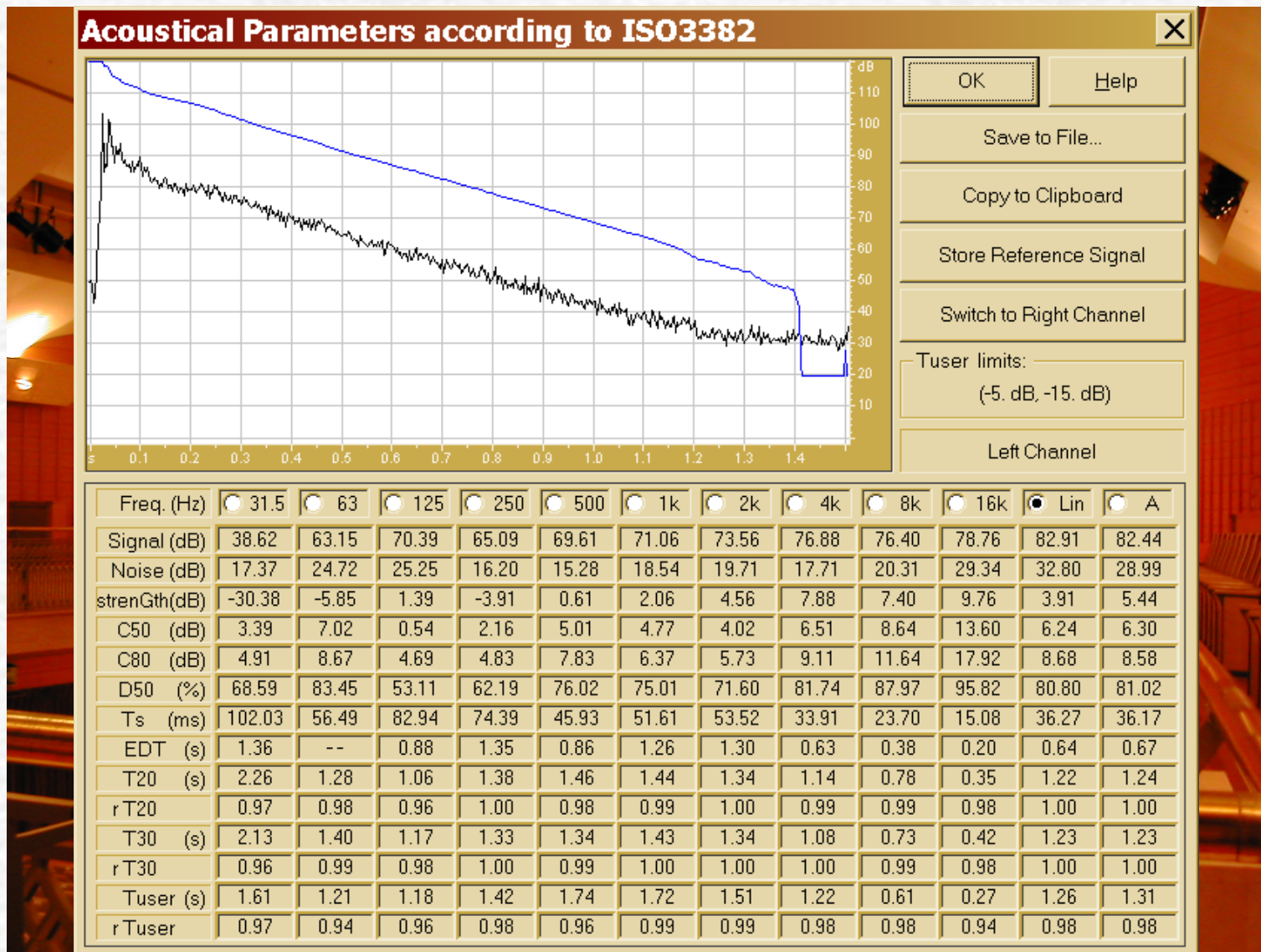
- A single measurement session play backs 36 times the test signal, and simultaneously record the 8 microphonic channels



Theatres measured



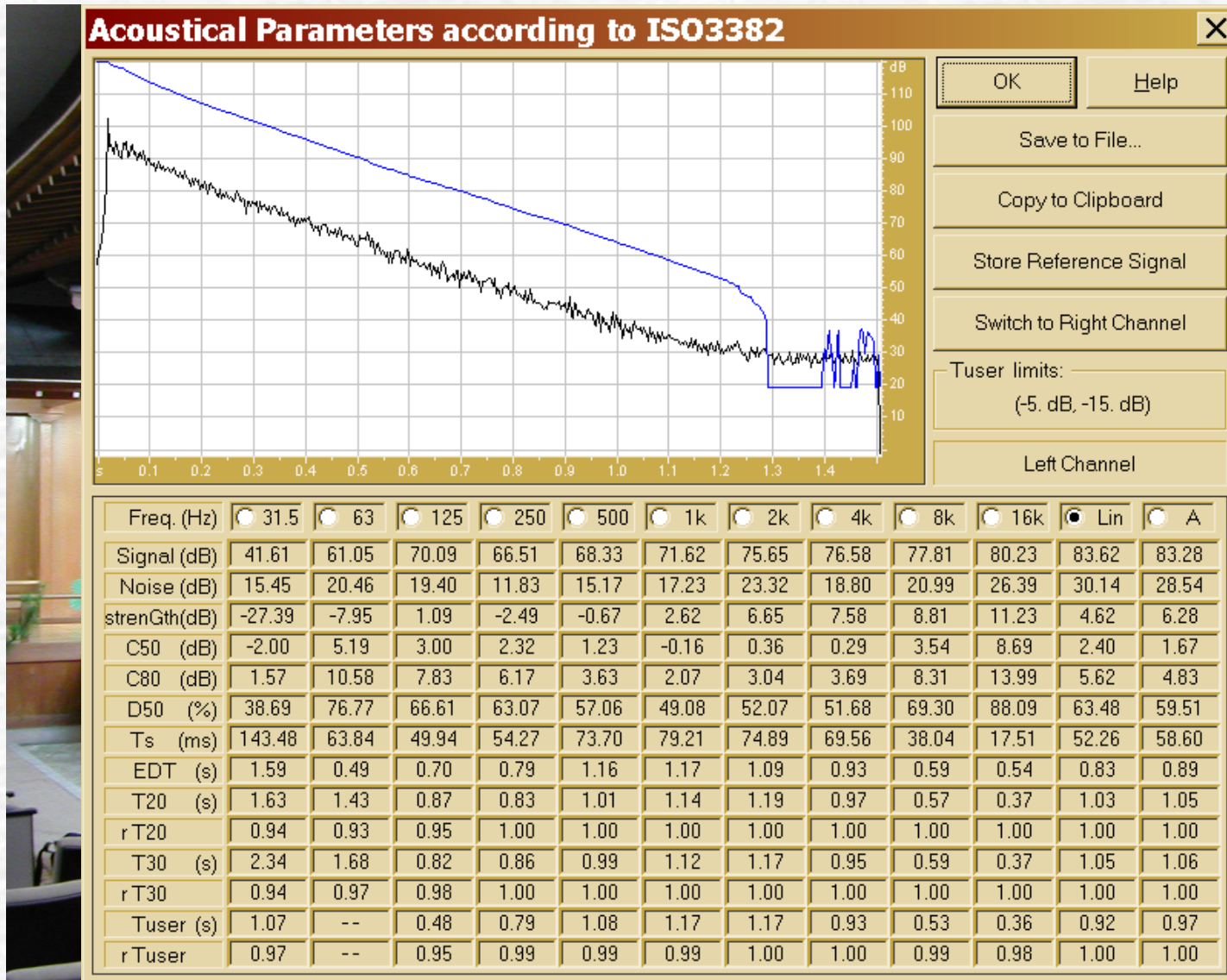
Uhara Hall, Kobe, Japan



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



Noh theater, Kobe, Japan



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



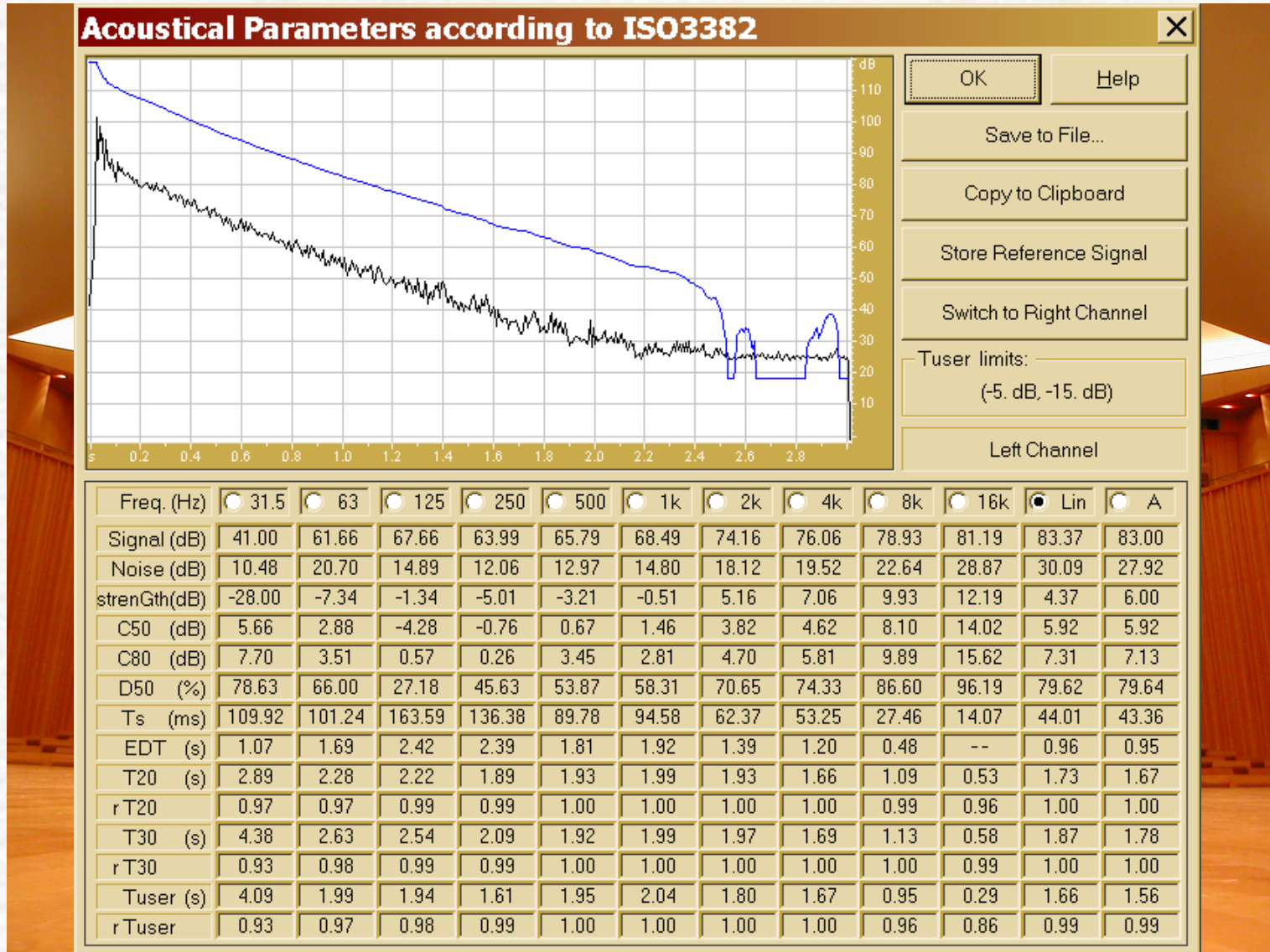
Kirishima Concert Hall, Japan



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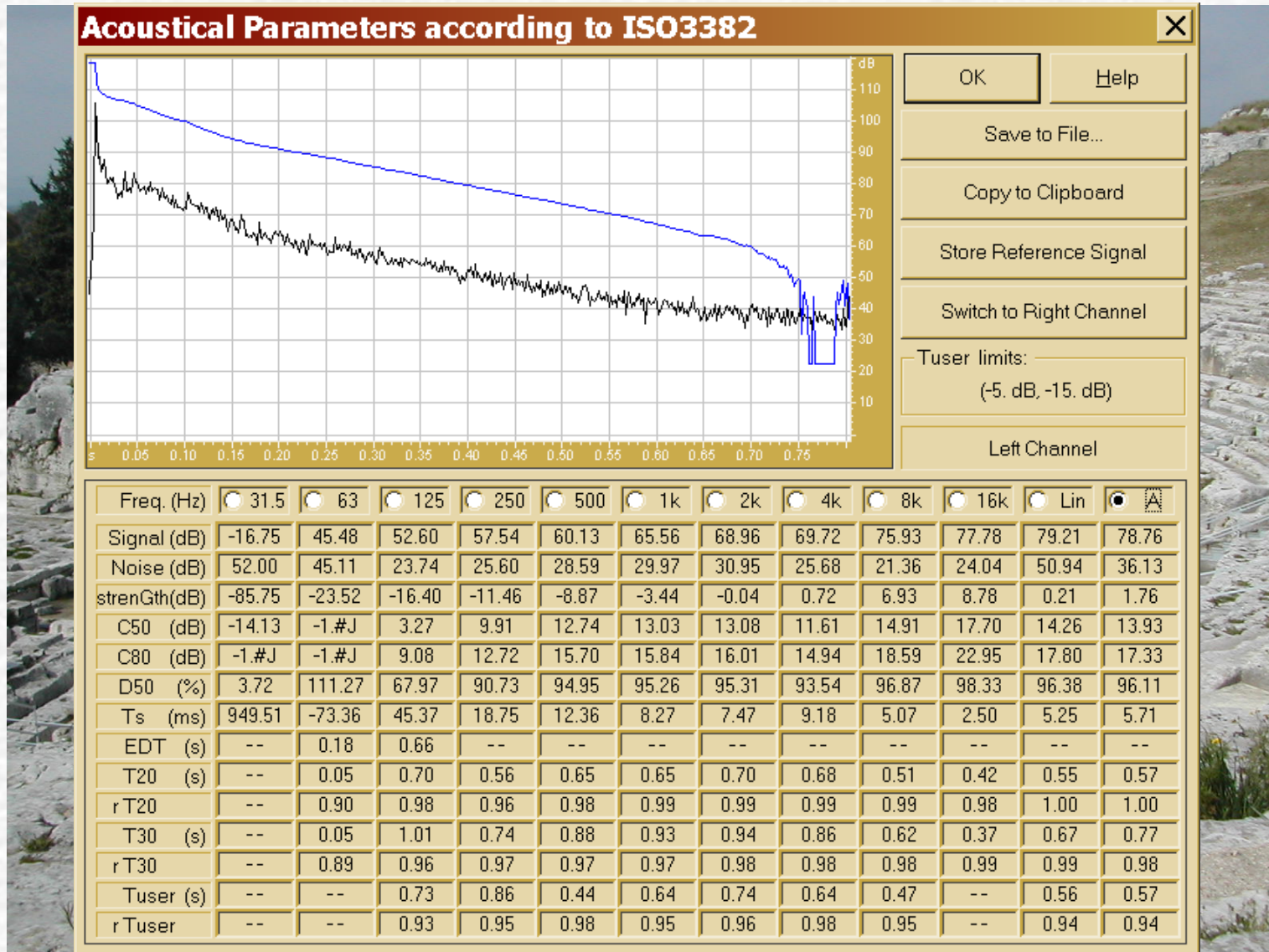
Kirishima Concert Hall, Japan



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



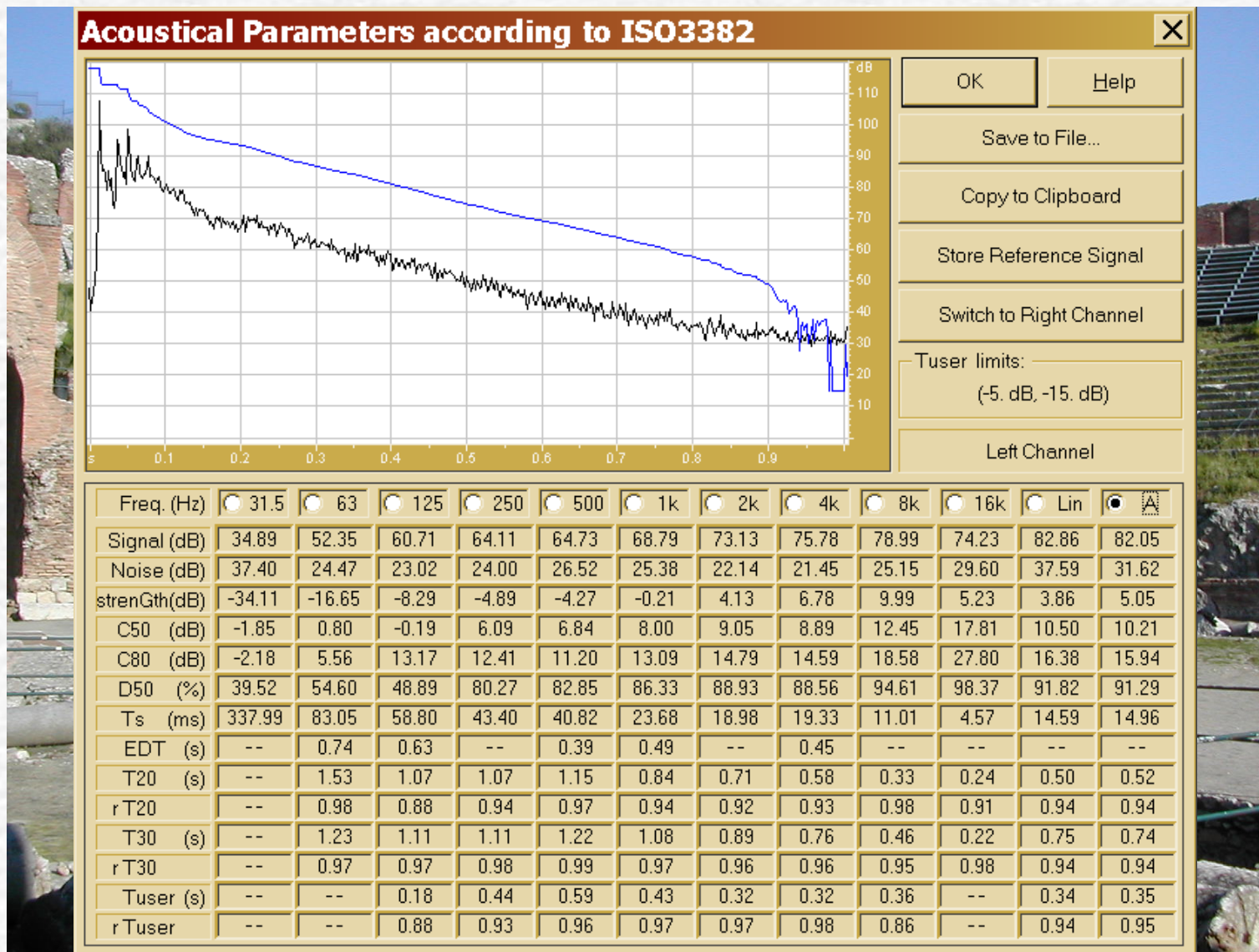
Greek Theater in Siracusa



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



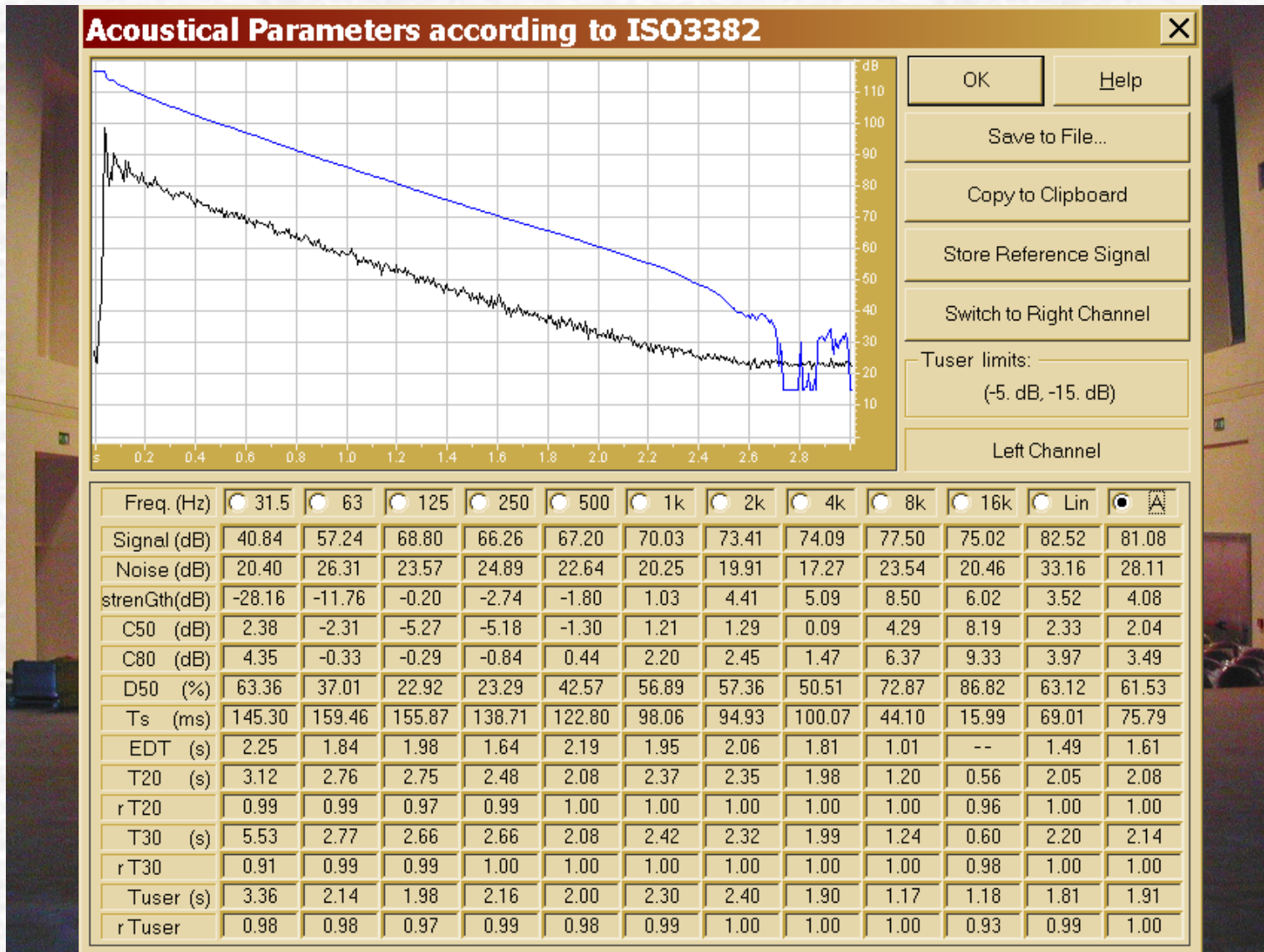
Roman Theater in Taormina



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



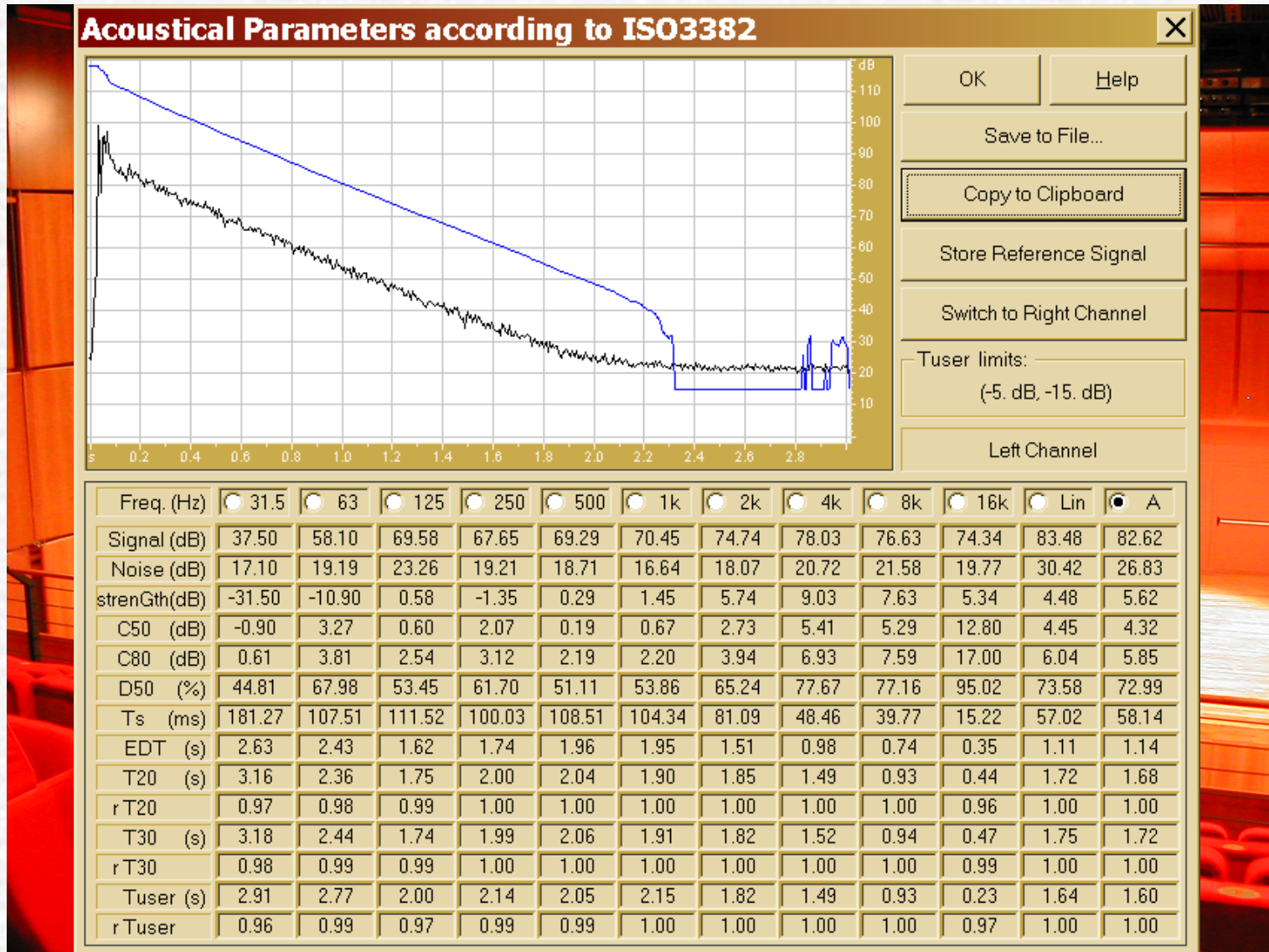
Parma Auditorium, Italy



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

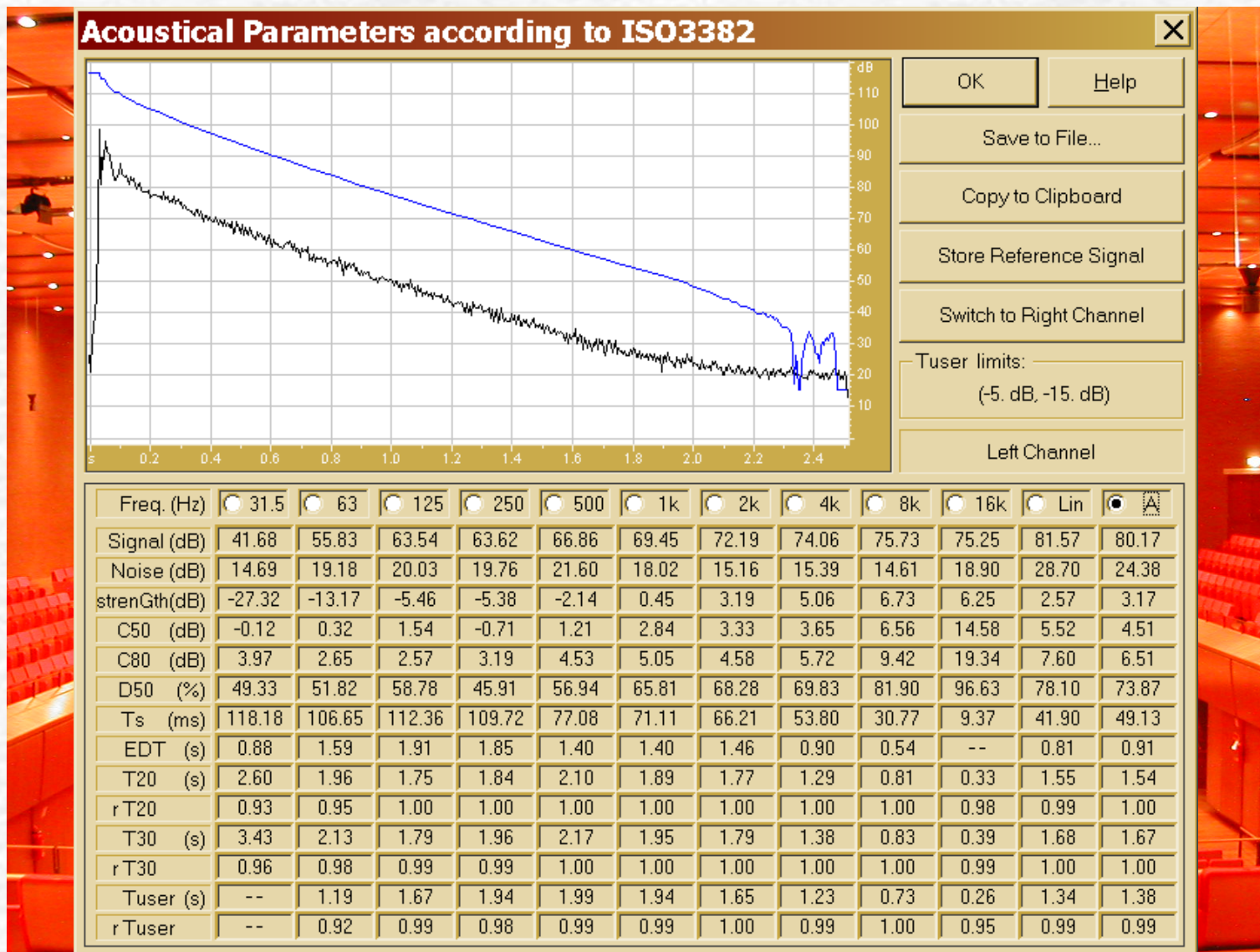


Rome Auditorium, 700 seats



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

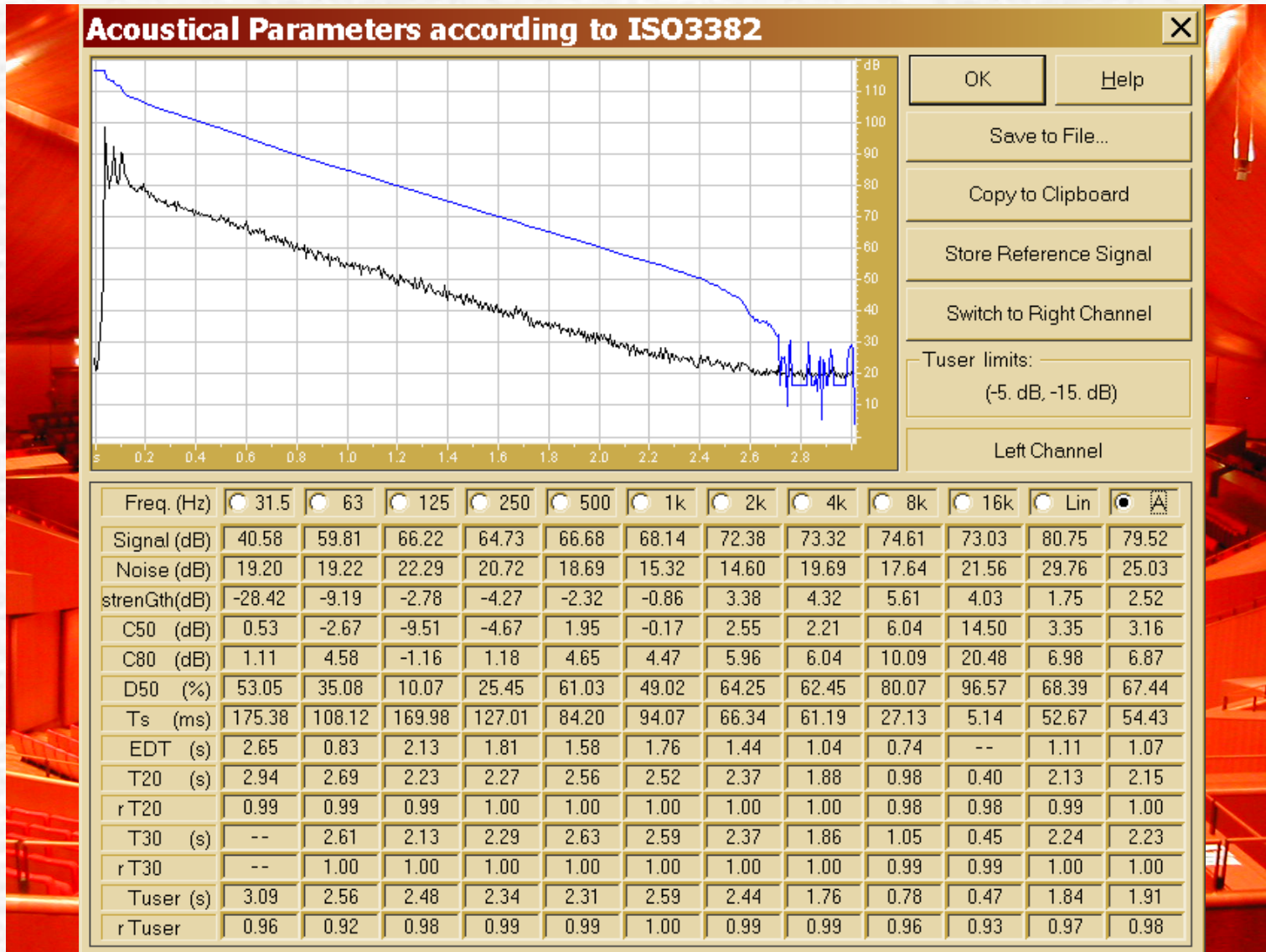
Rome Auditorium, 1200 seats



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



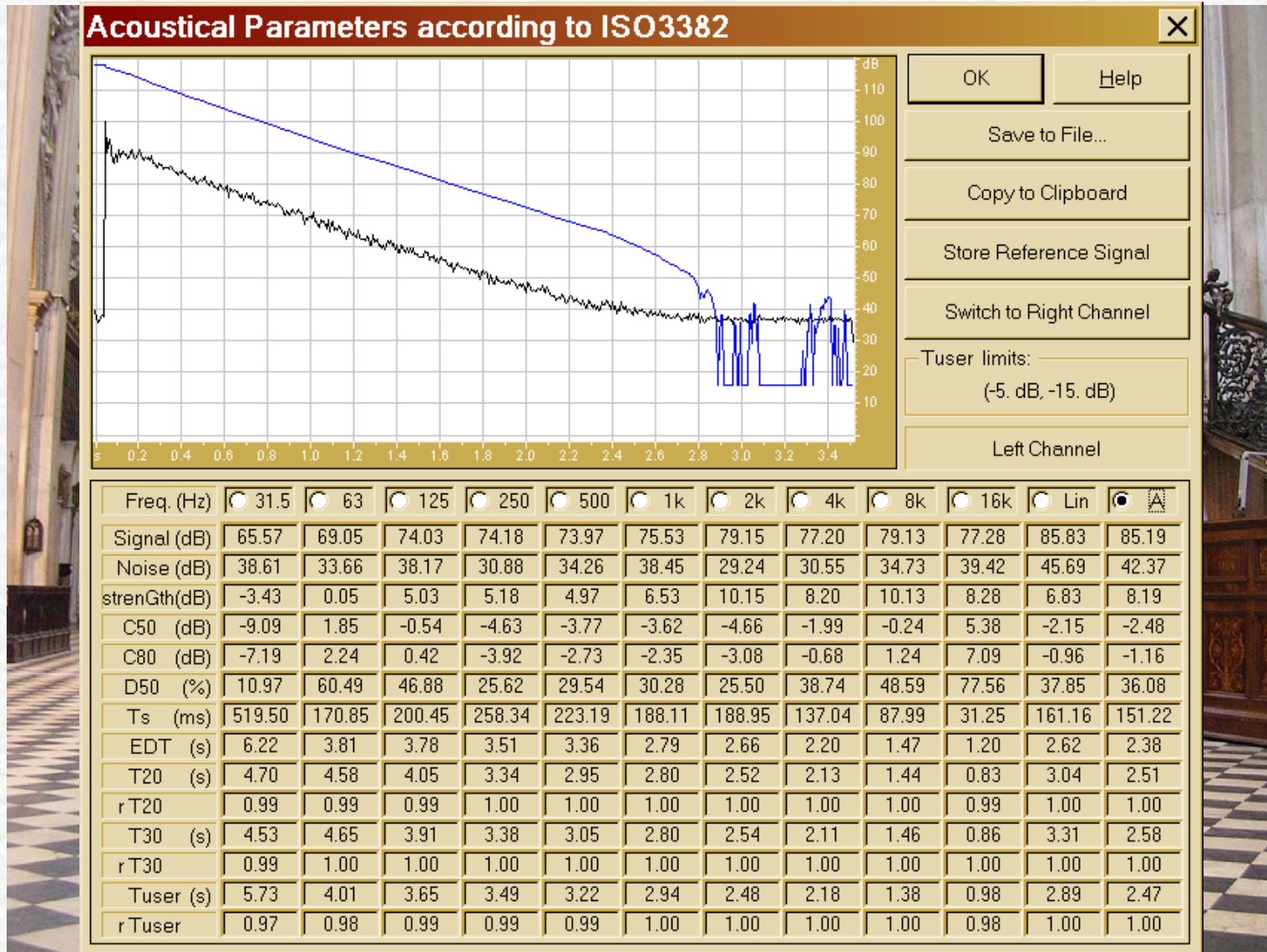
Rome Auditorium, 2700 seats



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



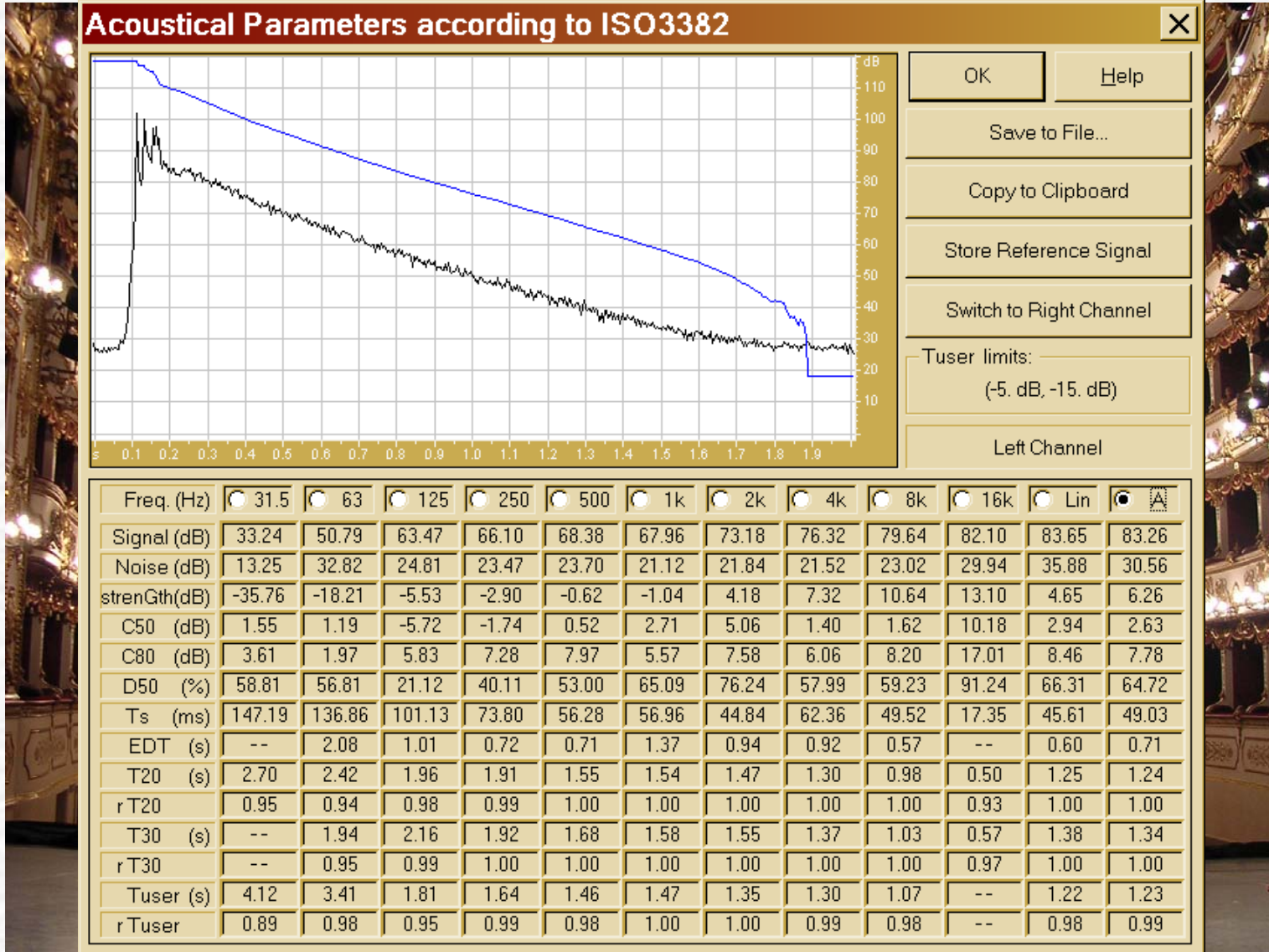
Bergamo's Cathedral, Italy



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



Teatro Valli, Reggio Emilia, Italy



T₂₀ = 1.55 s



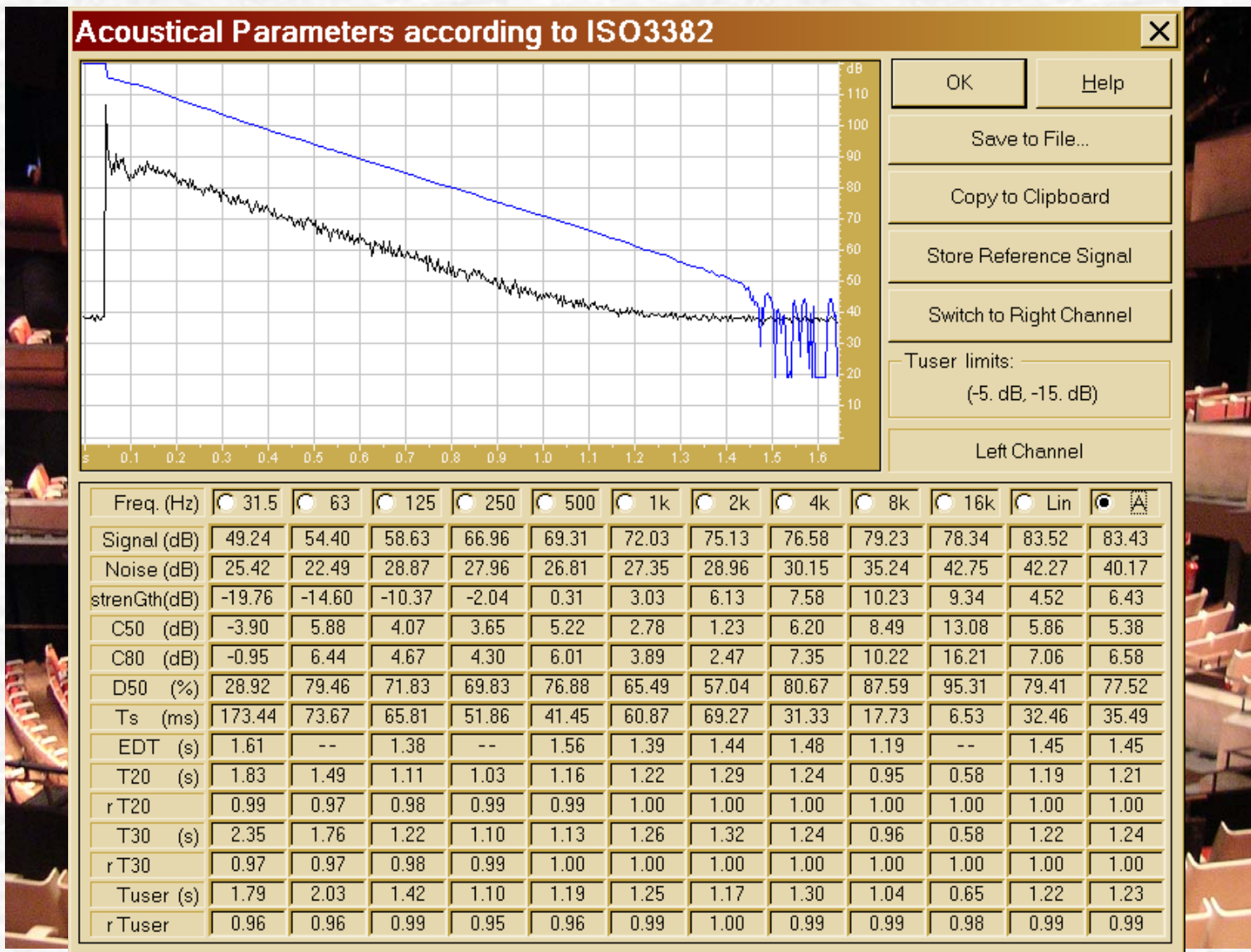
Sydney Opera House



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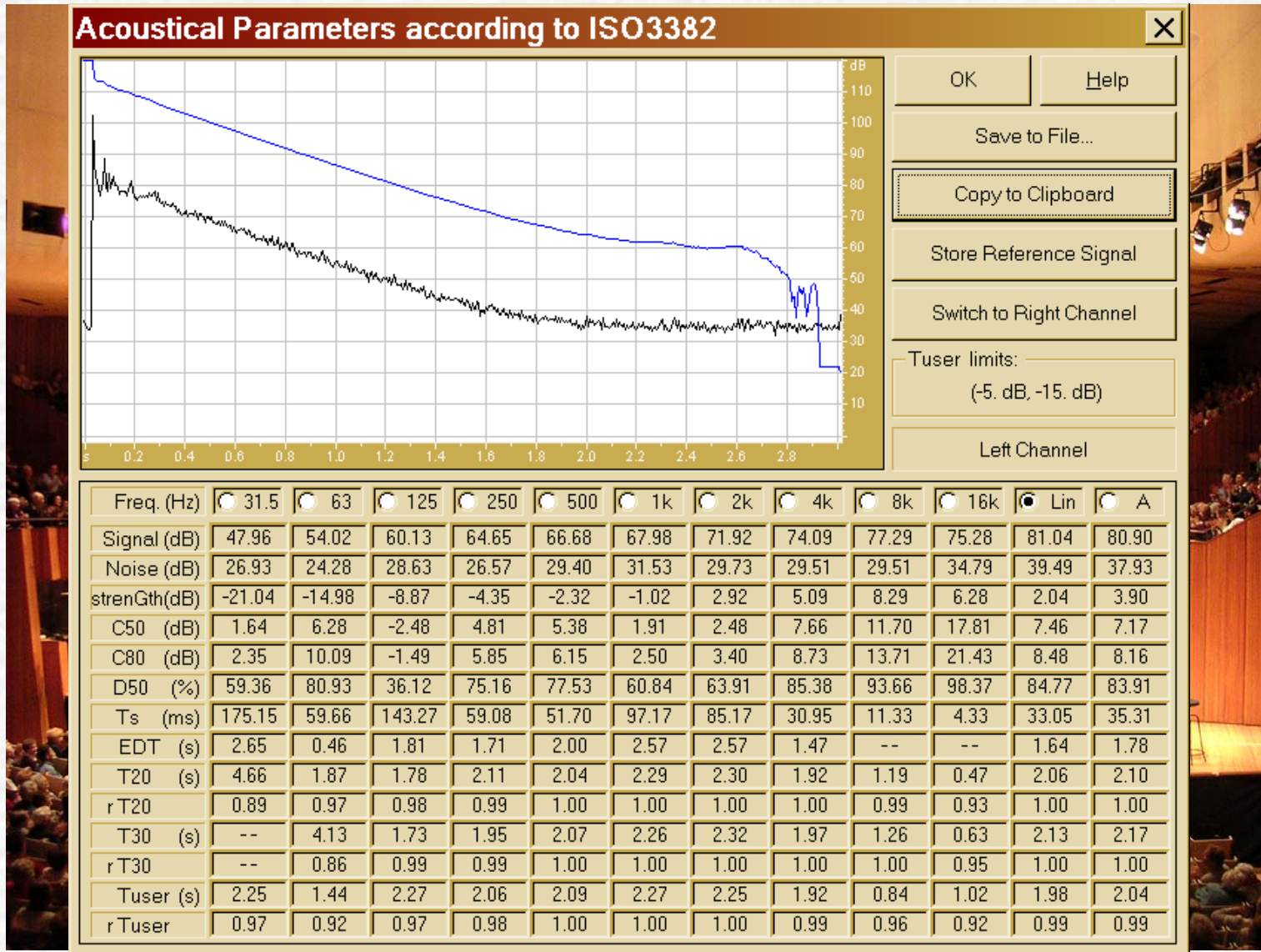
Sydney Opera House – opera theatre



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



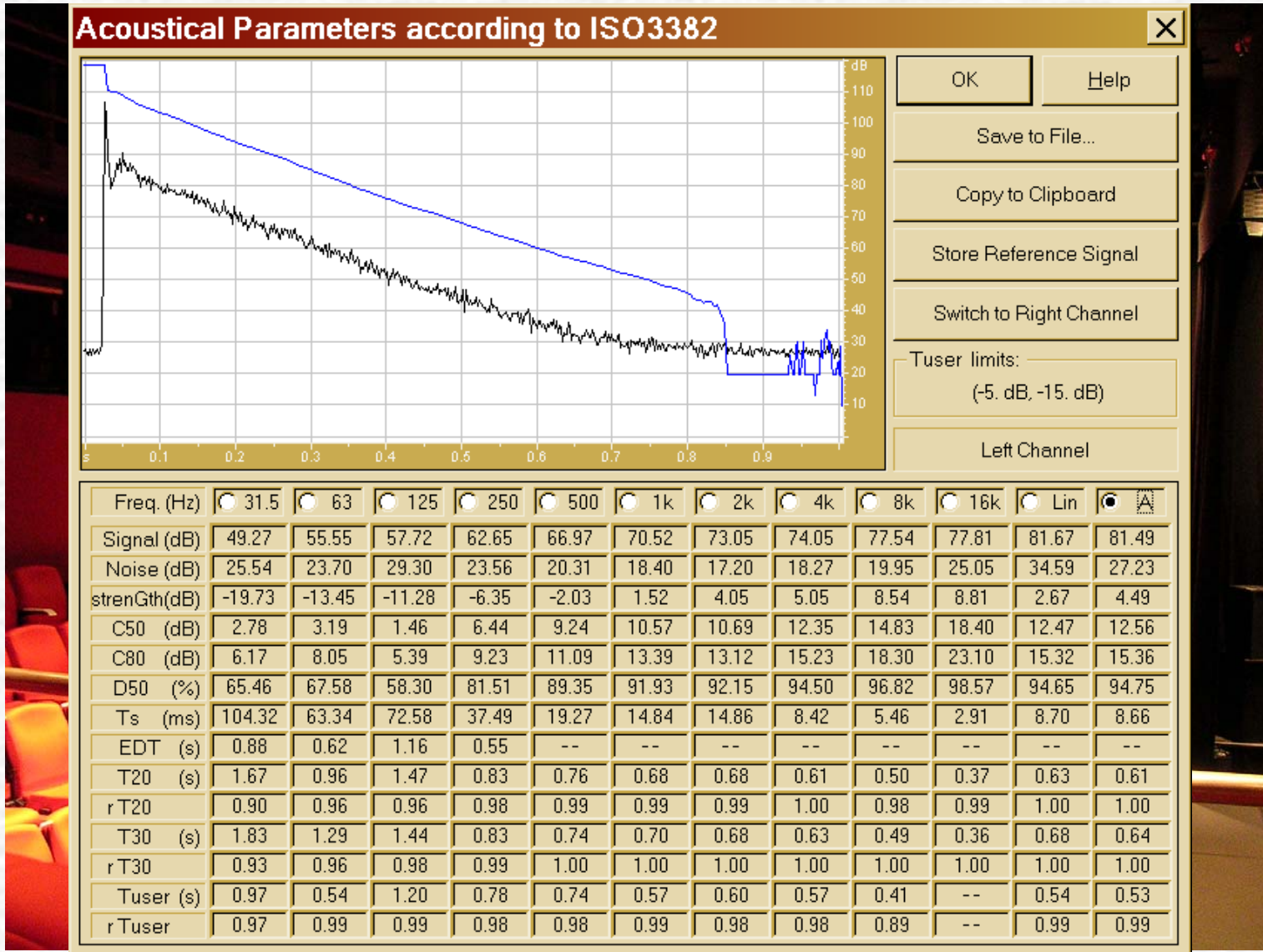
Sydney Opera House – concert hall



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



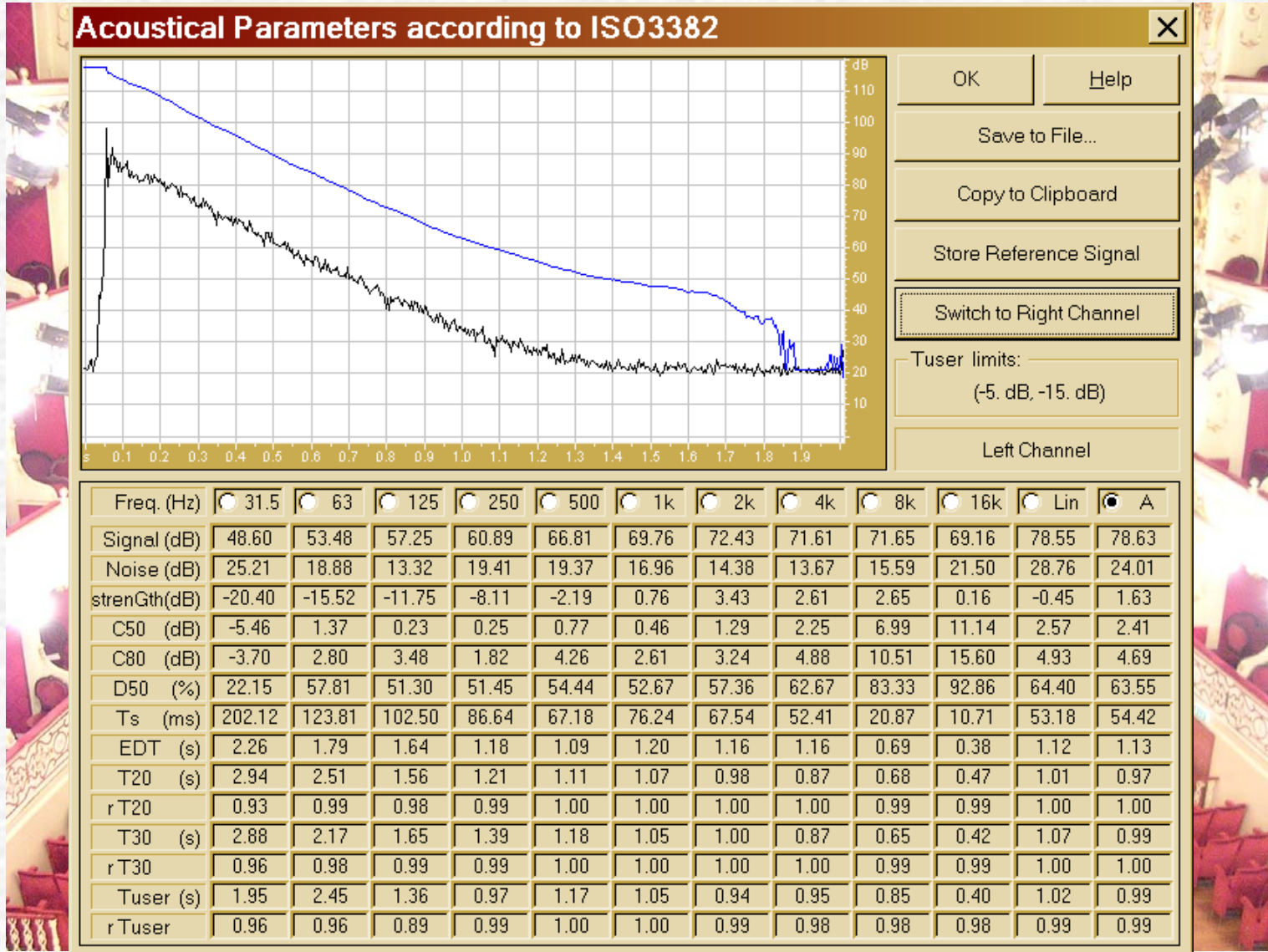
Sydney Opera House – the studio



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



Teatro Regio in Parma (Italy)



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



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).

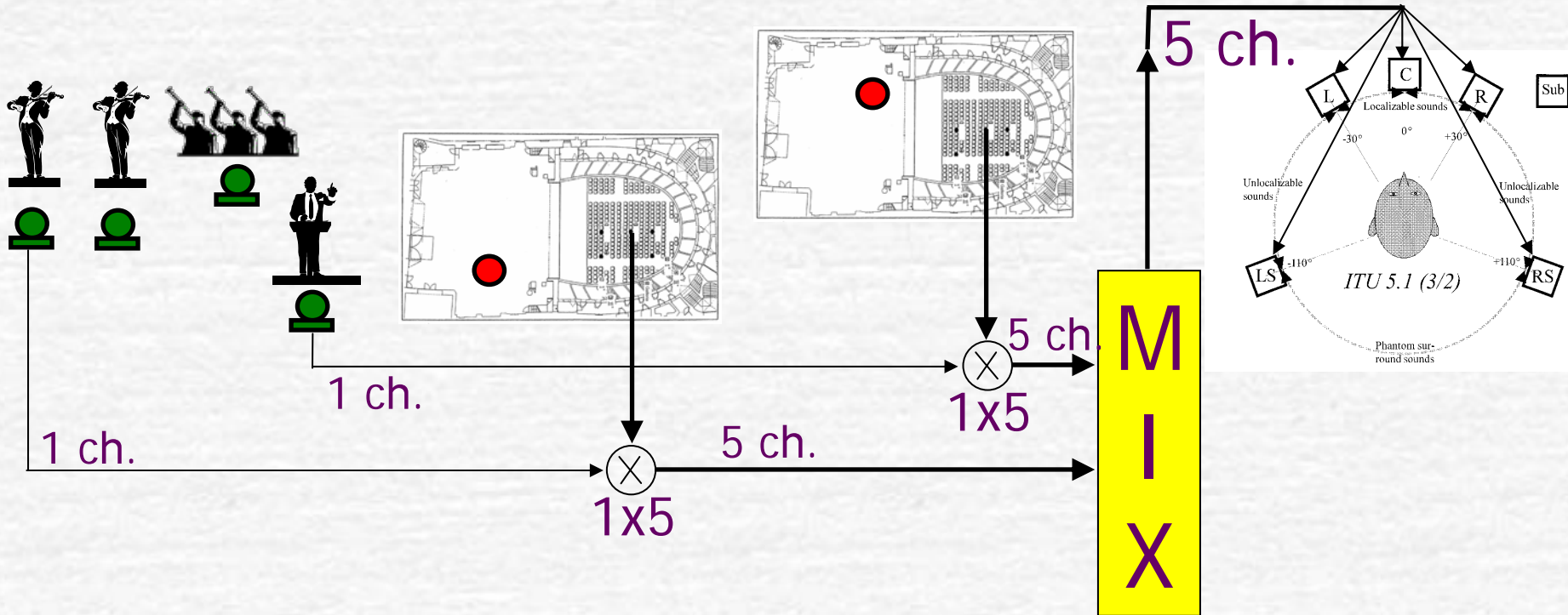


Auralization types

- Stereo (ORTF on 2 standard loudspeakers at +/- 30°)
- Rotation-tracking reproduction on headphones (Binaural Room Scanning)
- Stereo Dipole (cross-talk cancellation)
- Full 3D Ambisonics 1st order (decoding the B-format signal)
- ITU 5.1 "surround sound" systems
- 2D Ambisonics 3rd order (from Mark Poletti's circular array microphone)
- Wave Field Synthesis (from the circular array of Soundfield microphones)
- Hybrid methods (Ambiophonics)

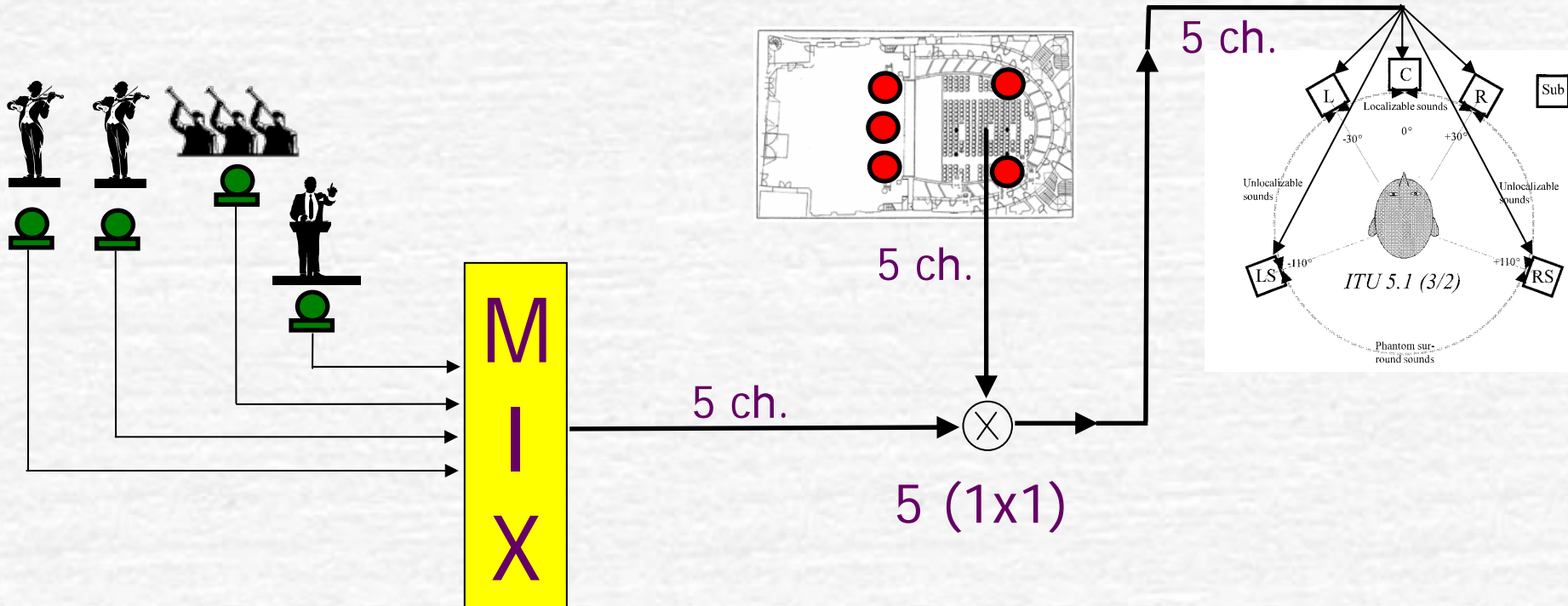


Approach # 1 – separate rendering for each recorded track (source)



- Each of the dry recordings represents a source in a different position, so it must be separately convolved with its own set of impulse responses

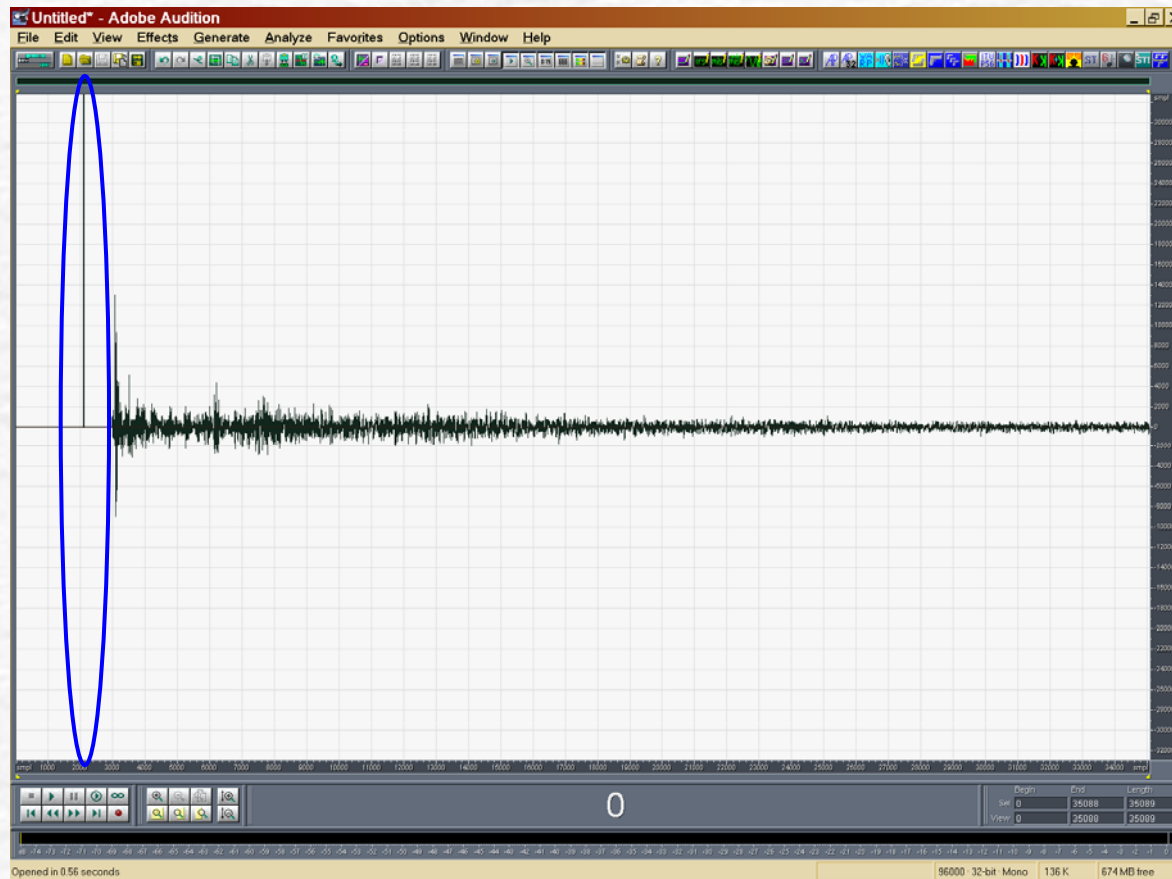
Approach # 2 – the recorded tracks are first panned to 5.0, then the “room” is added



The “room effect” is a global filtering applied to a 5.0 “dry mix” of several tracks

Full Auralization vs. Reverb

- In "full auralization" the dry signal is direct employed in place of the direct sound IRs, and the measured IRs are employed only for adding subsequent reverberant tail



Software tools

- **Dave** (VST), **Mike** (VST & Standalone), **Emily** (VST & Standalone), **Michael** (VST & Standalone), **Greg** (VST & Standalone), **Rob** (VST & Standalone), **Arif** (VST & Standalone), **Chris** (VST & Standalone), **Victor** (VST & Standalone), **John** (VST & Standalone), **David** (VST & Standalone)

DirectX PlugIn - Waves IR1 Efficient

Factory Preset:
Kirishima Concert Hall (Row 10)

IR-1 Undo A: Custom IR- IR_PR-Aud_ORTF_Pt1L A->B Load Save ? WAVES

Full CPU Reverse Bypass Gain Envelope Clear R 1.00 F 600 R 1.00 F 2500

Name: IR_PR-Aud_ORTF_Pt1L
SR: 44100Hz
Full Path: C:\Users\Farina
/Articoli/AES-24/IR-Waves/
IR_PR-Aud_ORTF_Pt1L.wav

	Original	Current
Convolution:	3.00s	3.00s
RT60:	2.2s	2.2s
Channels:	2	2
Size:		
Distance:		

0.000Sec 2.000Sec

Damping

Equalizer

	16	62	250	1k	4k	16k
G	0.0	0.0	0.0	0.0	0.0	0.0
F	100	301	1200	5005		
Q	1.00	1.28	1.64	1.00		

Reverb Time
Convolution Full
RT60 2.2s
Ratio 1.00

Size 1.00 Density 1.00 Reso 1.00 Decorr 0

Latency
Wet 11ms
Dry 0.0ms
Crosstalk 0.0
ER|TR-% 0ms

Dry/Wet 100

Direct 0.0 Predelay 0.0

ER 0.0

Tail 0.0

Output -13.4 -5.4

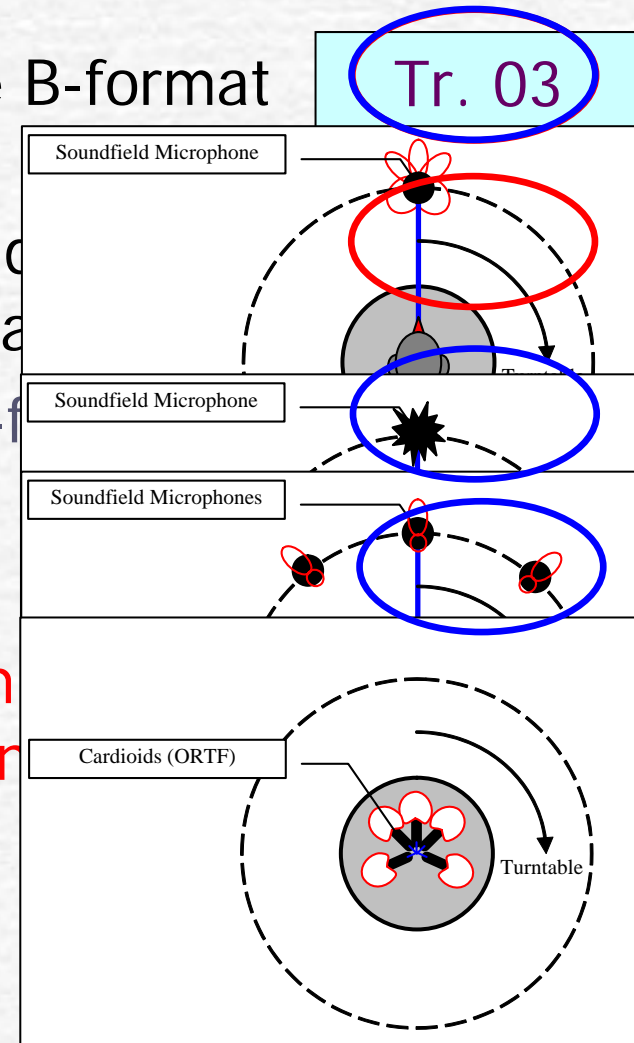
Bypass

Stop Cancel

Close OK

Approach # 1 – single source rendering – choice of 1x5 filter set

1. Ambisonics (1° order) from a single B-format impulse response
2. SIRR according to Ville Pulkki (sound intensity analysis of a single B-format)
3. 5 “virtual mikes” from 5 different B-format impulse responses
4. 5 selected Neumann cardioids
5. (future) – 5°-order Ambisonics from whole set of cardioid impulse responses



Dry s Dry music



Approach # 2 – the recorded tracks are first panned to 5.0, then the “room” is added

Multichannel Encoder

Track List

- Track 1
- Track 2

Surround Panner:

Track Options:

Panning Assignment: Surround panner, summed to mono

Sub Channel Level: 0

Center Channel Level: 100

Track Level: 100

Pan Envelopes

Splines

Clear All

1266645

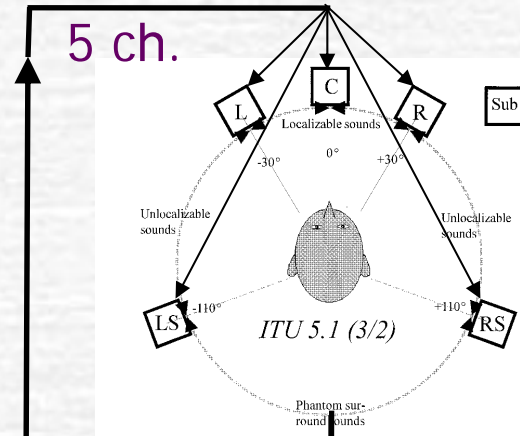
Preview Device, Format: YAMAHA AC-XG WDM Audio, 16-bit

Preview Volume: 100

FL FR C LFE Ls Rs

Master Level: 0

Export... Close Help



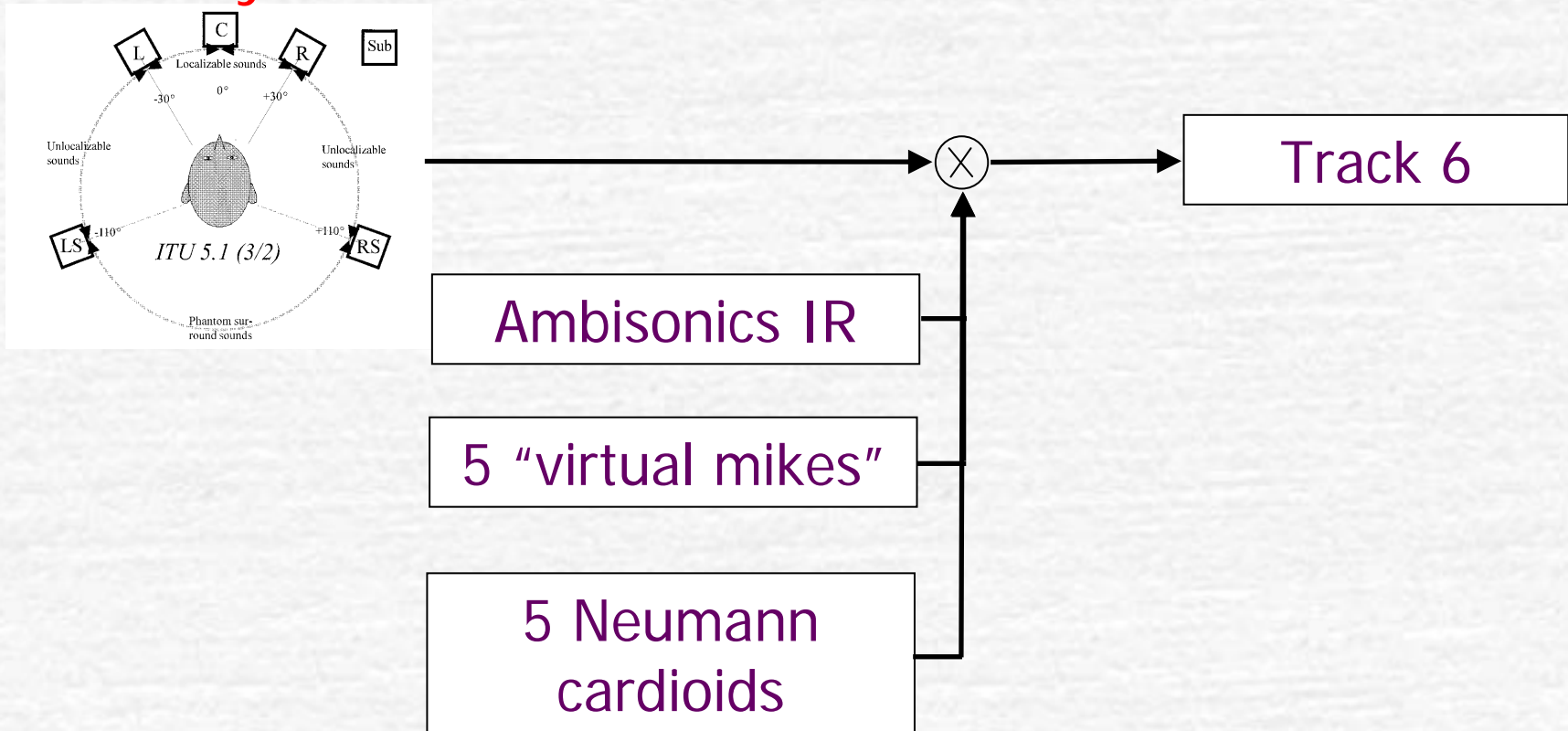
Play track 5

to convolver...



Approach # 2 – the recorded tracks are first panned to 5.0, then the “room” is added

Dry



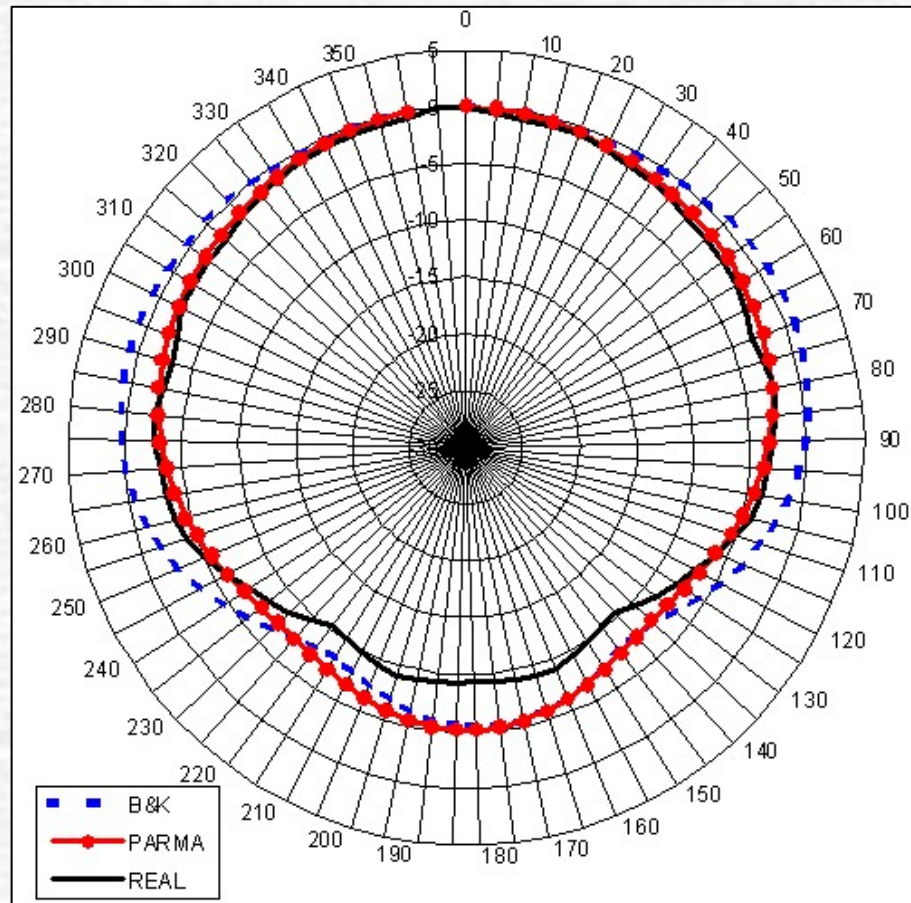
Conclusions

- It is now possible to sample accurately the spatial room impulse response, making it possible to store, analyze and preserve a “3D acoustical photography”
- We are still learning what is the best way to render these sets of impulse responses over a standard 5.0 (or 5.1) setup
- The only point which requires substantial enhancement: sound sources (loudspeakers) used for IR measurements



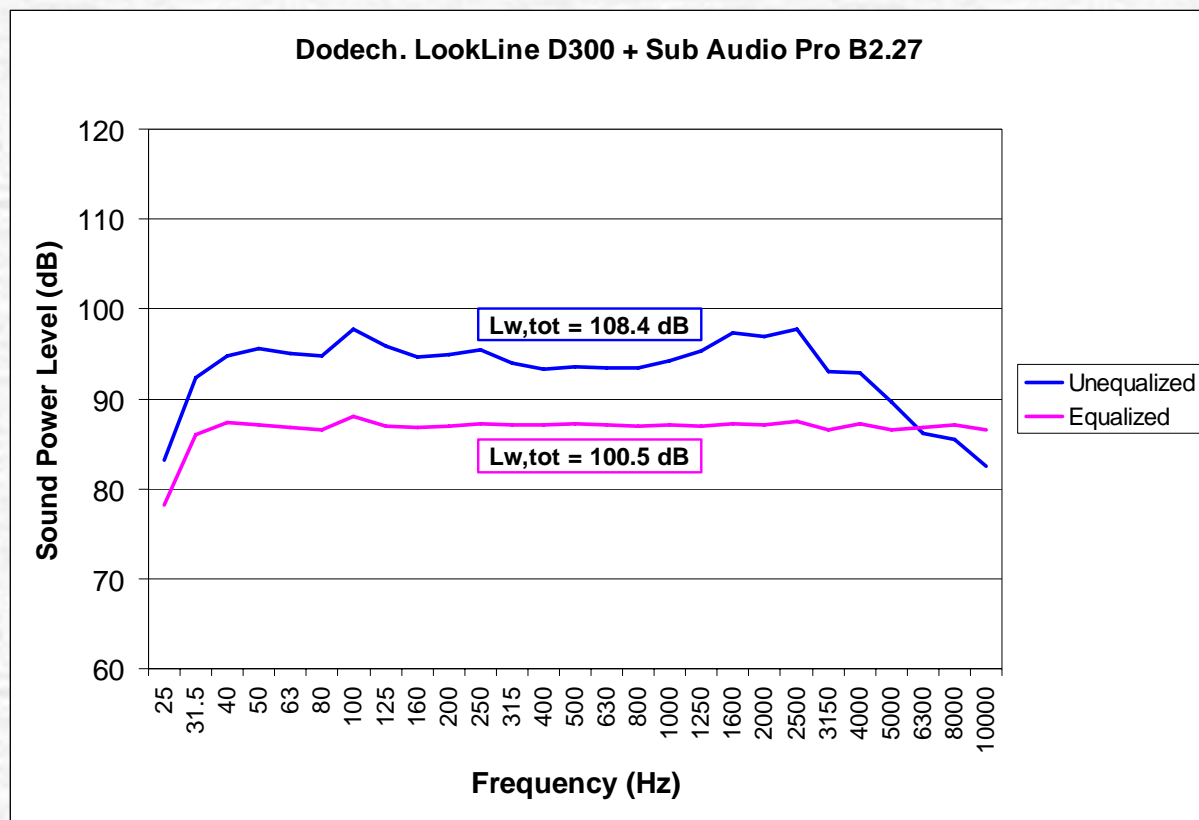
Future enhancements

- 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)
- The following software tools were provided free: Adobe Audition, Gerzonic Decopro

