IACMA

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Room Acoustics Measurements And Auralization

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Introduction

- Three main topics:

1. Introduction to sound sampling and processing
2. Room Acoustics Measurements
3. Auralization techniques: from monoaural to binaural and 3D auralization

In all of the three topics:

theoretical and experimental approach

Physical nature of sound

Origin of Sound:

Thermofluidodynamic phenomenon:

Particle velocity and variable density of medium (air)

Human body can detect sound (p and v) with:

ears, but also skin, chest, stomach

The trasducers should detect the same quantities

Transducers: loudspeakers

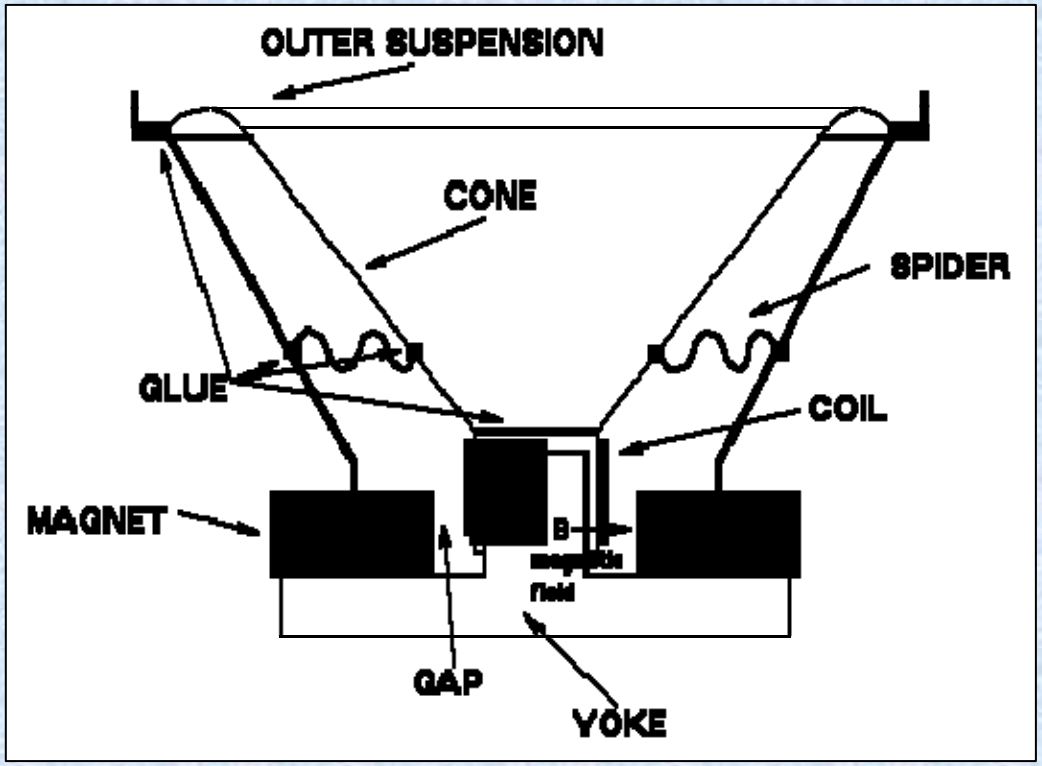
From electrical quantities (voltage, current) to acoustical radiation (volume velocity Q, sound power W)

Output signal:

- Particle velocity over a surface (volume velocity Q, m^3/s)
- The sound field reacts with an acoustical pressure p
- Depending on the phase relationship between pressure and particle velocity, a certain amount of sound power is radiated:

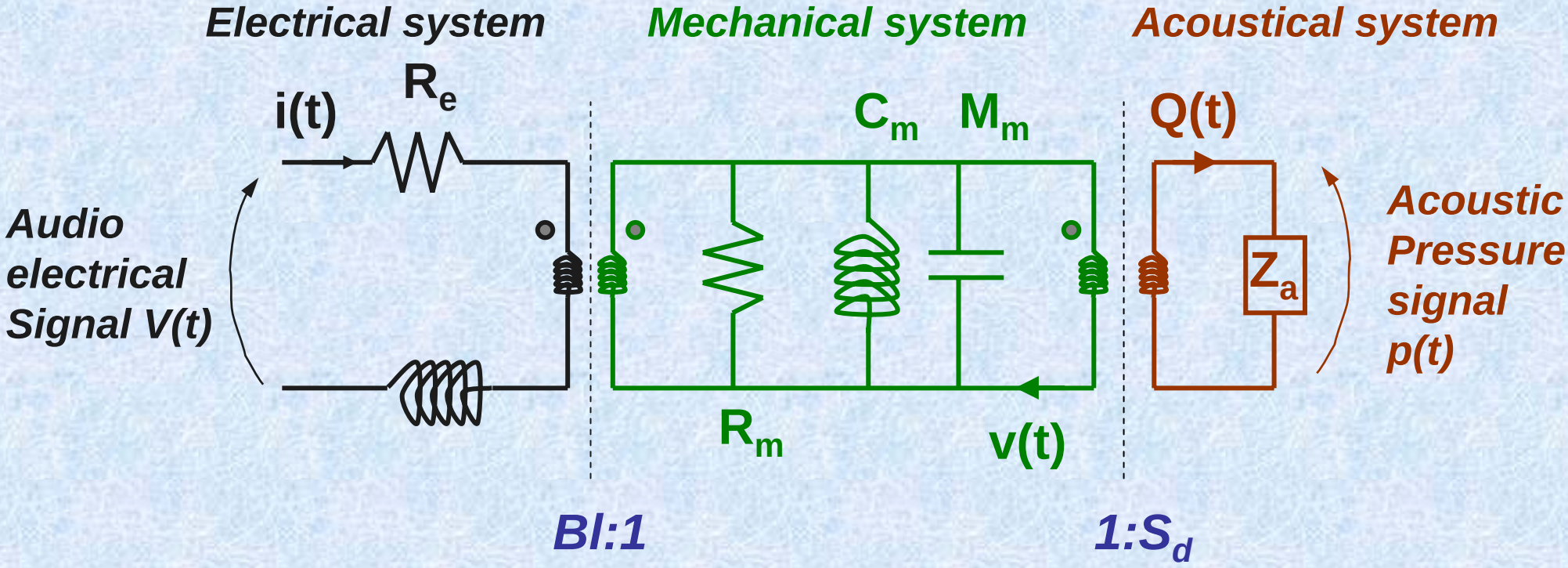
$$W = Q \cdot p \cdot \cos(\varphi)$$

Basic loudspeaker conformation



- Velocity of the moving cone is proportional to the voltage across the coil
- When connected to a voltage-controlled amplifier, the system is a voltage-to-volume velocity transducer
- However, better linearity and transient response is obtained employing current-controlled amplifiers

Loudspeaker modeling



Transducers: microphones

From acoustic pressure and particle velocity to physical (electrical) quantities

Output signal:

voltage (Volts), current (Amperes) or charge (Coulombs)

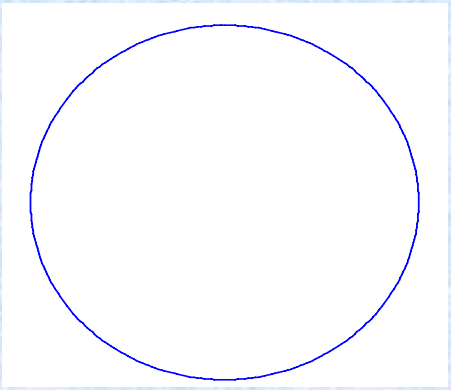
Pressure microphones (related with acoustic pressure)

Velocity microphones (related with particle velocity)

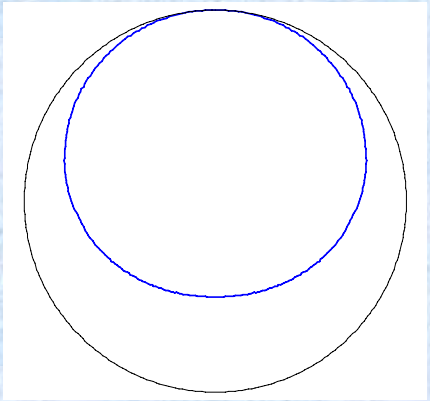
Hybrid microphones (a proper combination of both quantities)

From omnidirectional (100 p and 0 v) to figure of eight (0 p and 100 v)

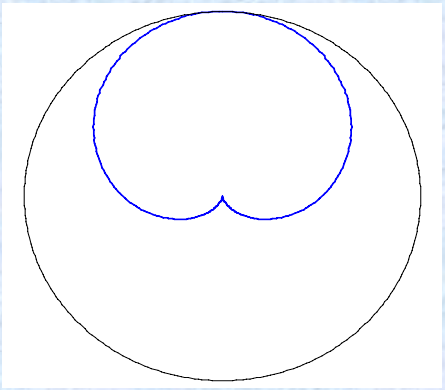
Microphone directivity patterns



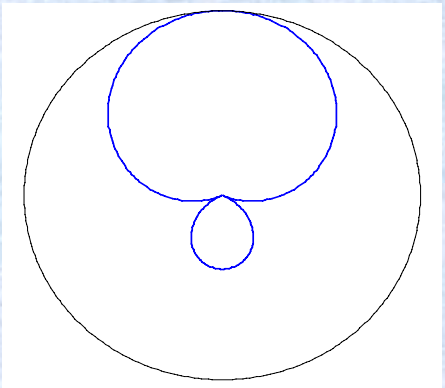
Omnidirectional (100,0)



Subcardioid (75,25)



Cardioid (50,50)



Hypercardioid (25,75)

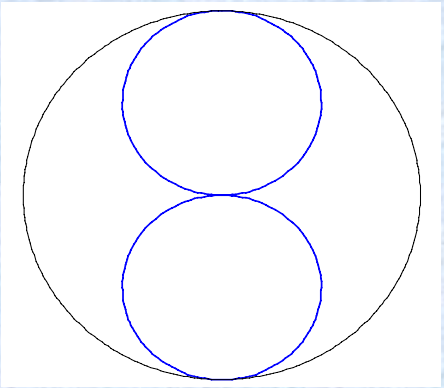


Figure-of-Eight (0, 100)

Microphones

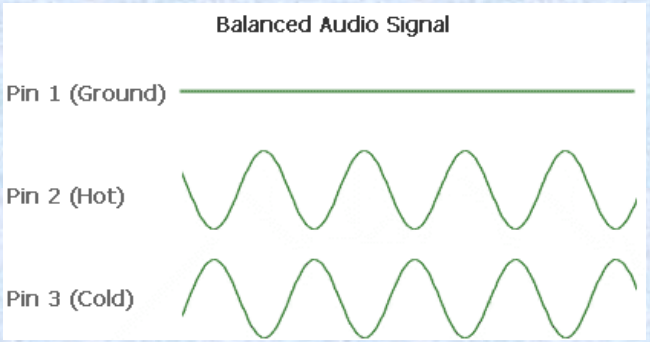
Variable pattern microphone:

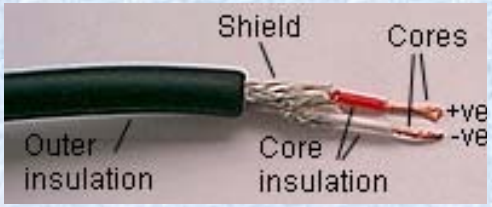
Neumann U89i variable-pattern microphone

Cables

The weak microphone signal (few mV/Pa) has to be amplified and transmitted by means of cables

Signal contamination can occur inside the cable, if not properly shielded (balanced) with two opposite-polarity signals





Balanced audio cables with XLR connectors (3 pins)

Preamplifiers

They should simply amplify the signals, but often they also process the signals:

- linearly (band pass frequencies), for phantom power supply to mics
- non-linearly (compression, harmonic distortion) it should be avoided during room acoustics measurements.

2-channels tube microphone preamplifier

ADC (Analog to Digital Converter)

Conceptually a black box connecting with two wires:

- Analog input (sound signal)
- Digital output (serial digital interface)

Two different types of ADCs:

1. PCM converters (Pulse Code Modulation – CD, DAT, DVD)
2. Bitstream converters (DSD, Direct Stream Digital, also called single-bit, employed in SACD).

ADC (Analog to Digital Converter) 2

PCM Converters

A master clock defines with high precision the instants at which the analog signal has to be “sampled” (Shannon theorem)

Applications	Resolutions
CDs	44100 Hz
DAT, DVD Video	48000 Hz
DVD audio HD rec.	96000 Hz
Special soundcards	192000 Hz

ADC (Analog to Digital Converter) 3

Low-pass filtering must be applied before entering the ADC, otherwise the signal will be aliased

Example: pure tone of 35 kHz

Sample rate 48 kHz, Nyquist freq. 24 kHz, difference = 11 kHz

After digital conversion will be 13 kHz (i.e. 24-11 kHz)

Solution: low-pass antialiasing filters, oversampling

ADC (Analog to Digital Converter) 4

Also vertical axis (amplitude) is discretized.

Typical resolutions:

16 bits; 20 bits; (24 bits)

Example: maximum voltage +5V

Discretization with 16 bit (32767 steps) means 90 dB

Discretization with 20 bit (524272 steps) means 114 dB

High-end, 2-channels ADC unit (24 bits, 192 kHz, Firewire interface)

ADC (Analog to Digital Converter) 5

Bitstream Converters

- The idea arises from oversampling: increasing sample frequency it would be possible to increase amplitude resolution (bits)
- Sample rate: 2.88 MHz and 1 bit resolution: dividing to 2 for 6 time is equivalent to CD audio sample rate, but only with 7 bits!
- In order to enhance high freq. resolution, a proper noise shaping of high order is required, suitable for static (non transient) signals.
- Below 88 Hz the Bitstream converters outperform PCM conv.

ADC (Analog to Digital Converter) 6

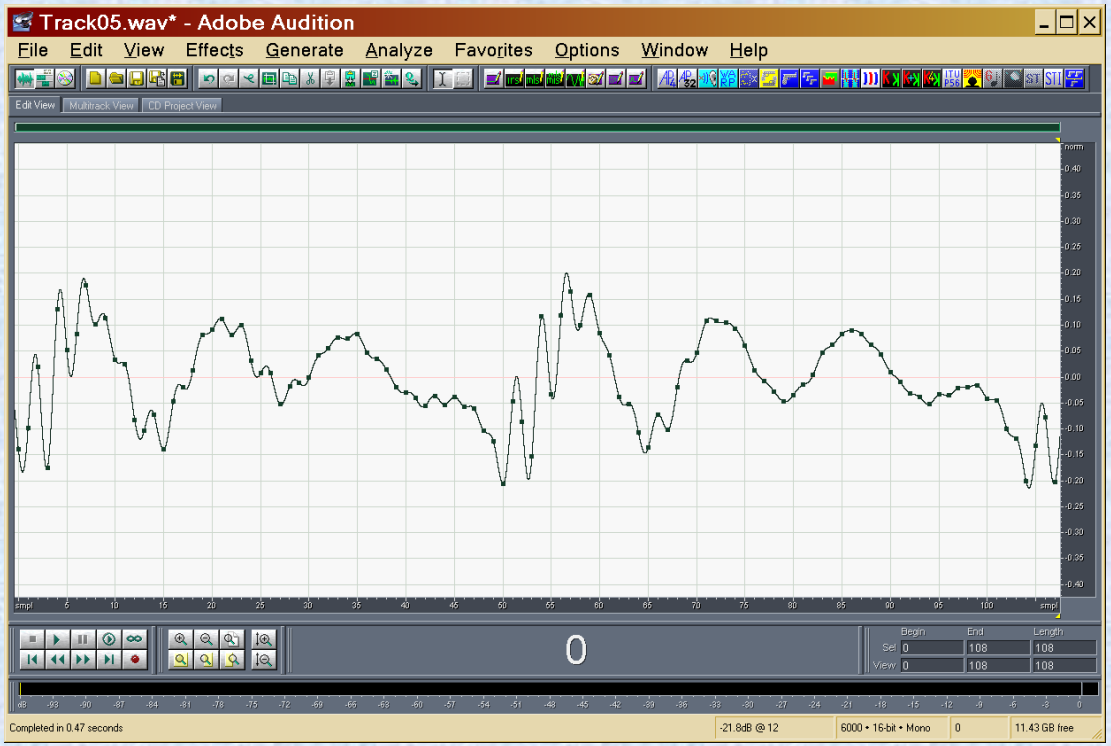
Bitstream Converters

- The Bitstream converters are widely employed with SACD system (Super Audio CD), co-developed by Sony and Philips.
- However they are much more expensive that 24 bit 96 kHz PCM

a low-cost multichannel USB-2 soundcard, equipped with 2 microphone preamps

Digital Signal Processing

Waveform editors



sampled waveform displayed as amplitude vs time (time domain)

Digital Signal Processing 2

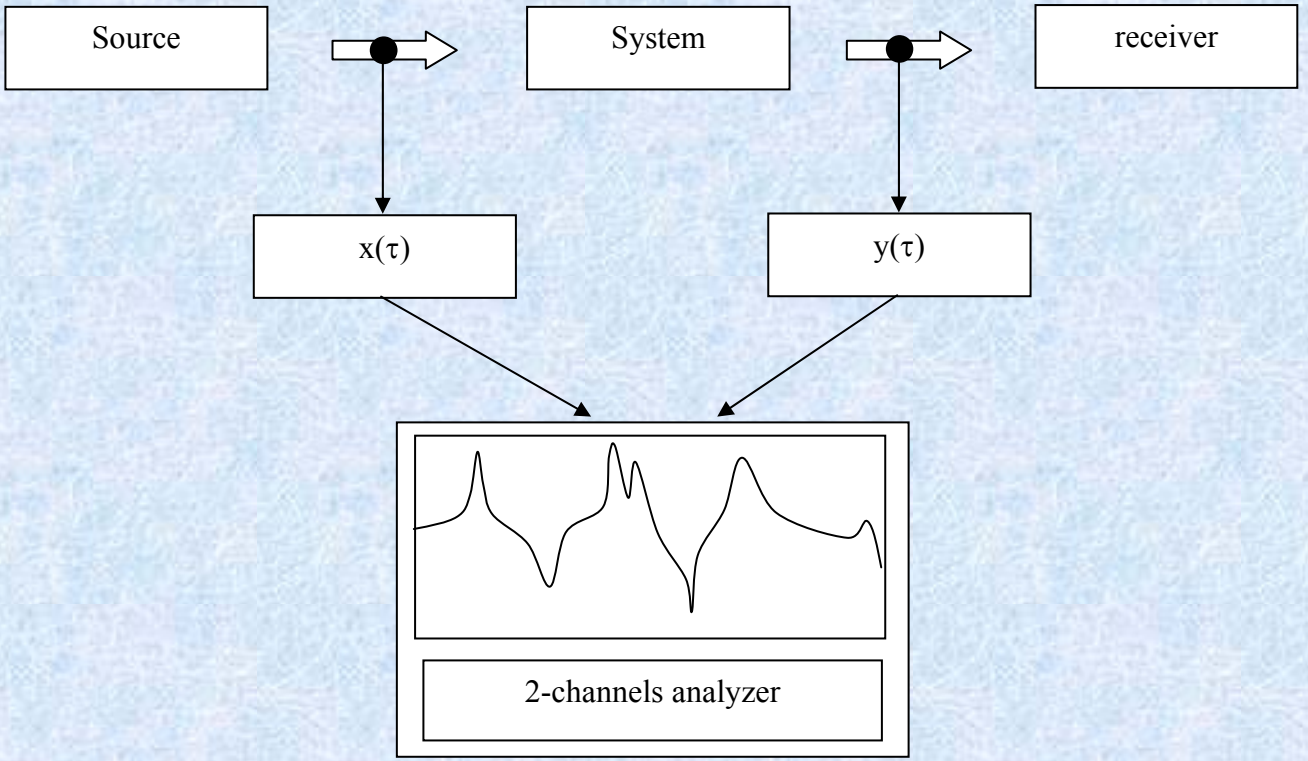
FFT: over a sample block (usually 1024, 4096 or 65565 points)

The FFT containes magnitude and phase. The phase is meaningless for single-channel, but very important during comparison between two channels (e.g.: binaural recordings)

Magnitude spectrum of the sound signal shown previously

Digital Signal Processing 3

Cross-spectrum analysis in a system



Dual-channel analysis of the input and output of a system

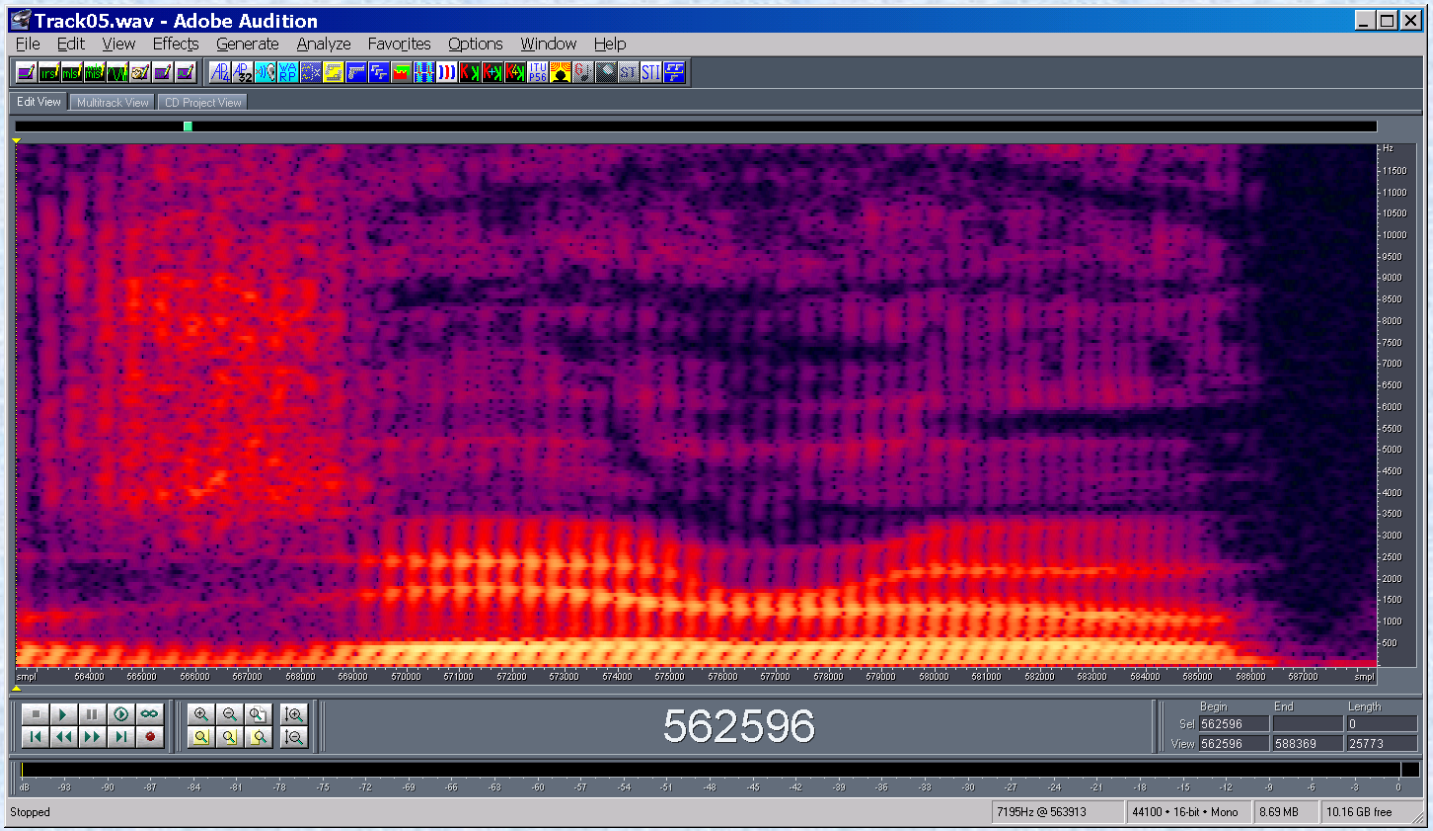
Digital Signal Processing 4

Limits in the FFT analysis

1. The frequency resolution is constant, whilst the human hearing exhibit narrow resolution at low frequency, and coarse resolution at high frequency.
2. The frequency resolution can only be improved employing longer FFT blocks, which means “averaging” over long times.
3. The signal should be perfectly steady and periodic, and having a period length exactly equal to the FFT block size (N).
4. For processing all the data with the same weight, it is necessary to employing partially overlapped FFT blocks with proper windows (Hanning, etc.).

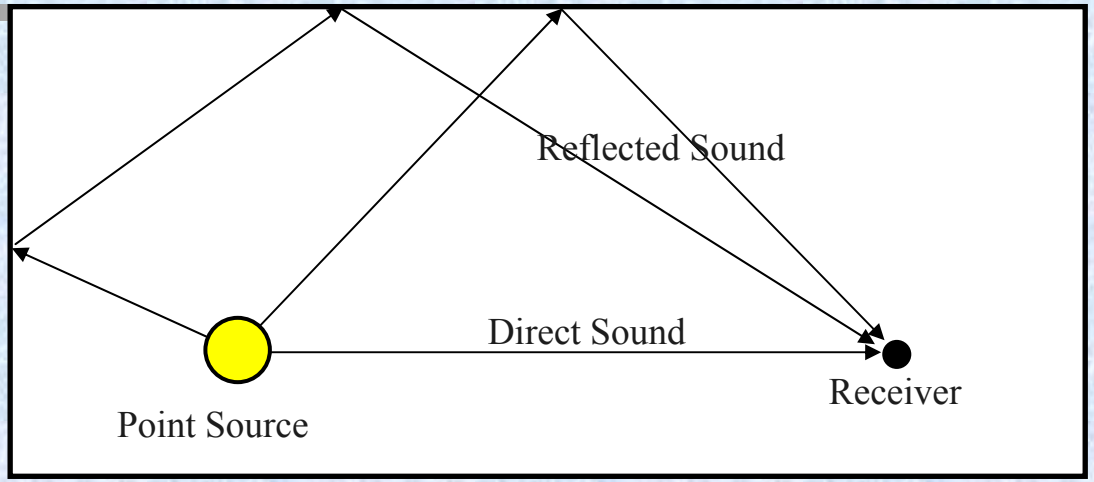
Digital Signal Processing 5

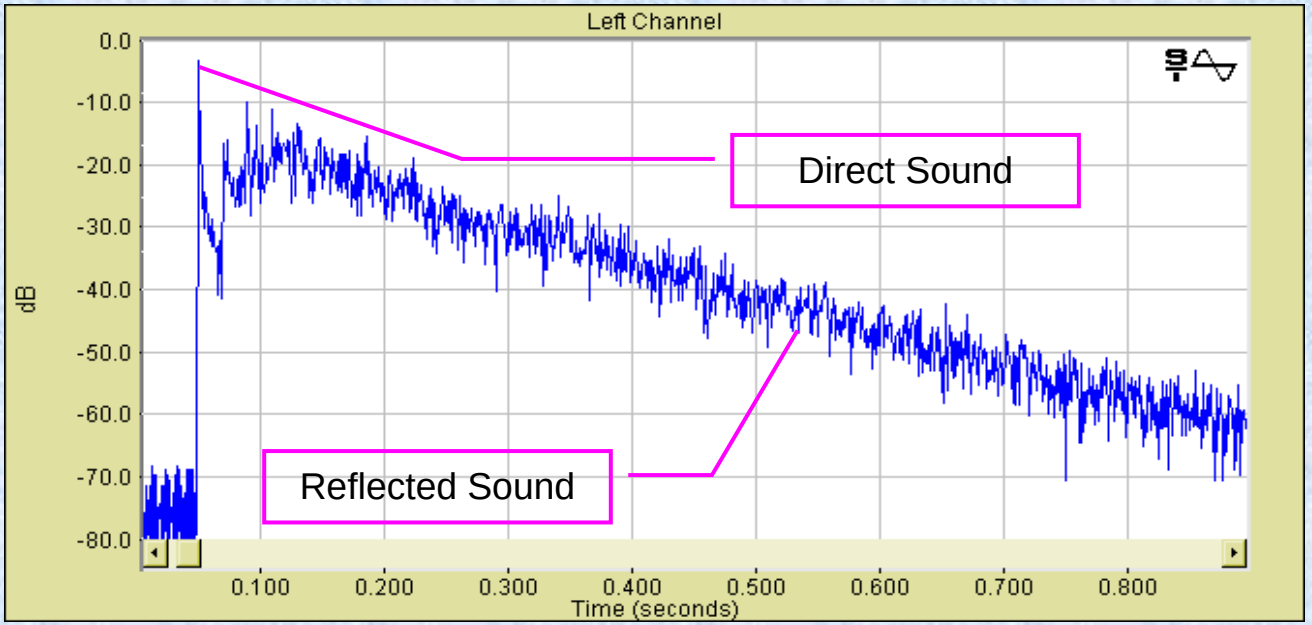
STFT (short term Fourier transform) for analyzing aperiodical signals



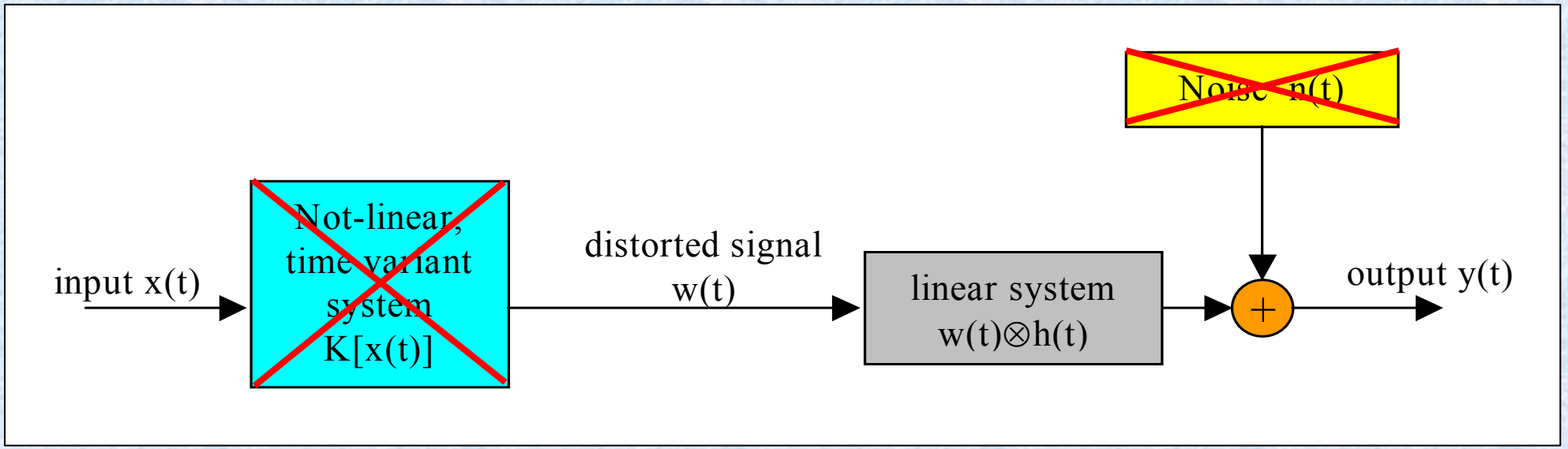
Sonogram of a piece of speech (word “zero”)

Sound propagation in rooms



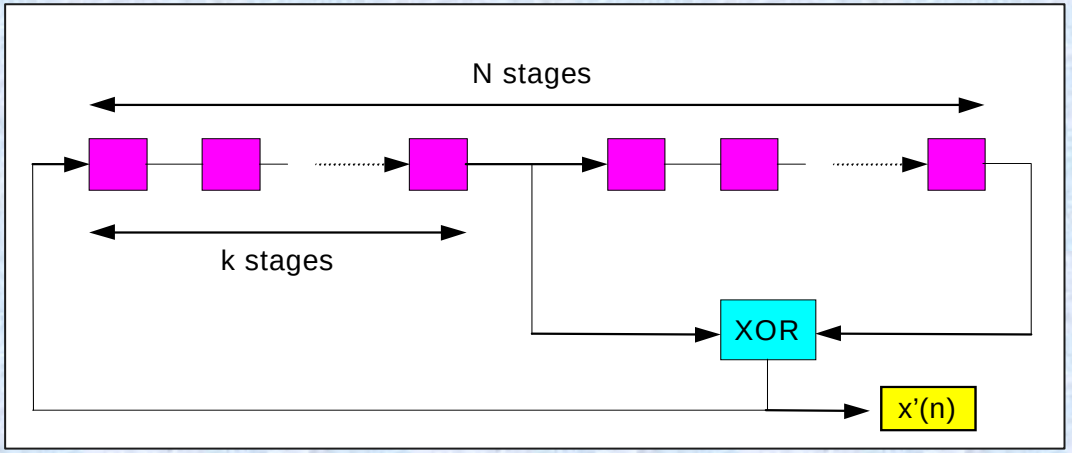


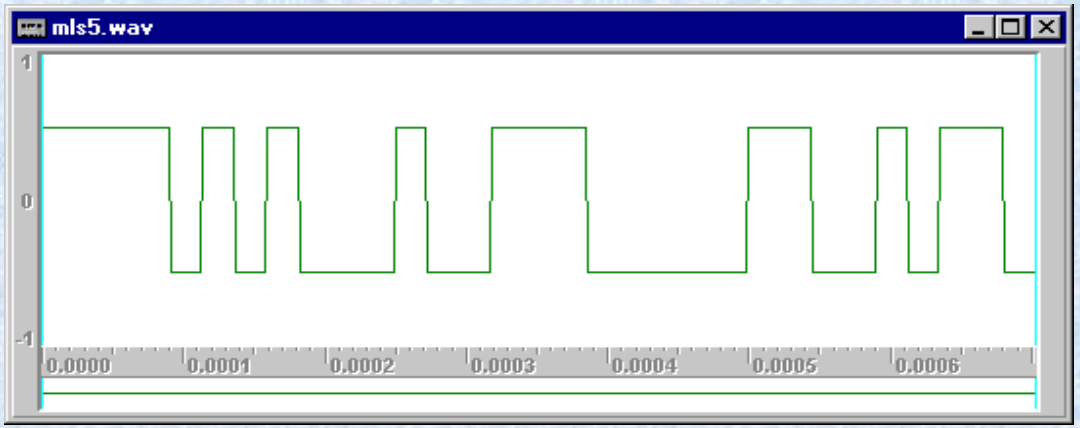
Measurement process



- We are interested in the linear impulse response h(t). This can be estimated by the knowledge of the input signal x(t) and of the output signal y(t).
- The influence of the not-linear part K and of the noise n(t) has to be minimized.

THE MLS method





- X(t) is a periodic binary signal obtained with a suitable shift-register, configured for maximum lenght of the period.

$$L = 2^{N-1}$$

MLS deconvolution

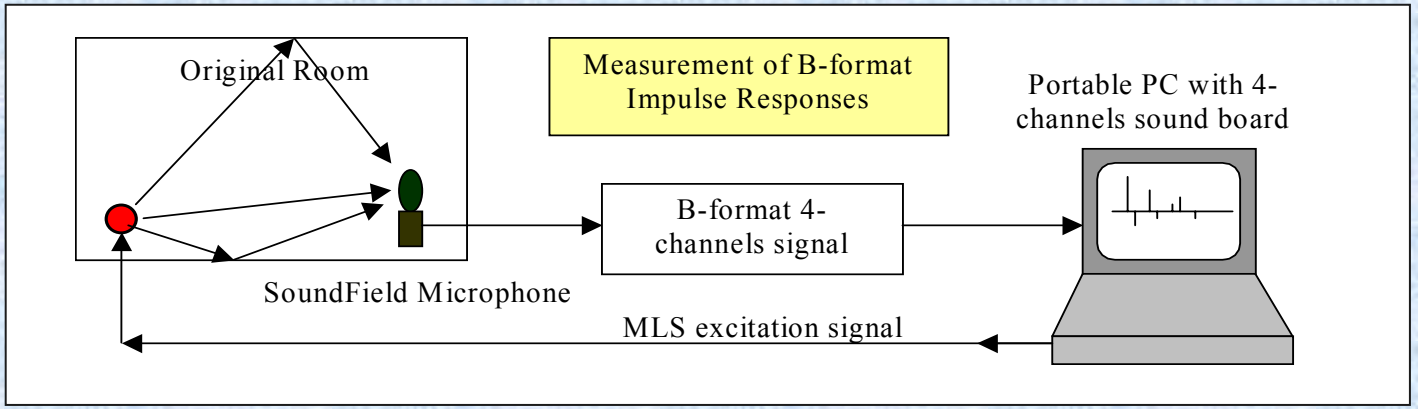
- The re-recorded signal y(i) is cross-correlated with the excitation signal thanks to a fast Hadamard transform. The result is the required impulse response h(i), if the system was linear and time-invariant

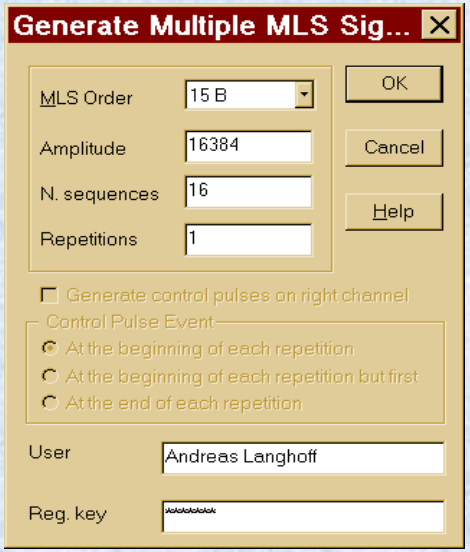
$$h = \frac{1}{L+1} \cdot \tilde{M} \cdot y$$

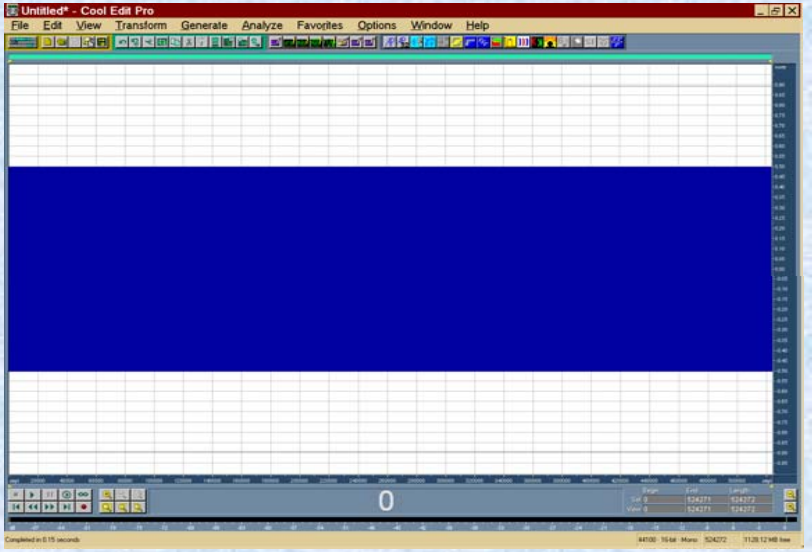
- Where M is the Hadamard matrix, obtained by permutation of the original MLS sequence m(i)

$$\tilde{M}(i, j) = m\left[(i + j - 2) \bmod L\right] - 1$$

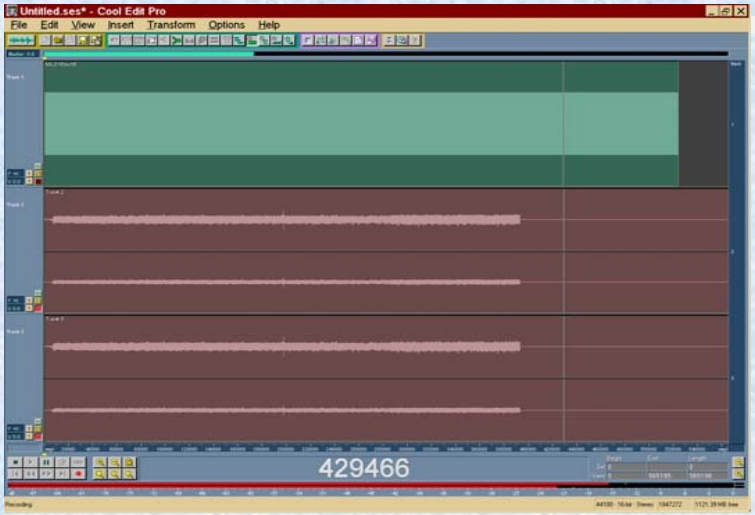
MLS example

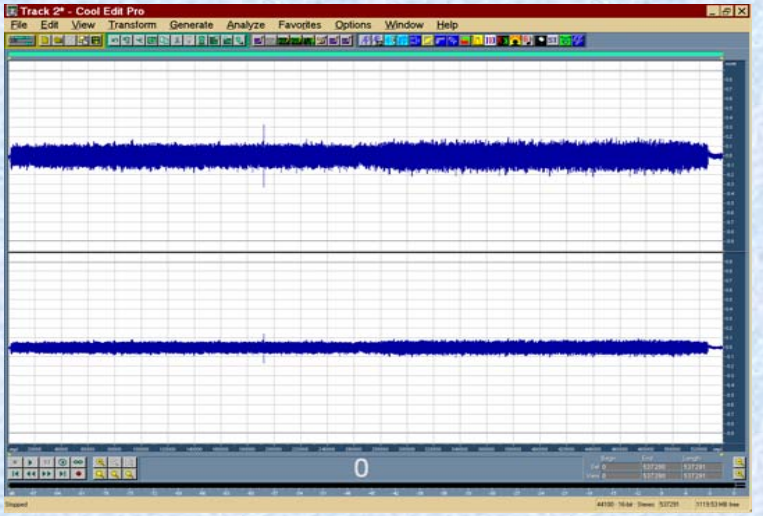

Original Room
Measurement of B-format Impulse Responses
Portable PC with 4-channels sound board
B-format 4-channels signal
SoundField Microphone
MLS excitation signal

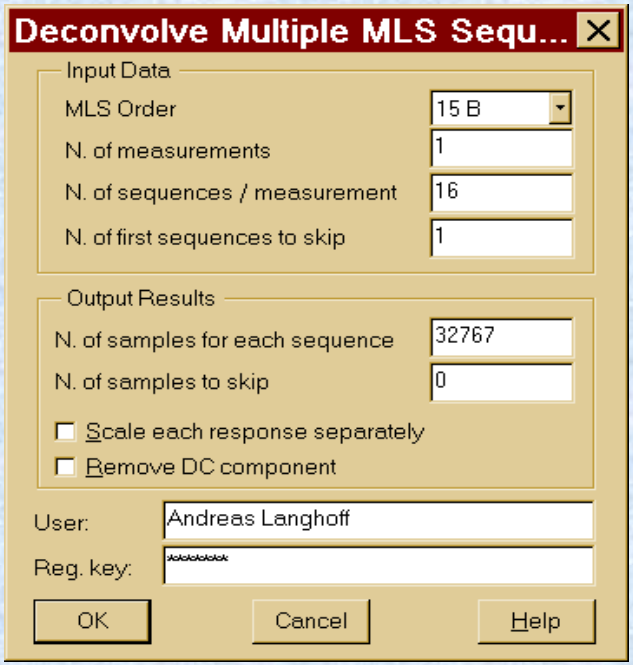

Generate Multiple MLS Sig...
MLS Order
15 B
Amplitude
16384
N. sequences
16
Repetitions
1
OK
Cancel
Help
Generate control pulses on right channel
Control Pulse Event
At the beginning of each repetition
At the beginning of each repetition but first
At the end of each repetition
User
Andreas Langhoff
Reg. key

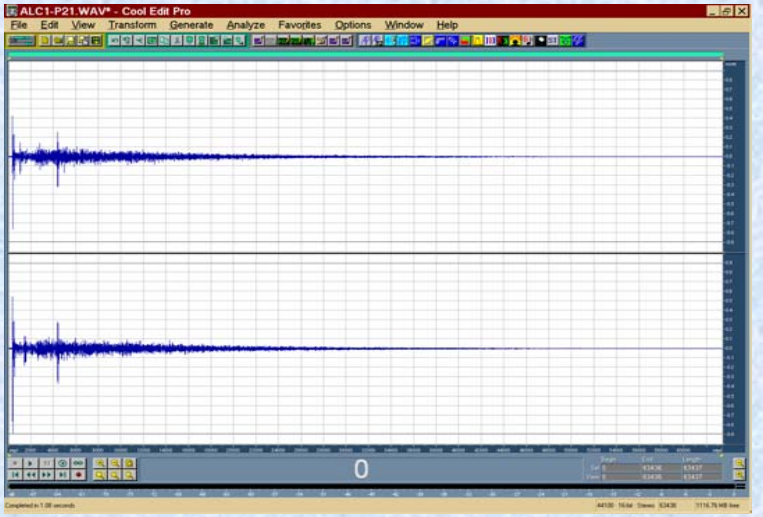

Untitled* - Cool Edit Pro
File Edit View Transform Generate Analyze Favorites Options Window Help

MLS example









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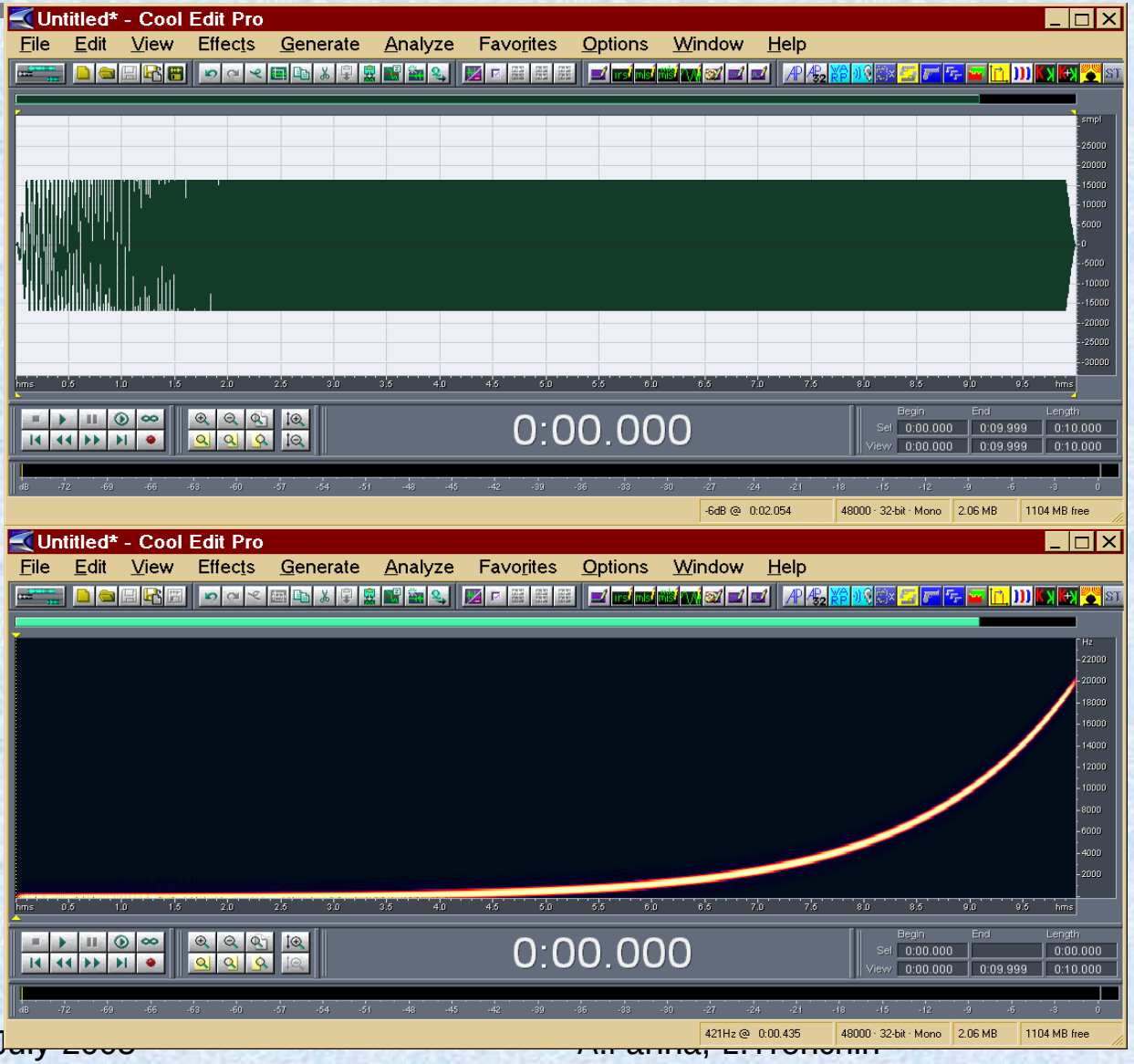
Test signal: Log Sine Sweep

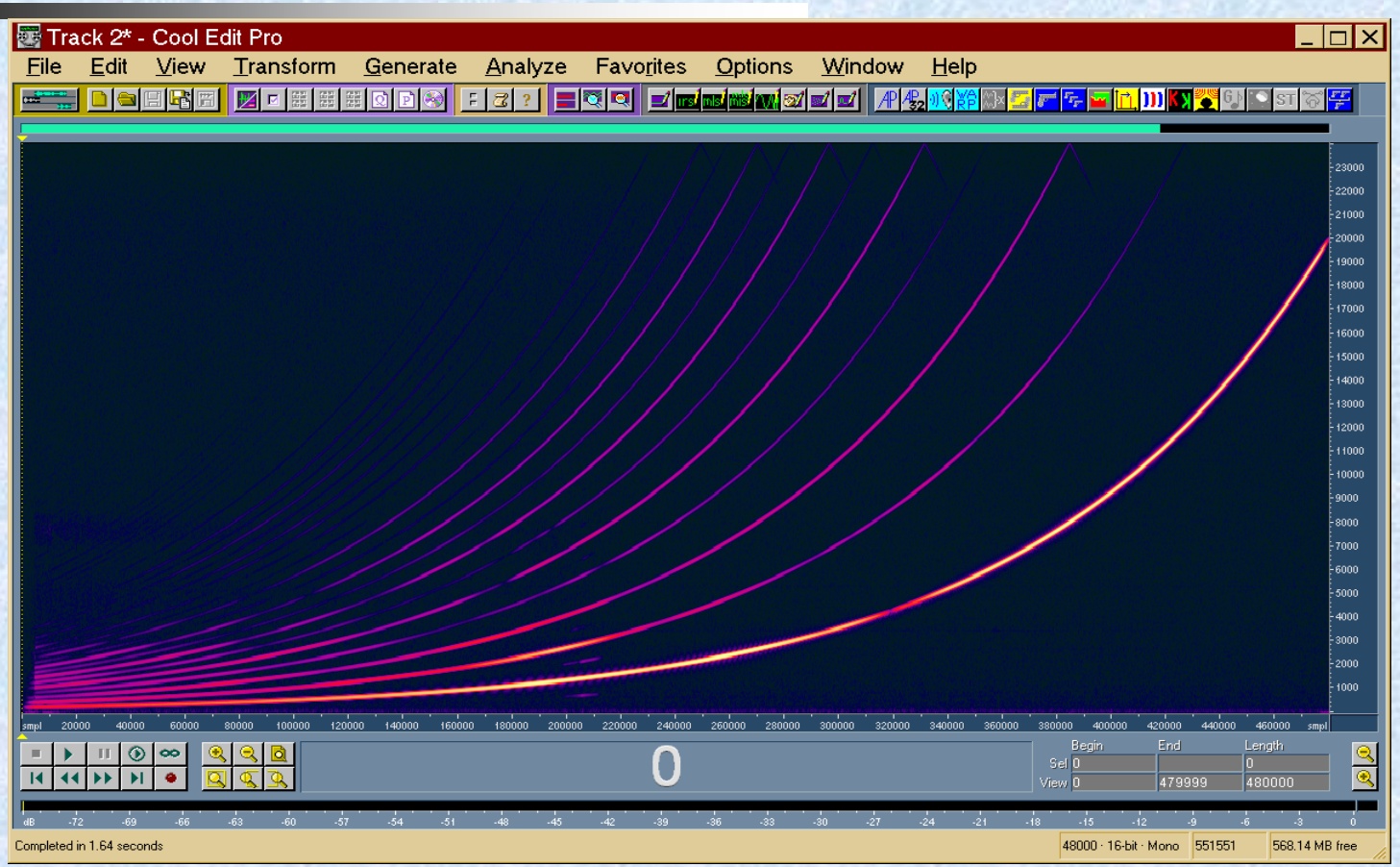
x(t) is a sine signal, which frequency is varied exponentially with time, starting at f_1 and ending at f_2.

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

Test Signal – x(t)



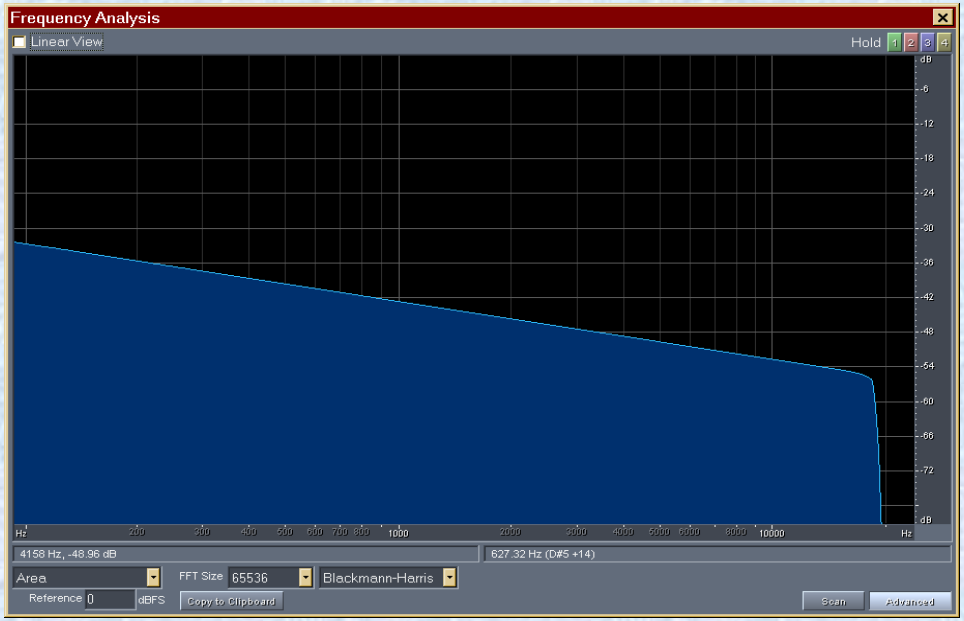
Measured signal - y(t)



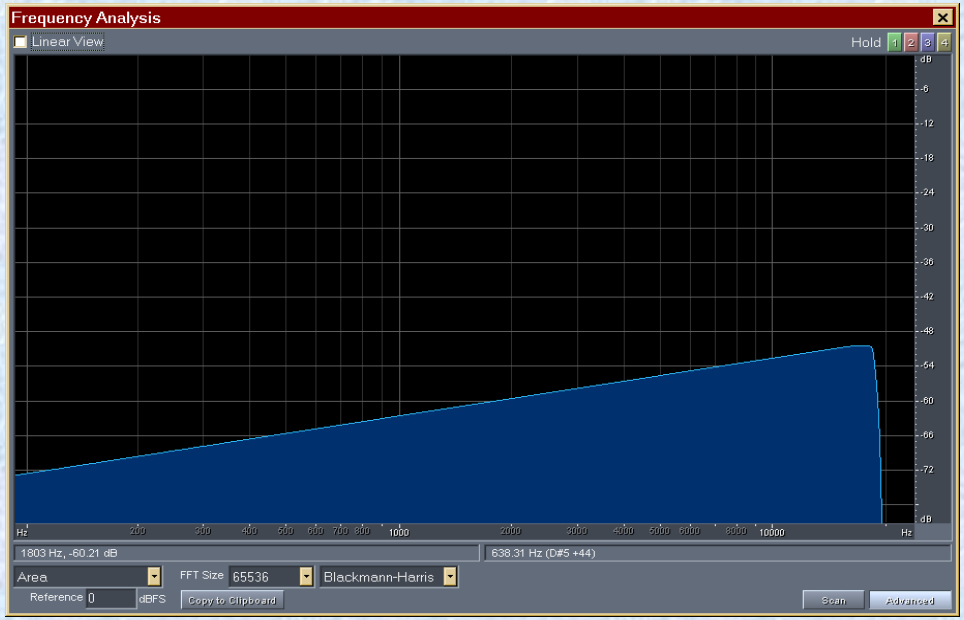
- The not-linear behaviour of the loudspeaker causes many harmonics to appear

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

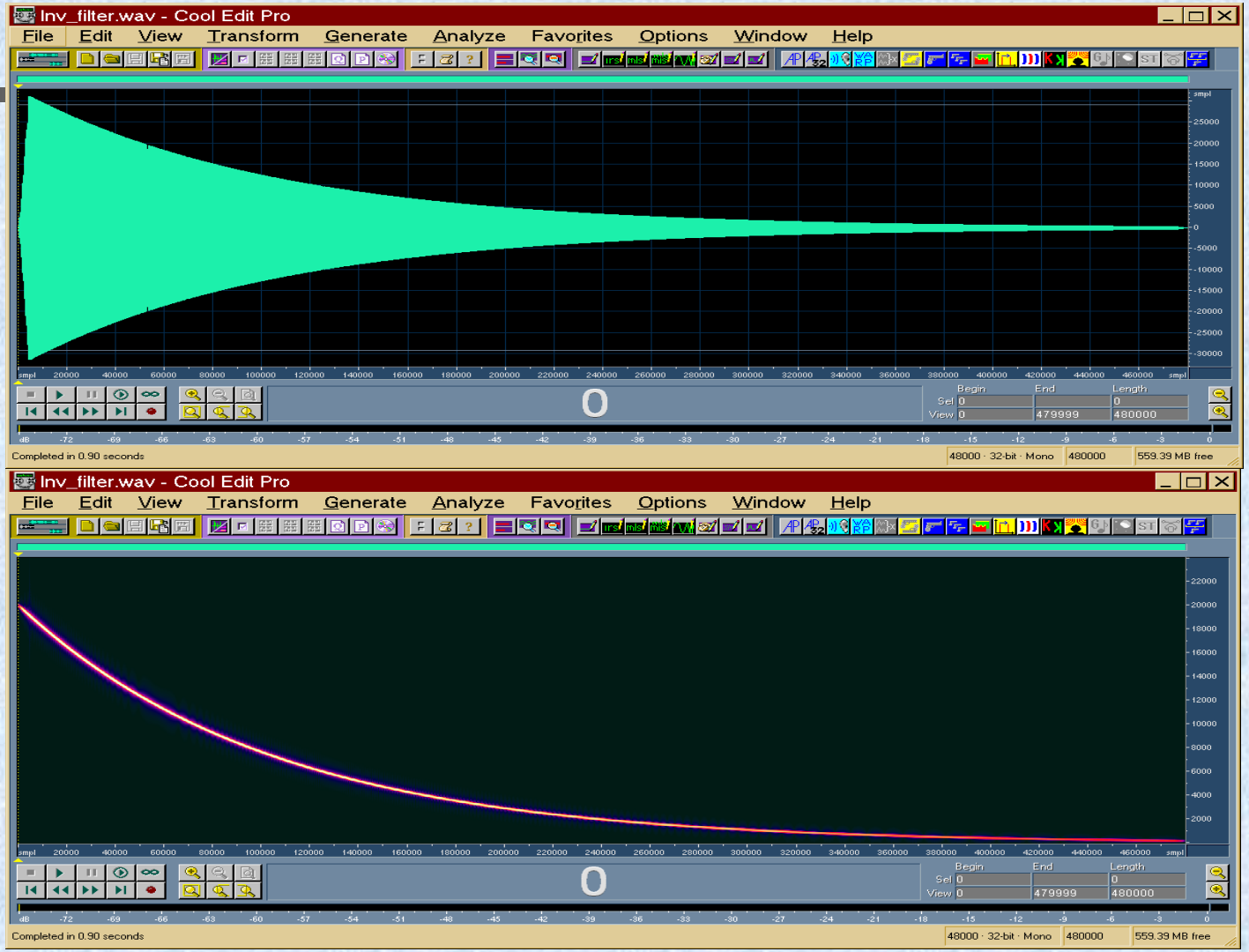


Test Signal x(t)



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) [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

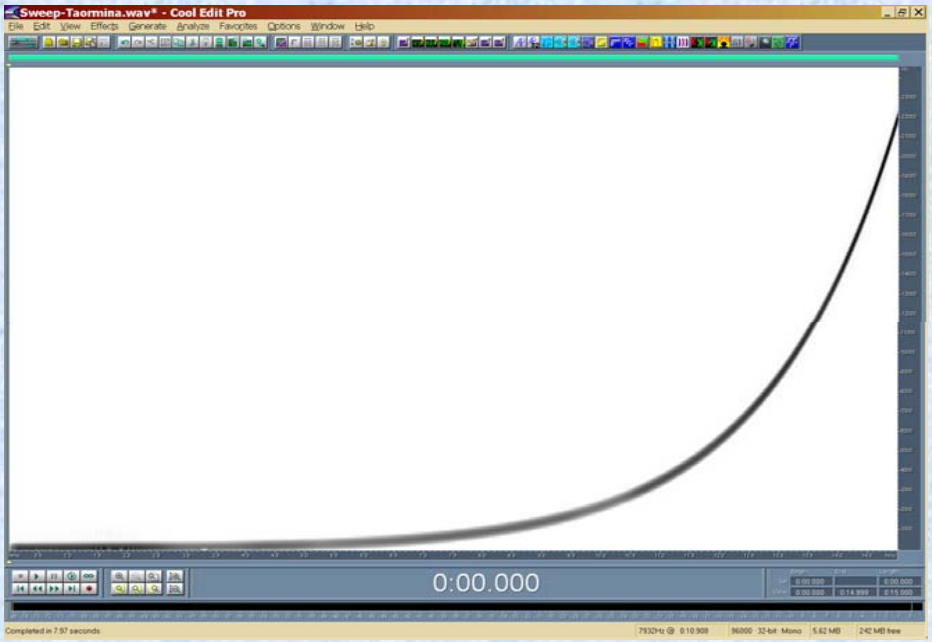
Measurement Setup

- It is possible to measure impulse responses in various formats:
  - Mono (Omnidirectional)
  - Stereo (ORTF)
  - Binaural (Dummy Head)
  - B-format (1st order Ambisonics, Soundfield microphone)
  - WFS (Wave Field Synthesis, circular array)
  - M. Poletti high-order virtual microphones
- Employing a multichannel sound card, all of these measurements can be performed simultaneously

Measurement Parameters

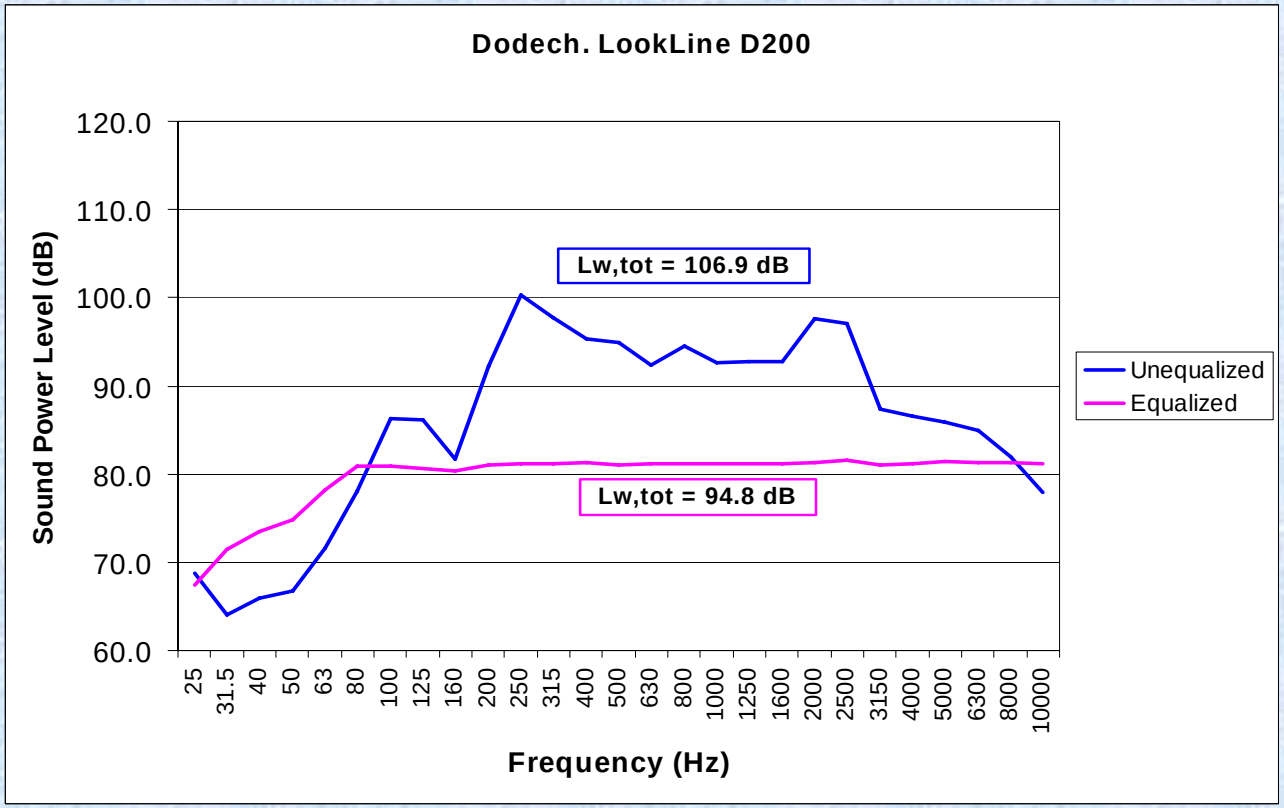
- Test Signal: pre-equalized sweep

Start Frequency	**22 Hz**
End Frequency	**22 kHz**
Sweep length	**15 s**
Silence between sweeps	**10 s**
Type of sweep	**LOG**



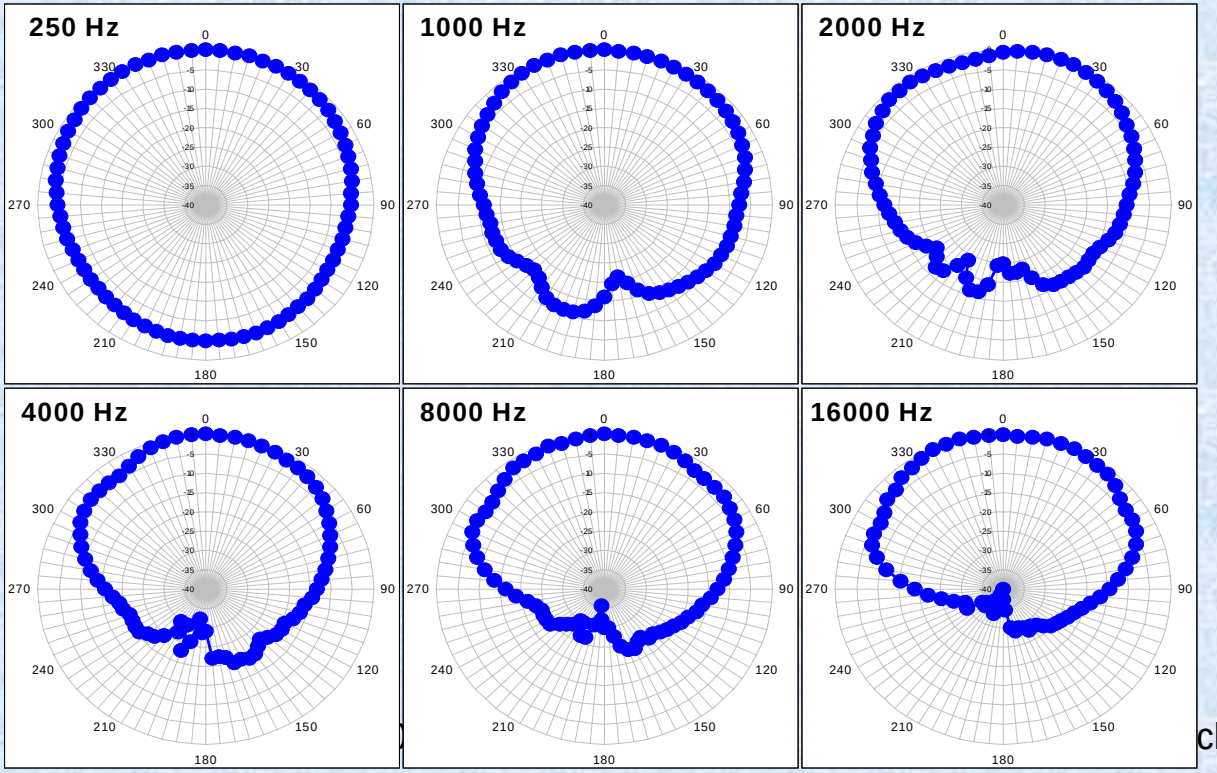
Transducers (sound source #1)

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



Transducers (sound source #2)

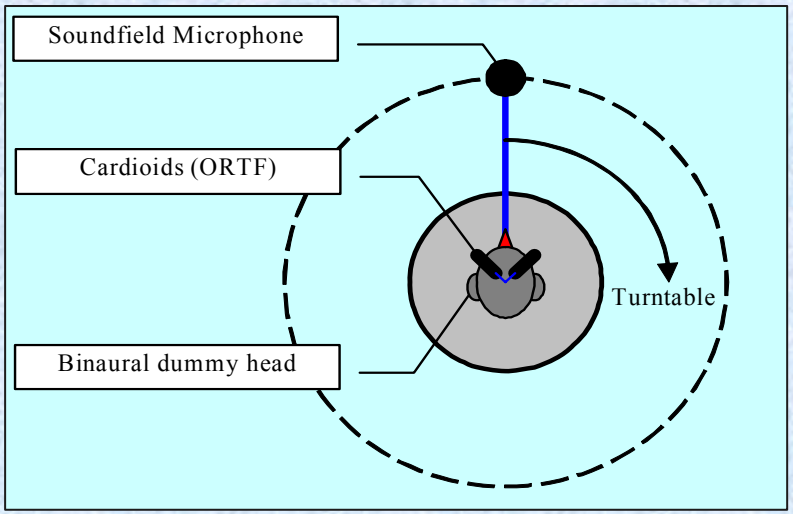
- Genelec S30D reference studio monitor:
  - Three-ways, active multi-amped, AES/EBU
  - Frequency range 37 Hz – 44 kHz (+/- 3 dB)



chin

Transducers (microphones)

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



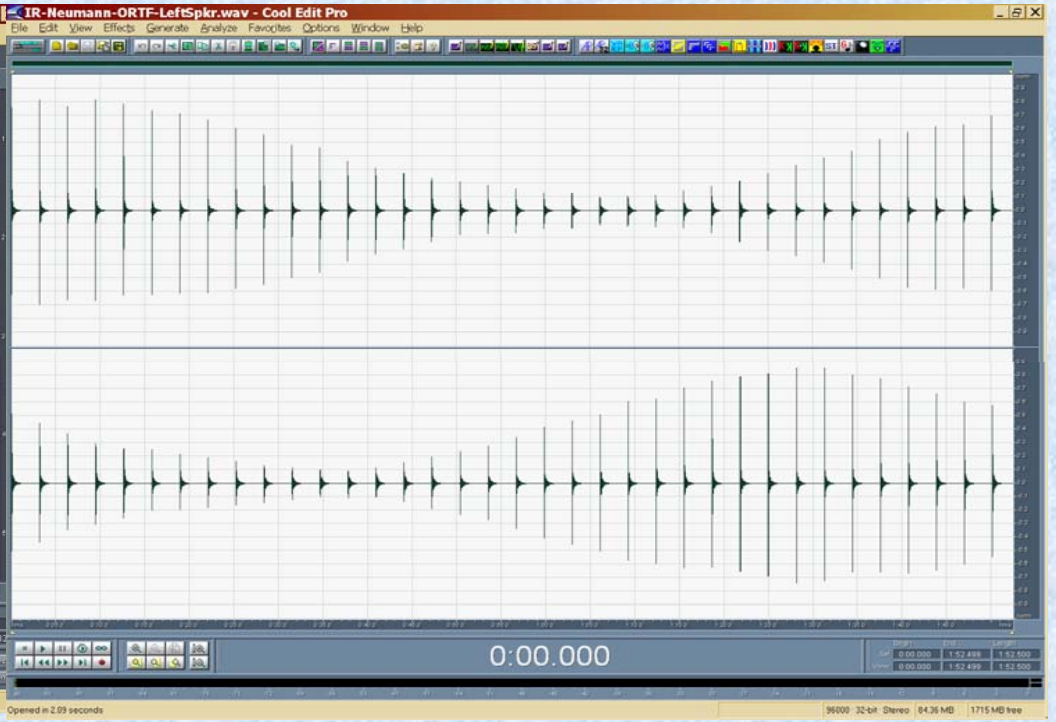
Other hardware equipment

- Rotating Table:
  - Outline ET-1

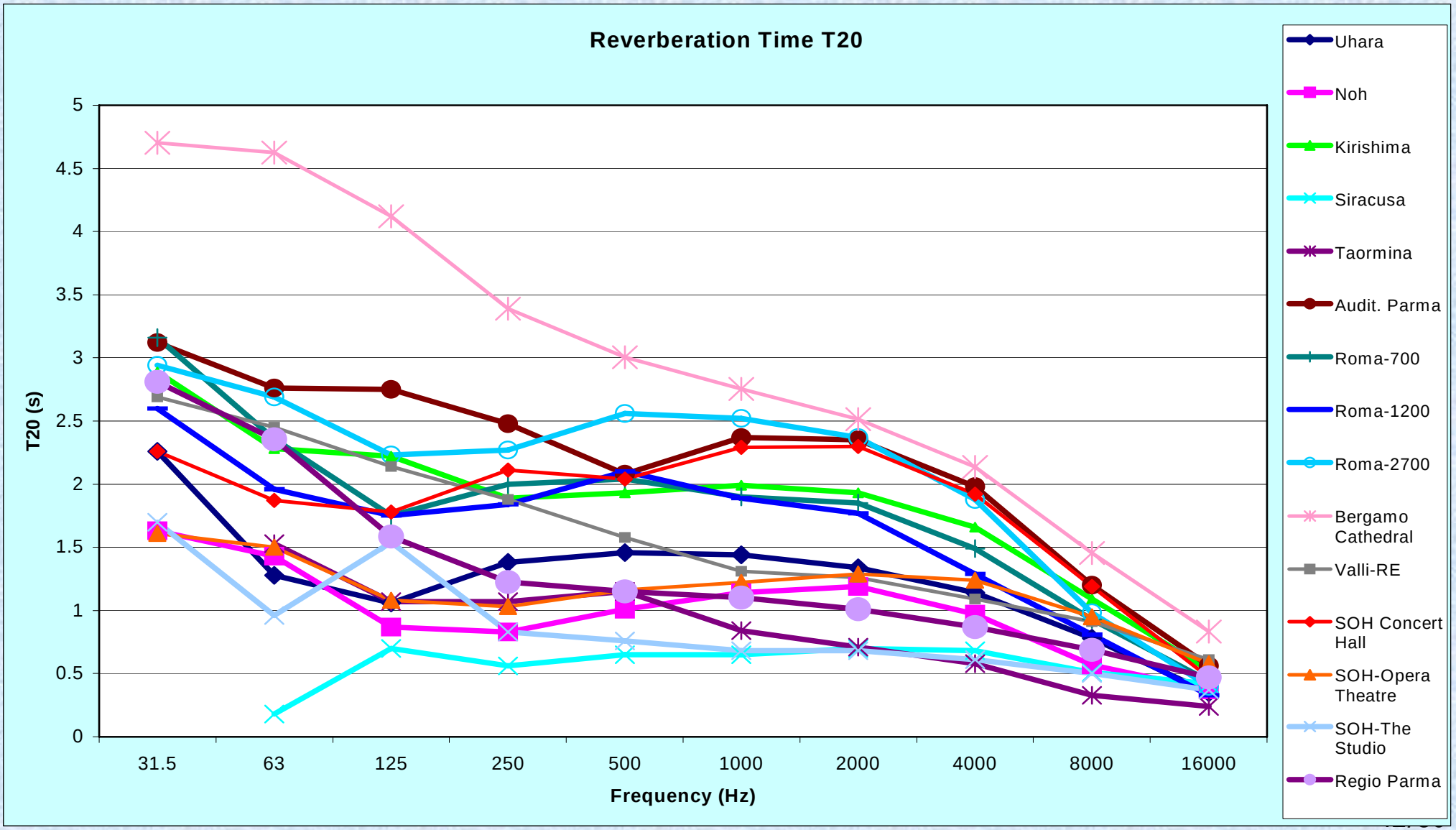
- Computer and sound card:
  - Signum Data Futureclient P-IV 1.8 GHz
  - Aardvark Pro Q-10 (8 ch., 96 kHz, 24 bits)

Measurement procedure

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



Theatres measured



Theaters measured (Waves, 2003)

Uhara

Noh

Kirishima

Siracusa

Taormina

Parma

Roma 700

Roma 1200

Roma 2800

Bergamo

Sydney Opera

Sydney Concert

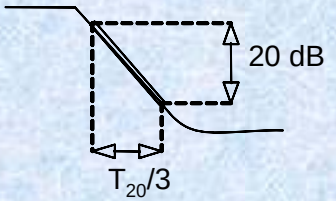
Sydney Studio

Teatro Valli RE

Teatro Regio PR

Acoustical Parameters (ISO 3382)

- Reverberation Time T_{20}:



- Clarity C_{80}:

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

- Definition D:

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

- Center Time T_S:

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

Acoustical Parameters (ISO 3382)

- Strength:

$$G = SPL - L_w + 31 \qquad dB$$

- 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:

$$LF = \frac{\int_{5ms}^{80ms} h_Y{}^2(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_W{}^2(\tau) \cdot d\tau}$$

- LFC:

$$LFC = \frac{\int_{5ms}^{80ms} h_Y(\tau) \cdot h_W(\tau) \cdot d\tau}{\int_{0ms}^{80ms} h_W{}^2(\tau) \cdot d\tau}$$

Analysis of spatial attributes

Acoustical Parameters v. ...

User Defined Reverberation Time Extremes:
(-5. dB , -15. dB)

Enable Noise Correction
EDT without linear regression

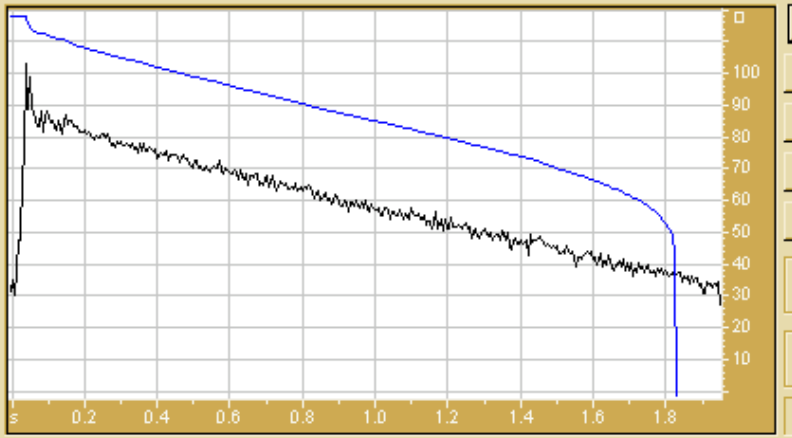
First Arrival Time Threshold (% of FS): 25
Peak SPL value corresponding to FS 120.

Stereo Mode
- 2 Omnidirectional Microphones
- Soundfield Microphone (W/Y)
- Omni/Eight microphone
- p-p Sound Intensity Probe
  d (mm): 12.0 c (m/s): 340.0
- Binaural Dummy Head
  IACC Integration: 0-80 ms (Early)

User: Angelo Farina
Reg. key: *******

OK Cancel Help

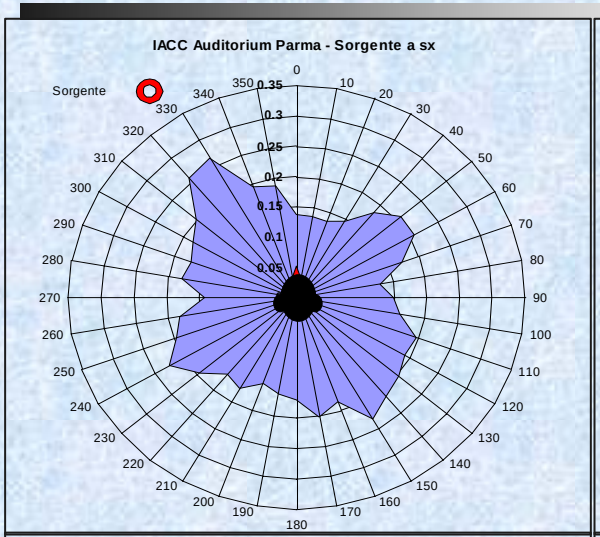
Acoustical Parameters according to ISO3382-1997 (v. 4.0)

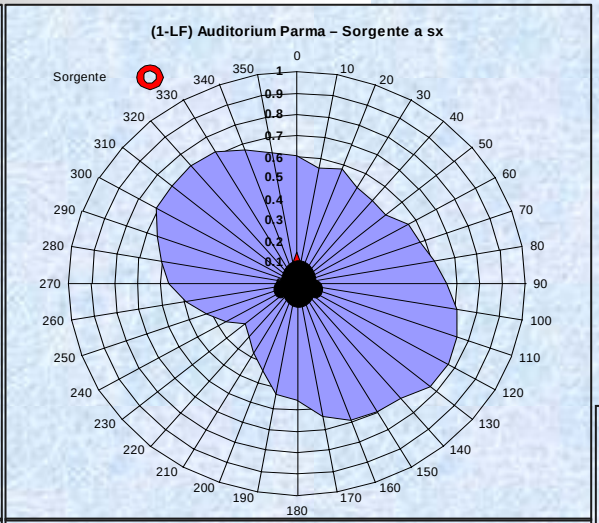


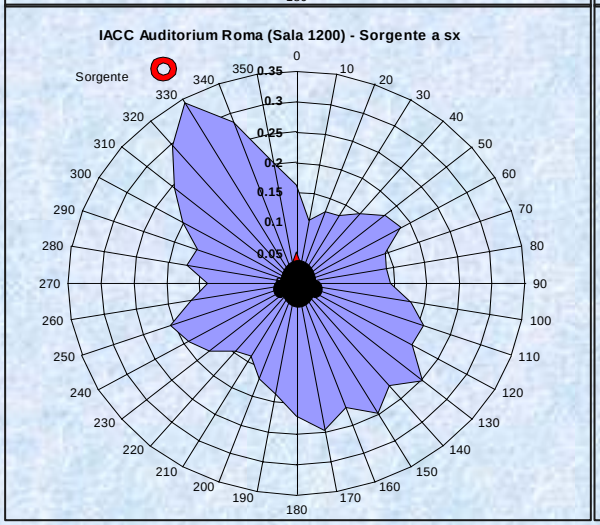
Close Help
OK - keep processed
Save Results to File...
Copy Results to Clipboard
Store G Reference Signal
Channel: Left Right
Tuser limits: (-5. dB, -15. dB)

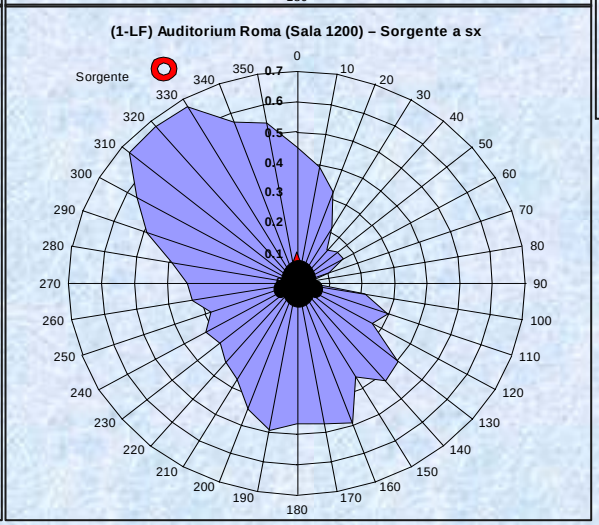
Freq. (Hz)	31.5	63	125	250	500	1k	2k	4k	8k	16k	A	Lin
Signal (dB)	32.48	53.58	57.89	65.10	69.83	70.61	73.62	73.77	78.44	72.79	81.51	82.42
Noise (dB)	-3.86	19.42	26.07	25.44	28.47	30.50	33.80	26.38	11.59	12.20	37.28	37.88
trenGth(dB)	-36.52	-15.42	-11.11	-3.90	0.83	1.61	4.62	4.77	9.44	3.79	4.51	5.42
C50 (dB)	3.66	1.59	-2.55	0.53	1.41	1.47	0.19	1.89	6.95	11.41	3.34	4.02
C80 (dB)	5.25	2.77	-0.70	1.12	2.75	2.54	1.67	3.47	9.30	15.90	4.85	5.57
D50 (%)	69.88	59.07	35.72	53.04	58.04	58.35	51.08	60.74	83.22	93.27	68.34	71.63
Ts (ms)	122.57	115.34	177.75	102.66	87.02	93.68	106.23	76.41	24.35	11.29	61.63	54.19
EDT (s)	1.70	1.66	2.98	1.60	1.86	2.07	2.13	1.68	0.95	0.17	1.57	1.46
T20 (s)	2.73	2.07	2.48	2.15	2.10	2.26	2.19	1.79	1.03	0.53	2.00	1.96
r T20	0.99	0.98	0.99	1.00	1.00	1.00	1.00	1.00	1.00	1.00	1.00	1.00
T30 (s)	2.49	2.54	2.65	2.26	2.15	2.24	2.19	1.83	1.05	0.57	2.06	2.05
r T30	0.99	0.97	0.99	1.00	1.00	1.00	1.00	1.00	1.00	1.00	1.00	1.00
Tuser (s)	3.20	1.74	2.05	2.27	1.97	2.19	2.29	1.75	0.99	0.56	1.87	1.78
r Tuser	0.91	0.98	0.98	0.98	1.00	1.00	1.00	1.00	1.00	0.94	0.99	0.99
IACC (Early)	0.99	0.86	0.70	0.78	0.70	0.59	0.23	0.18	0.67	0.37	0.42	0.47
t IACC (ms)	0.24	0.21	0.16	0.23	0.22	0.24	0.29	0.67	0.23	0.24	0.23	0.23
w IACC (ms)	0.00	1.89	1.09	0.52	0.27	0.16	0.07	0.04	0.03	0.02	0.03	0.03

Polar diagrams of IACC and (1-LF)





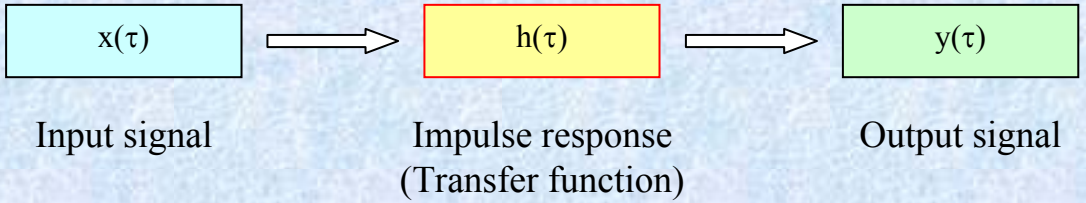




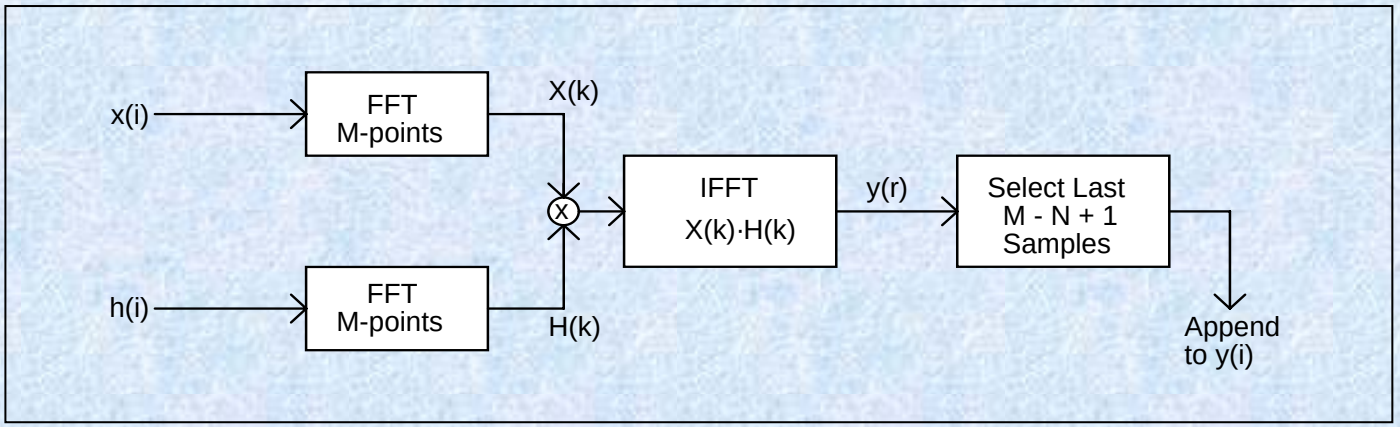
Auditorium	1-LF	IACC
Parma	0.725	0.266
Roma	0.676	0.344

Auralization by convolution

Auralization is the process for making audible a virtual acoustical environment, obtained by convolving anechoic signal with the room's impulse response



The convolution operation is very computationally intensive if implemented directly in time domain. However, it becomes very efficient if performed in frequency domain through the "overlap-and-save" FFT algorithm



Convolution (1):

Convolution of a continuous input signal *x(t)* with a linear filter characterized by an Impulse Response *h(t)* yields an output signal *y(t)*:

$$y(t) = x(t) \otimes h(t) = \int_{-\infty}^{\infty} x(t-\tau) \cdot h(\tau) \cdot d\tau$$

If the input signal and the Impulse Response are digitally sampled ($t = i \cdot \Delta t$) and the Impulse Response has finite length N, we can write:

$$y(i) = \sum_{j=0}^{N-1} x(i-j) \cdot h(j)$$

Convolution (2):

$$y(i) = \sum_{j=0}^{N-1} x(i-j) \cdot h(j)$$

Multiply and ACcumulate

```
y:=0;
FOR n:=0 TO N-1 DO
    y:= y + a[n]·x[n];
```

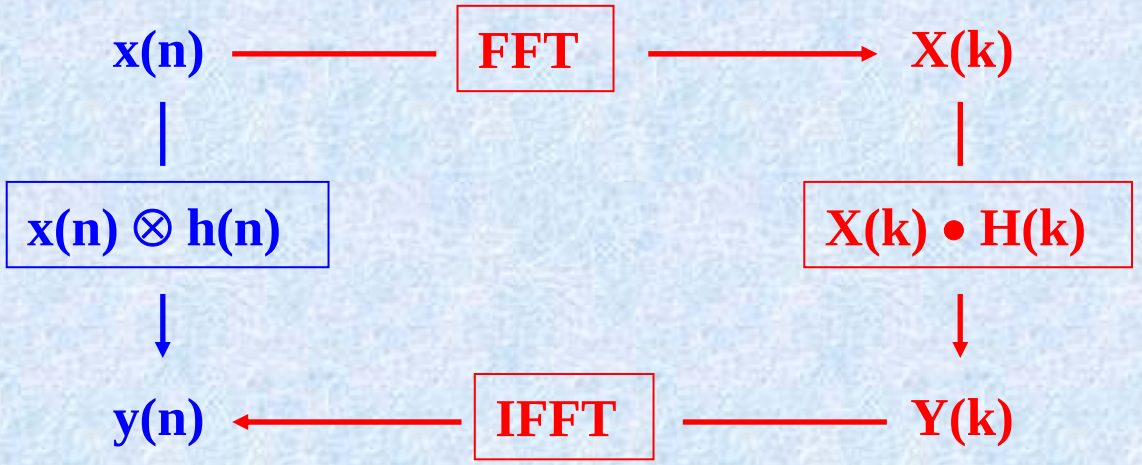
On a DSP board this instruction is performed in one cycle

- **Clock core = 100 MHz**
- **Sample frequency f_S = 48 KHz**

⇒ **Upper limit is 2000 MAC per sample**

Filtering in the frequency domain:

Could be better operated in the frequency domain



Problems

- **Filtering can be performed only when all data are available**
- **Order of FFT is too high.**

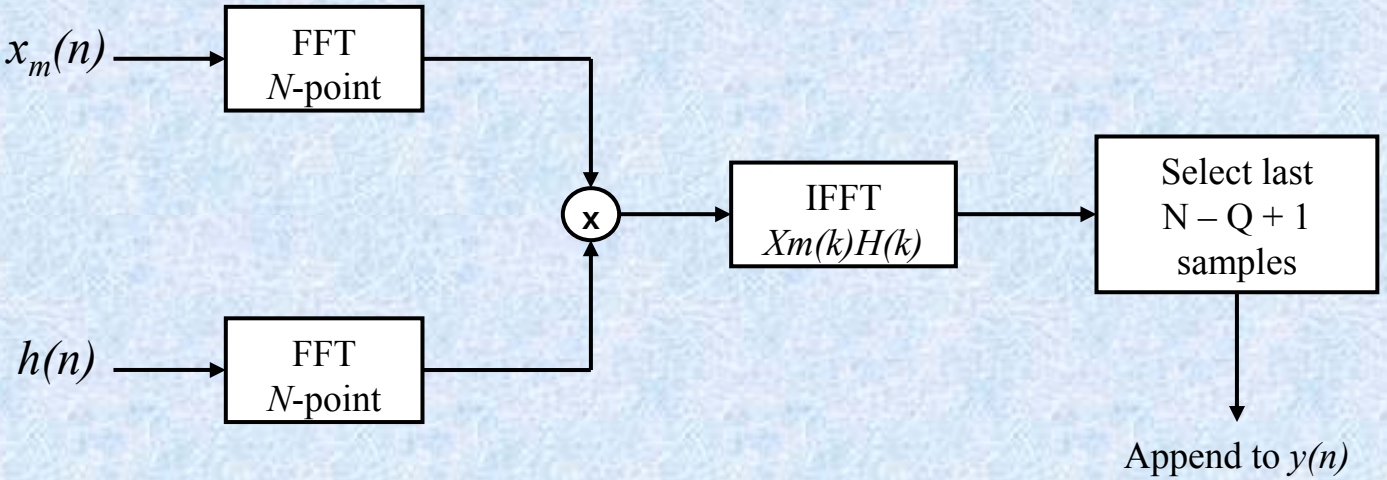
Solution

- **Overlap & Save algorithm.**

Overlap & Save algorithm:

Industrial Engineering Dept.
University of Parma – Italy

Overlap & Save convolution process:



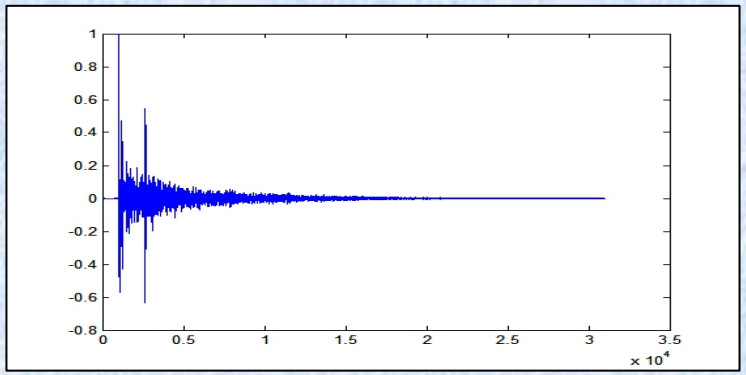
Problems

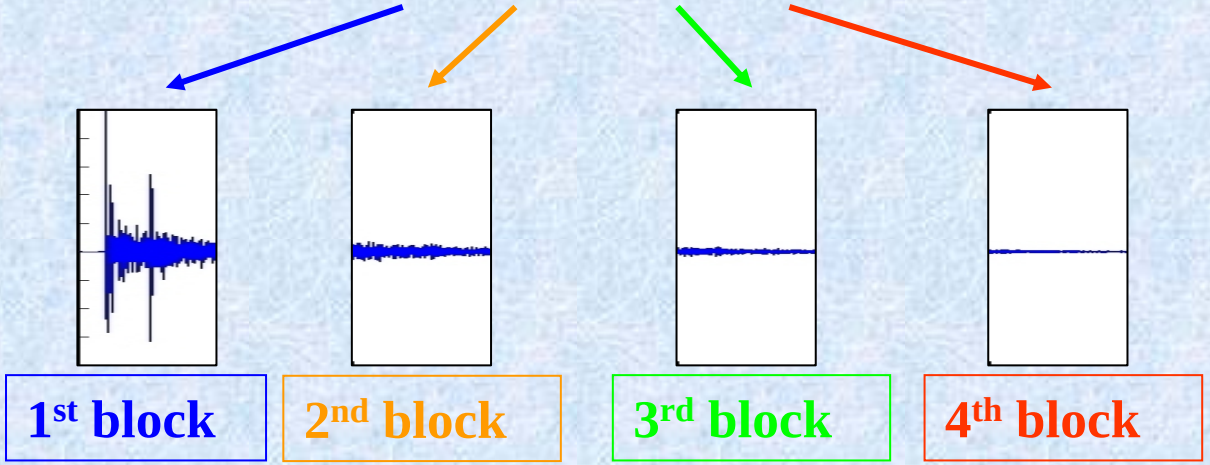
- **Latency between Input and Output data is too high.**
- **Requires to allocate very large memory buffers.**

Solution

- **Uniformly-partitioned Overlap & Save algorithm.**

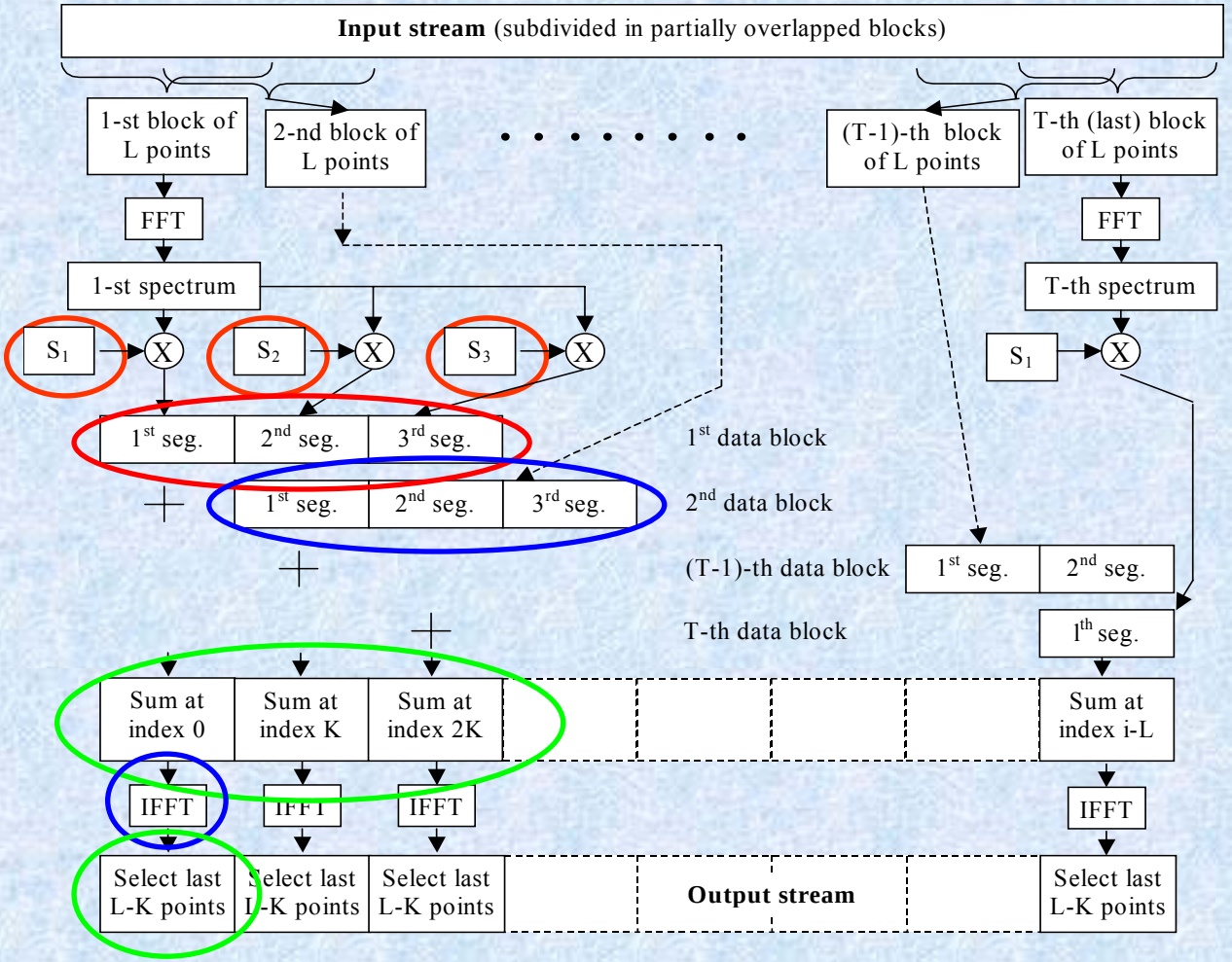
Uniformly-partitioned O&S (1)





The impulse response *h(n)* is partitioned in a reasonable number *P* of equally-sized blocks (i.e. P = 4), where each block is *K* points long.

Uniformly-partitioned O&S (2):



Each block is treated as a separate result, zero-padded to multiple of L and transformed with FFT in order to obtain a collection of P frequency-domain blocks.

Every filter S_i is convolved, using the Overlap&Save method, to L-point blocks of input data (each block begins $L-K$ points after the previous).

The P results are added with the corresponding ones of the previous filters in order to obtain a unique stream of frequency-domain data.

Then, an IFFT is done on each block, and at the end the first K points of each block are kept.

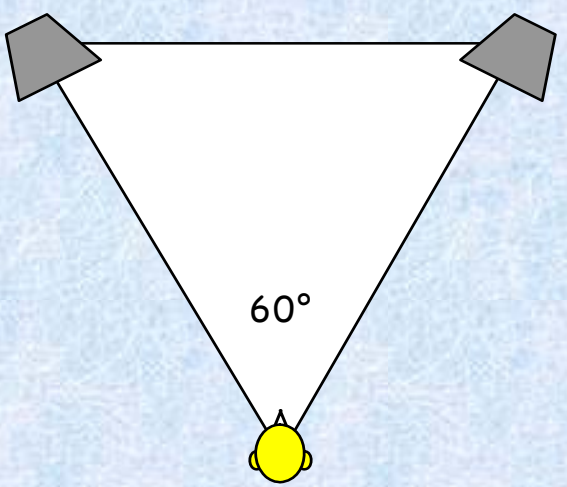
Anechoic recording

(anechoic room at ASK Industries, Reggio Emilia)

Auralization types

- Stereo (ORTF on 2 standard loudspeakers at +/- 30°)
- Binaural on headphones
- Binaural on loudspeakers (Stereo Dipole)
- 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)

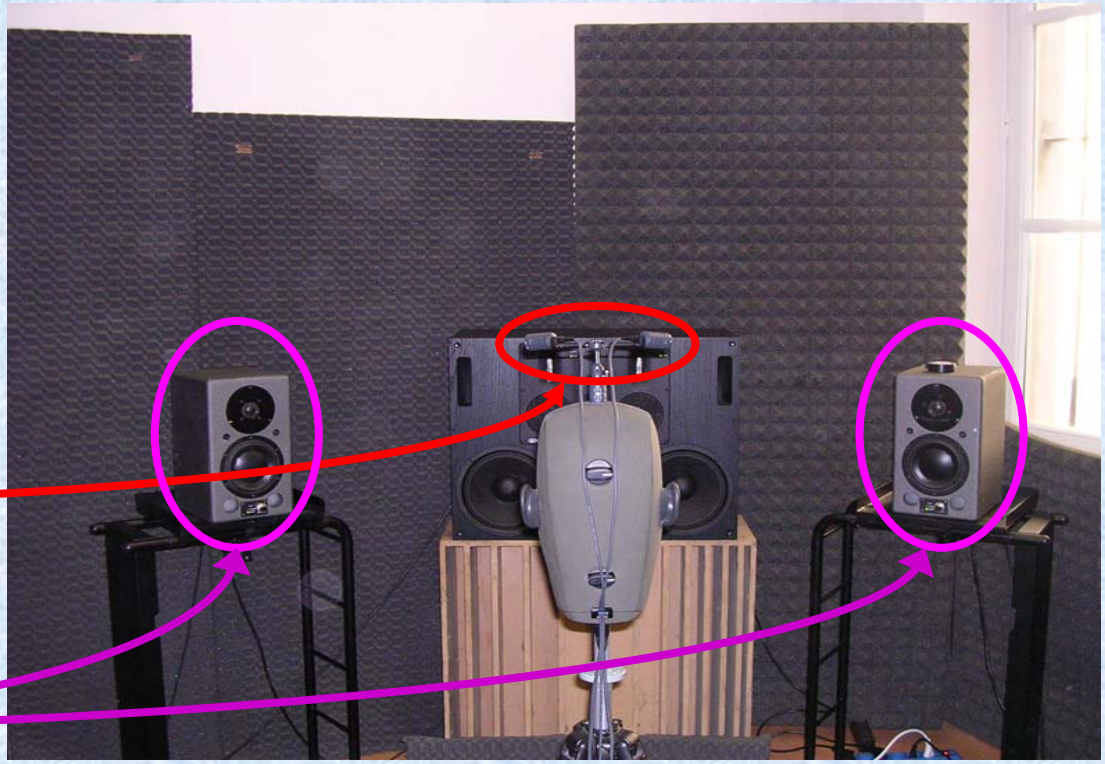
ORTF STEREO



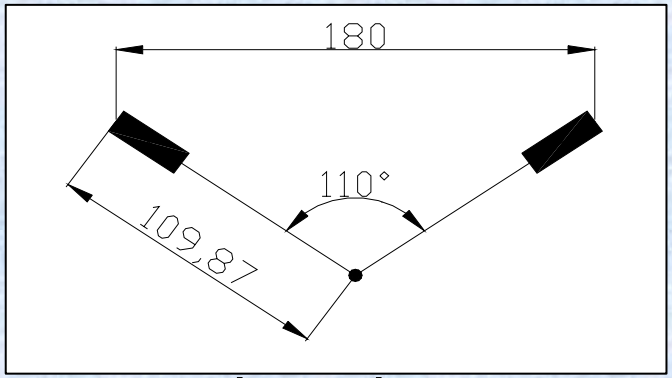
- *Dynaudio* self-powered studio monitors

-The sound is picked up by the two cardioid microphones

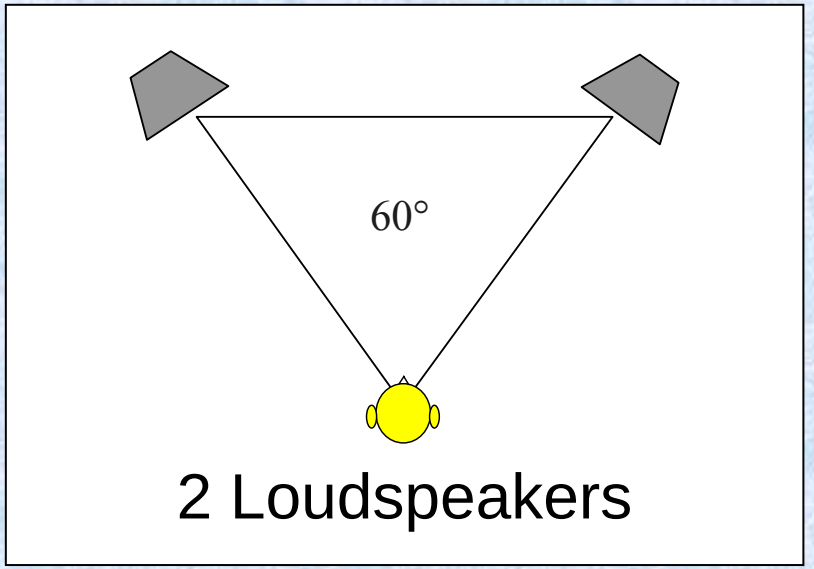
- Each of the two signals is passed through a proper inverse filter, computed after a measurement performed placing the ORTF microphones at the listening position, in front of the loudspeakers



ORTF Stereo



2 Microphones



2 Loudspeakers

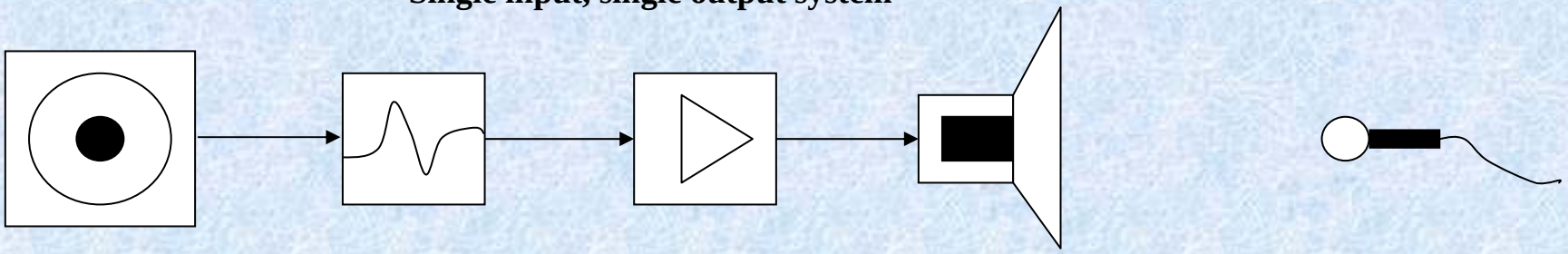
Playback occurs over a pair of loudspeakers, in the standard configuration at angles of +/- 30°, each being fed by the signal of the corresponding microphone

Headphones

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

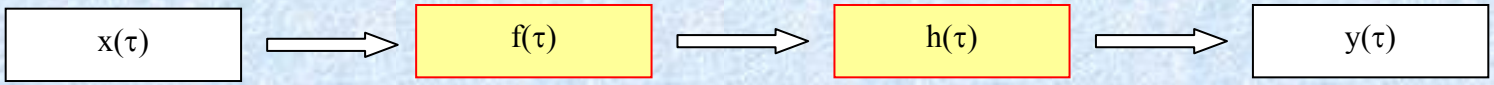
Theory of inverse filters

Single input, single output system



Signal source (CD) Filter Amplifier Loudspeaker Microphone

Block diagram



Input signal

Filtering coefficients

System's Impulse Response (Transfer function)

Output signal

Combined transfer functiom

As all the stages are linear:

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

The gola 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 δ(i) is the Dirac's Delta function (a single sample having unit value, peceded and followed by thousands of zeroes) – this way the total effect of the filter+system is simply a delay of a few millisecond, with no other evident alteration

Theory of Kirkeby inversion

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

$$H(\omega) = \mathrm{FFT}\left[h(\tau)\right]$$

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

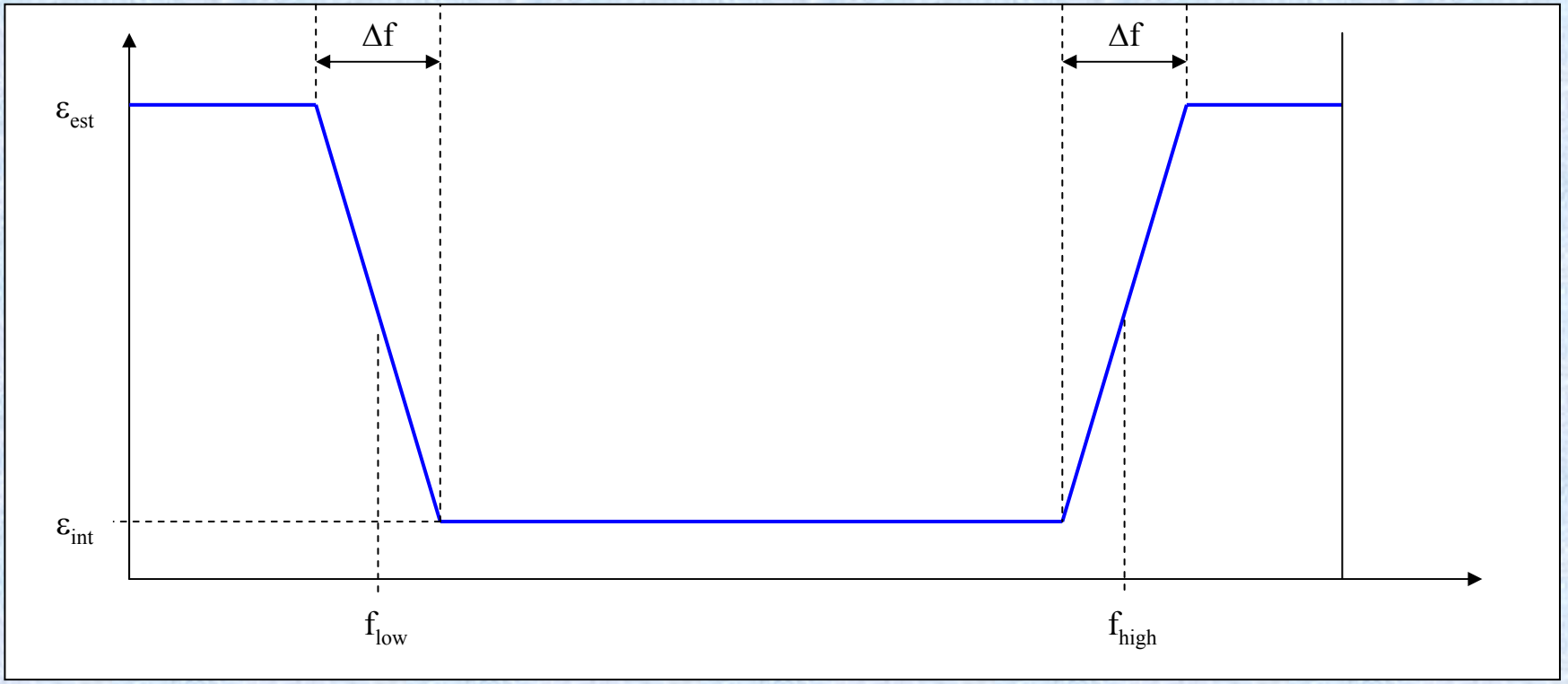
$$F(\omega) = \frac{\mathrm{Conj}\left[H(\omega)\right]}{\mathrm{Conj}\left[H(\omega)\right] \cdot H(\omega) + \varepsilon(\omega)}$$

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

$$f(\tau) = \mathrm{IFFT}\left[F(\omega)\right]$$

Parametro di regolarizzazione

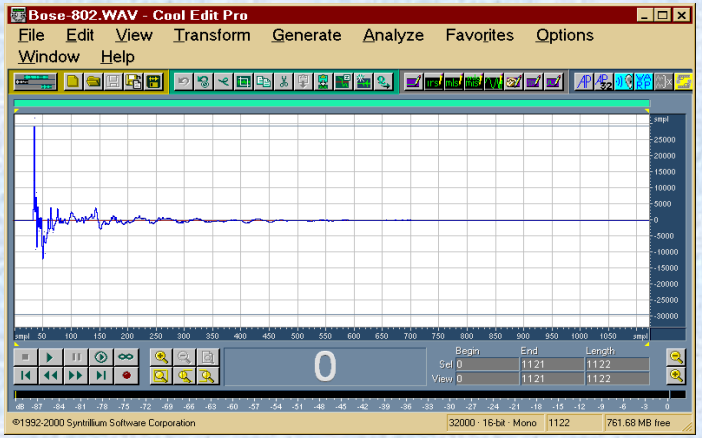
Regularization parameter $\varepsilon(\omega)$

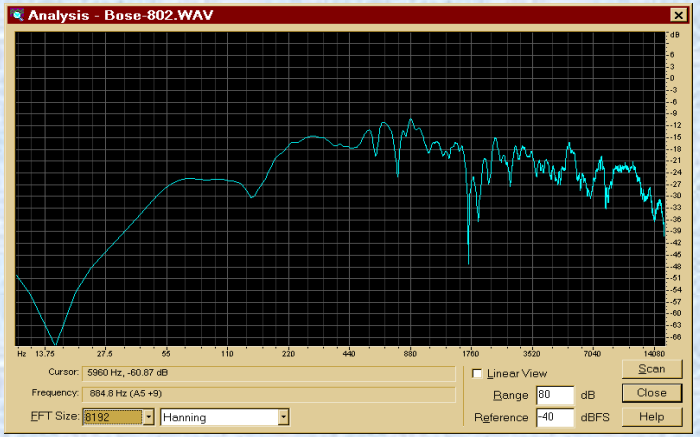


Changing the value of the regularization parameter allows for a very accurate filter at central frequency, and progressively a less aggressive filtering at very low or very high frequencies.

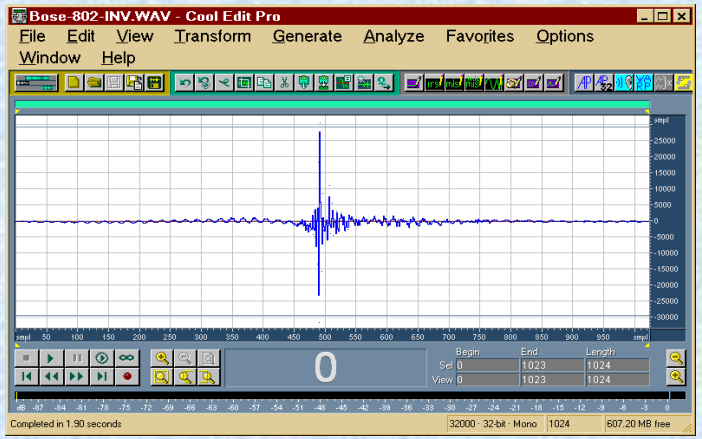
Inverse filter example

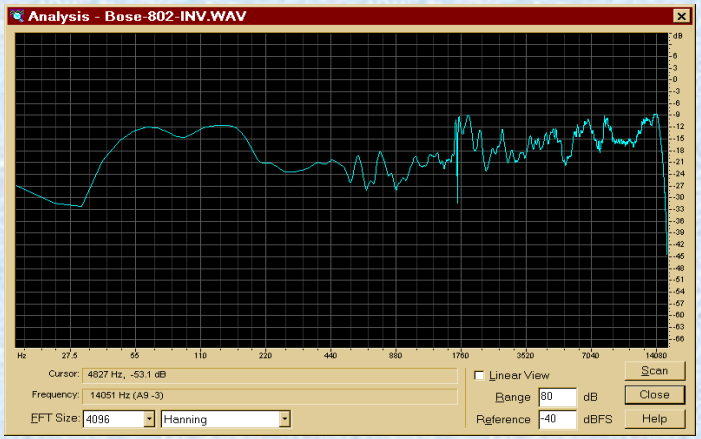
- **System's impulse response**





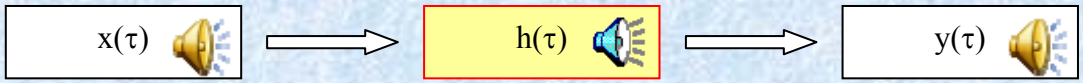
- **Inverse Filter**





Inverse filter example

Not filtered system

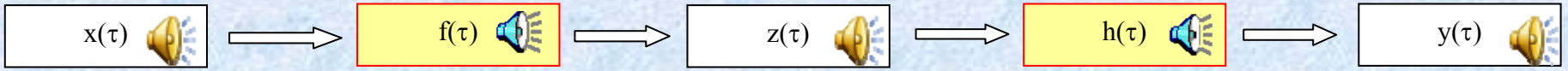


Input signal

System

Output signal

System with filter



Input signal

Filtering coefficients

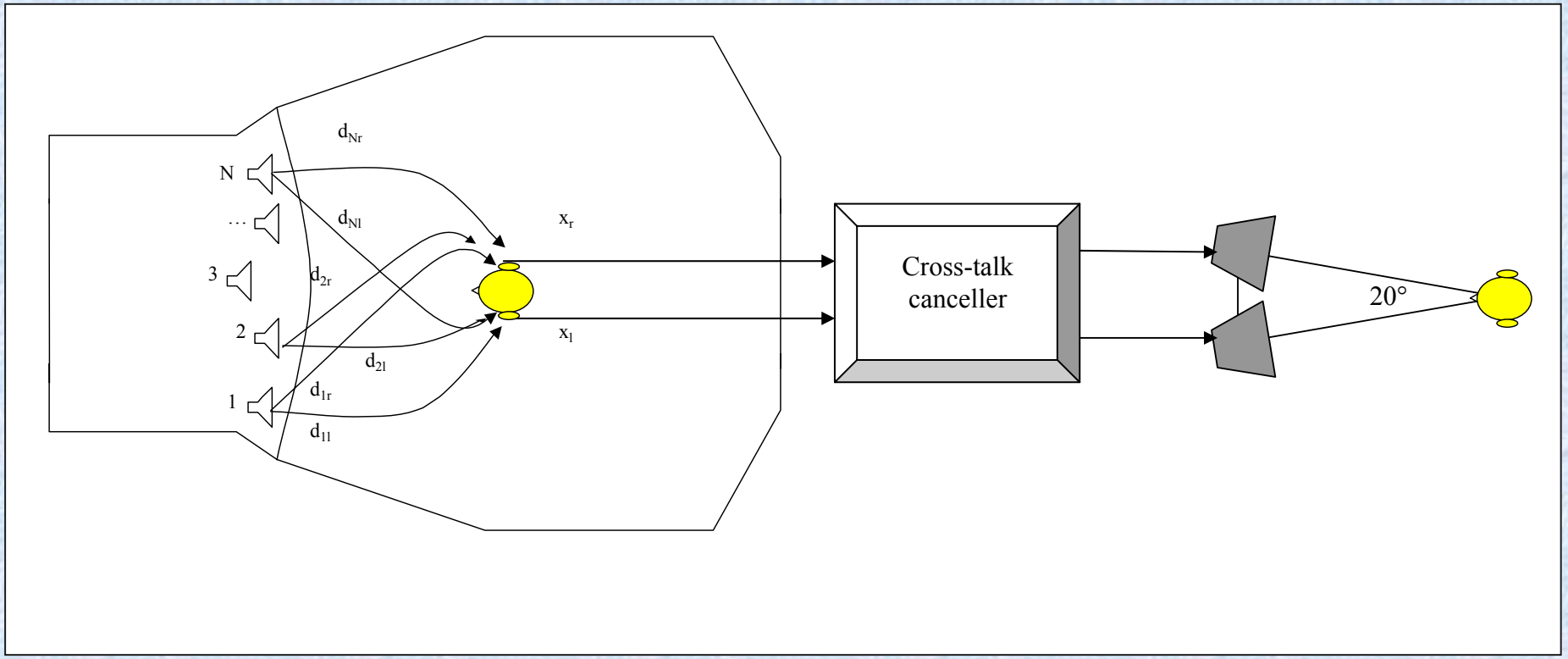
Filtered signal

System

Output signal

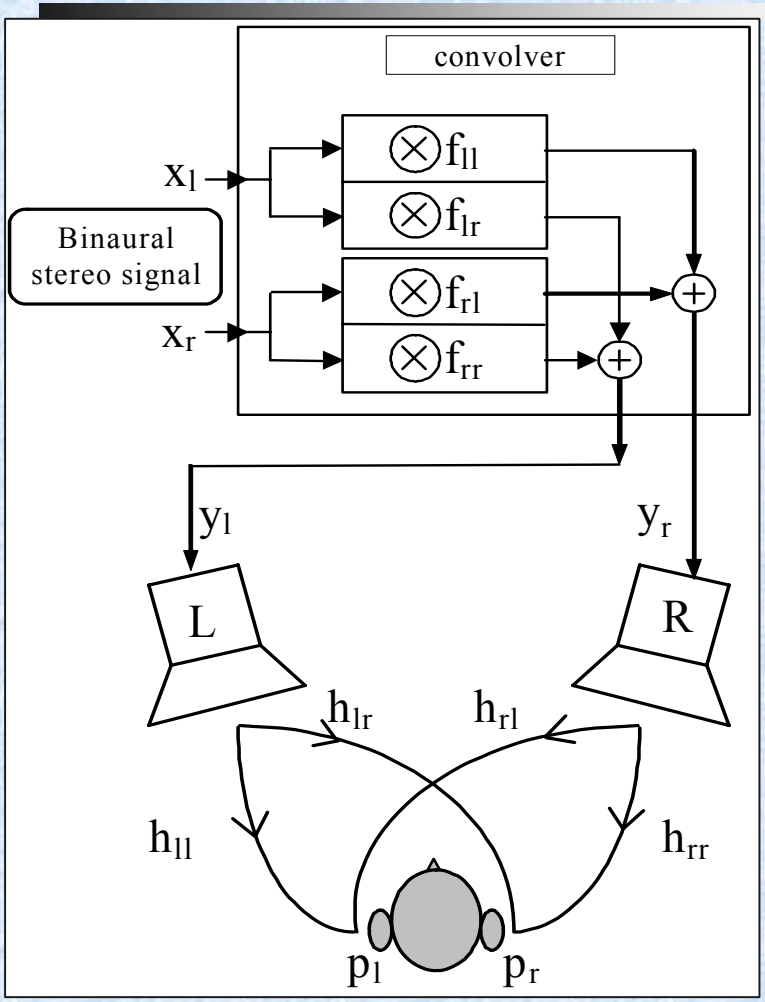
Binaural (Stereo Dipole)

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



Reproduction occurs over 2 loudspeakers angled at +/- 10°, being fed through a “cross-talk cancellation” digital filtering system

Binaural (Stereo Dipole#2)

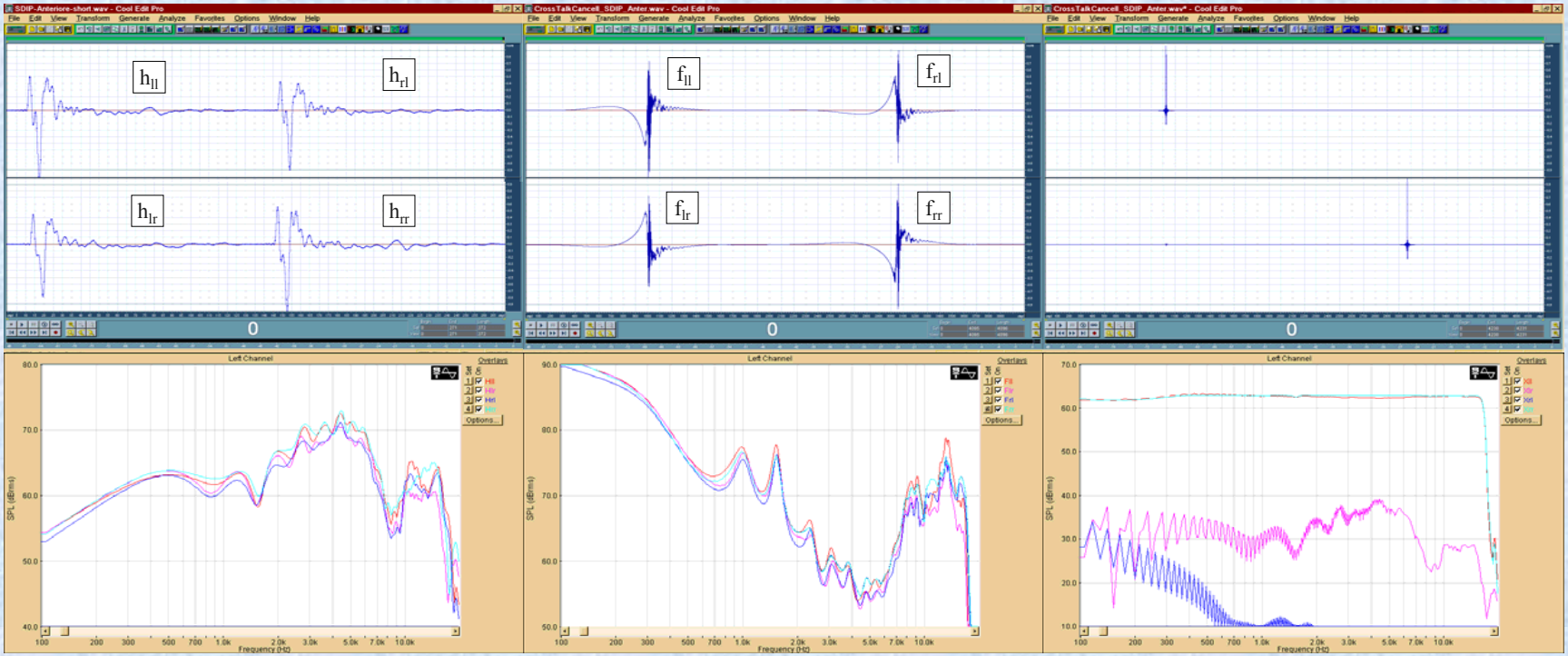


Binaural (Stereo Dipole#3)

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

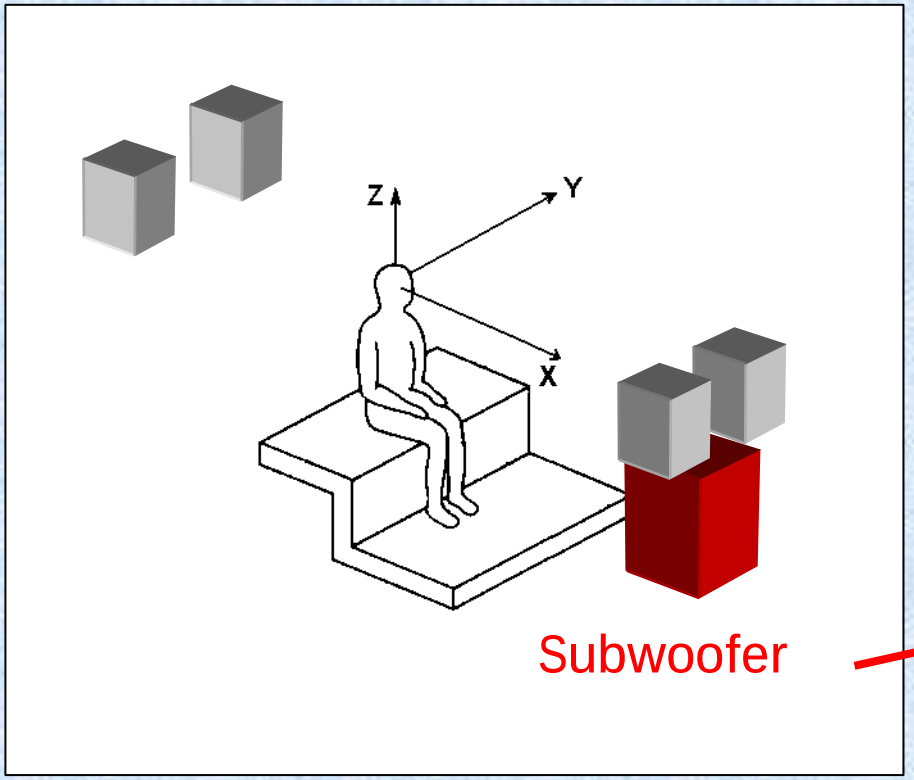
$$C(\omega) = FFT(h_{ll}) \cdot FFT(h_{rr}) - FFT(h_{lr}) \cdot FFT(h_{rl})$$

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



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)

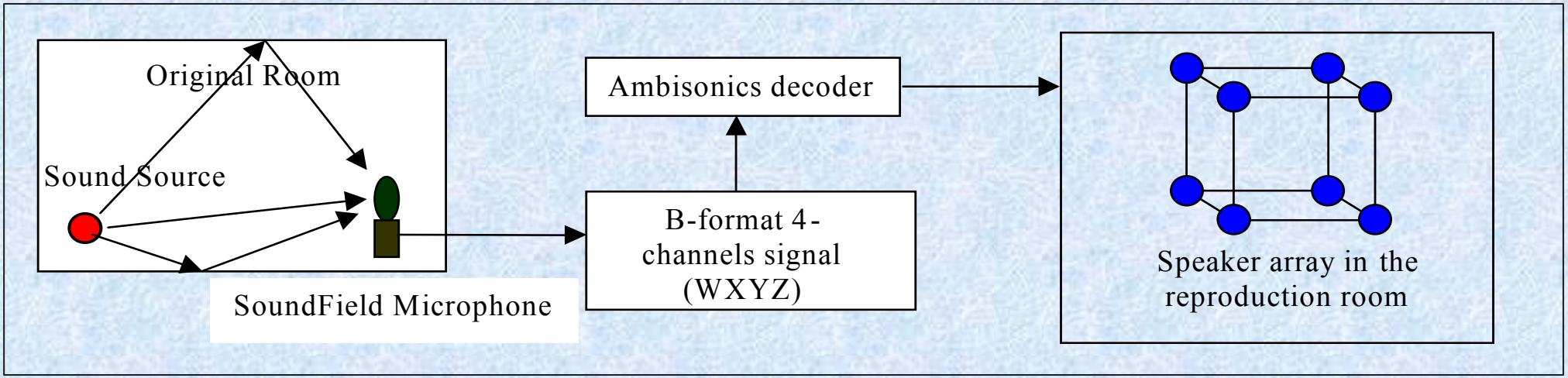
Frontal

Quested 2108 monitors

Rear

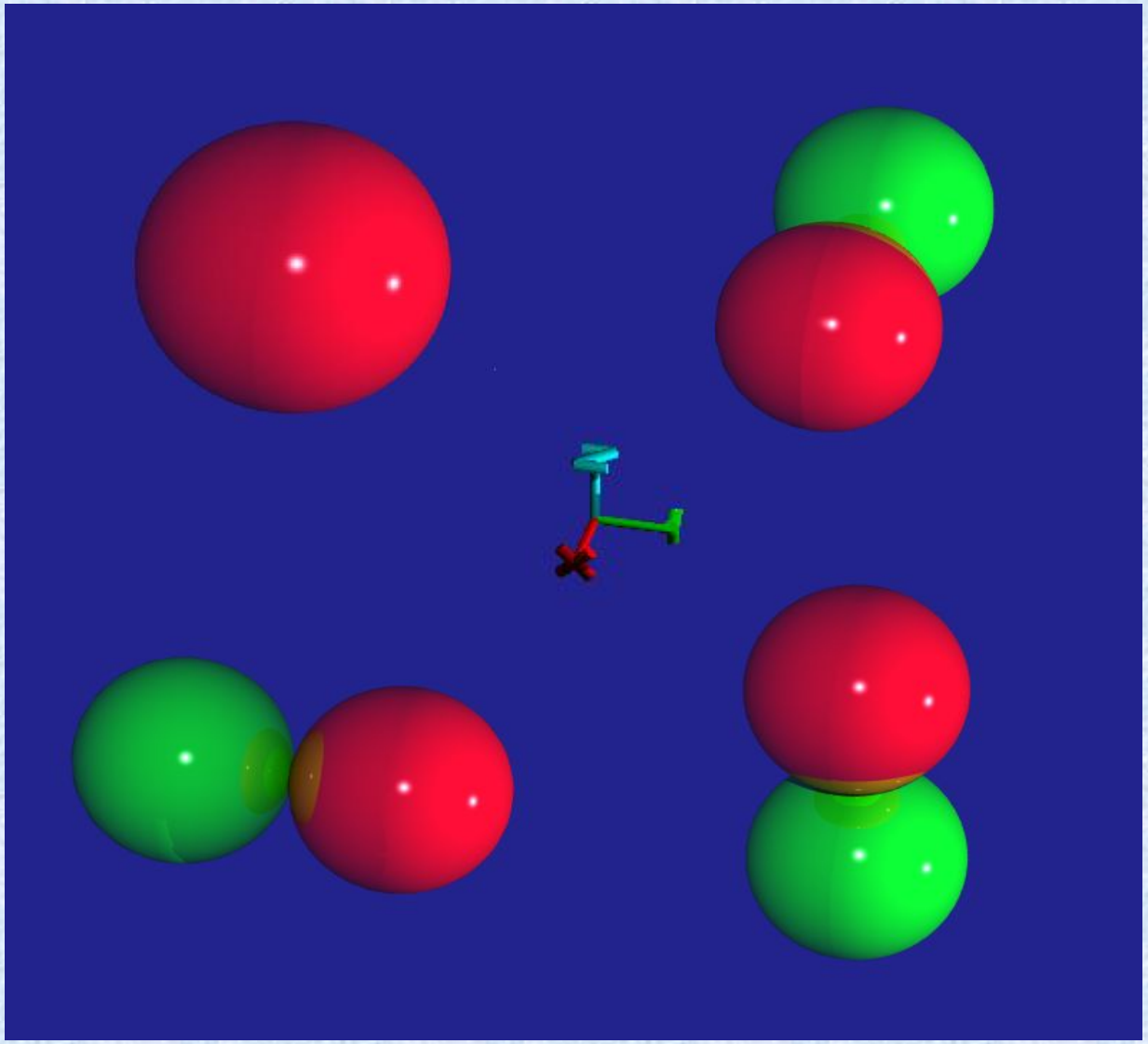
Quested F11P monitors

Ambisonics 3D 1st order



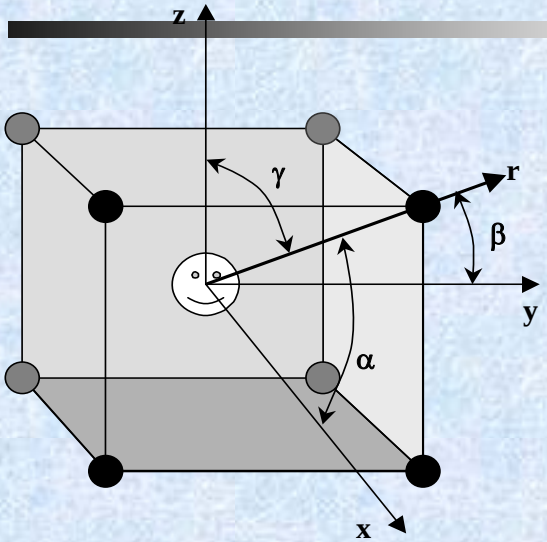
Reproduction occurs over an array of 8-24 loudspeakers, through an Ambisonics decoder

The B-format components

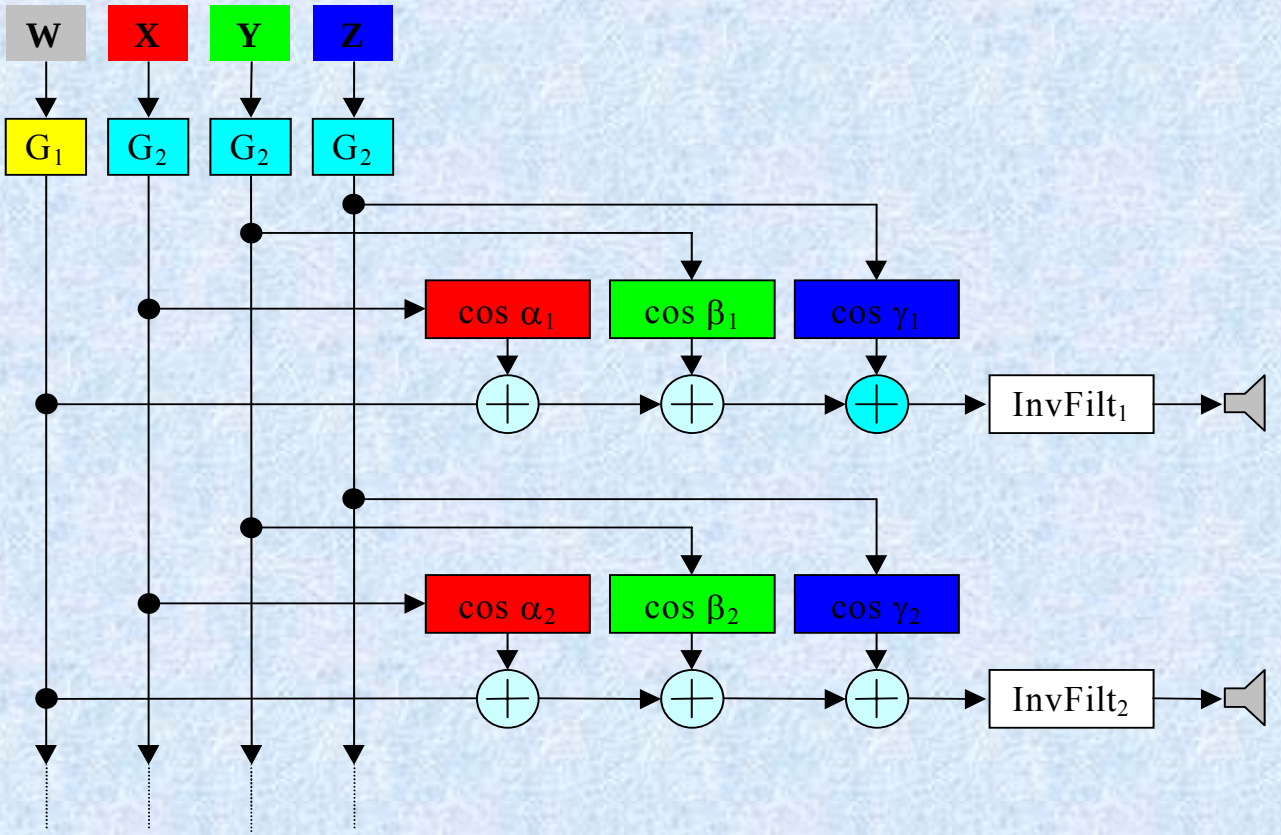


- Physically, W is a signal proportional to the pressure, XYZ are signals proportional to the three Cartesian components of the particle velocity
- when a sound wave impinges over the microphone from the “negative” direction of the x-axis, the signal on the X output will have polarity reversed with respect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

Ambisonics decoding



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

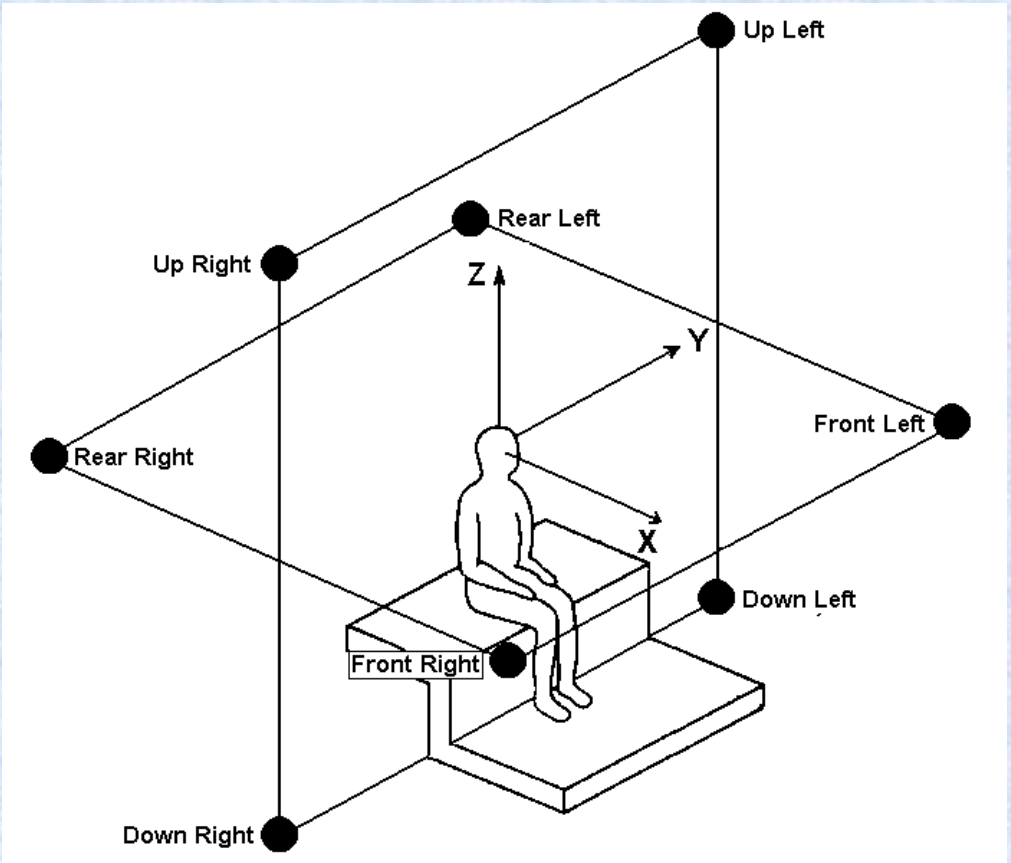


Each speaker feed is simply a weighted sum of the 4 B-format signals.

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

Ambisonics reproduction system

Bi-square Ambisonics array



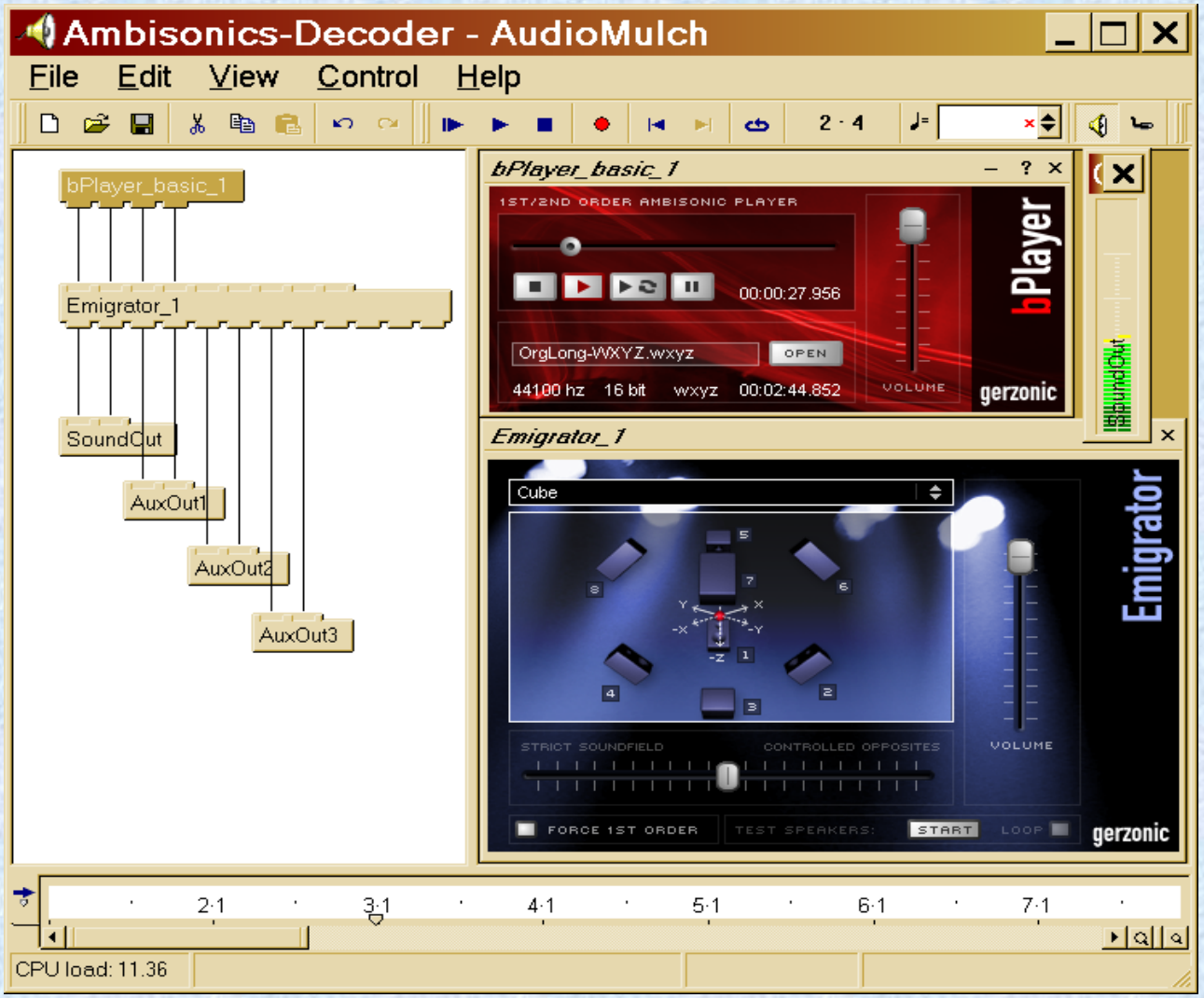
advantages:

- Three-dimensional
- Good lateral perception
- Good bass response
- Wide sweet spot, no colouring outside it

disadvantages:

- Not isotropic
- Requires advanced decoding (Y treated differently from X,Z)

A software Ambisonics decoder



Audiomulch VST host

Gerzonic bPlayer

Gerzonic Emigrator

Rooms for Ambisonics 3D 1st order

University of Parma

University of Bologna

ITU 5.1 surround

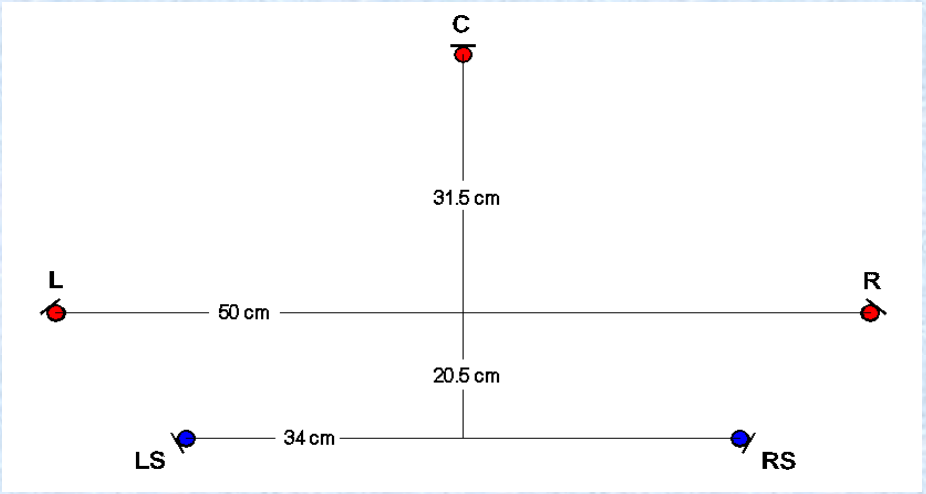
Williams MMA

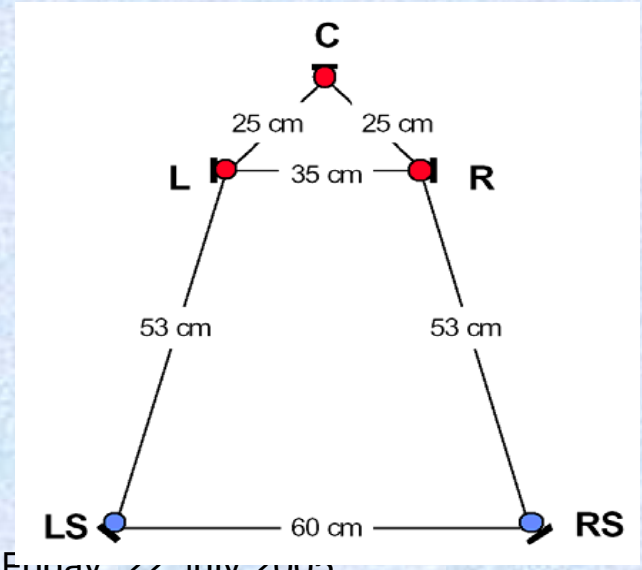
Schematic of the setup

C : Cardioid, 0°

L, R : Cardioid, ± 40°

LS, RS : Cardioid, ± 120°





INA-5

Schematic of the setup

C : Cardioid, 0°

L, R : Cardioid, ± 90°

LS, RS : Cardioid, ± 150°

ITU 5.1 surround

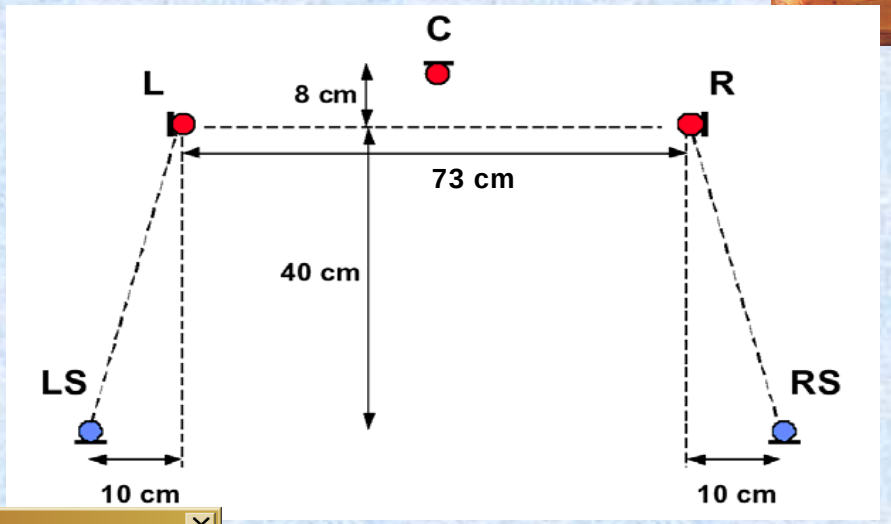
OCT

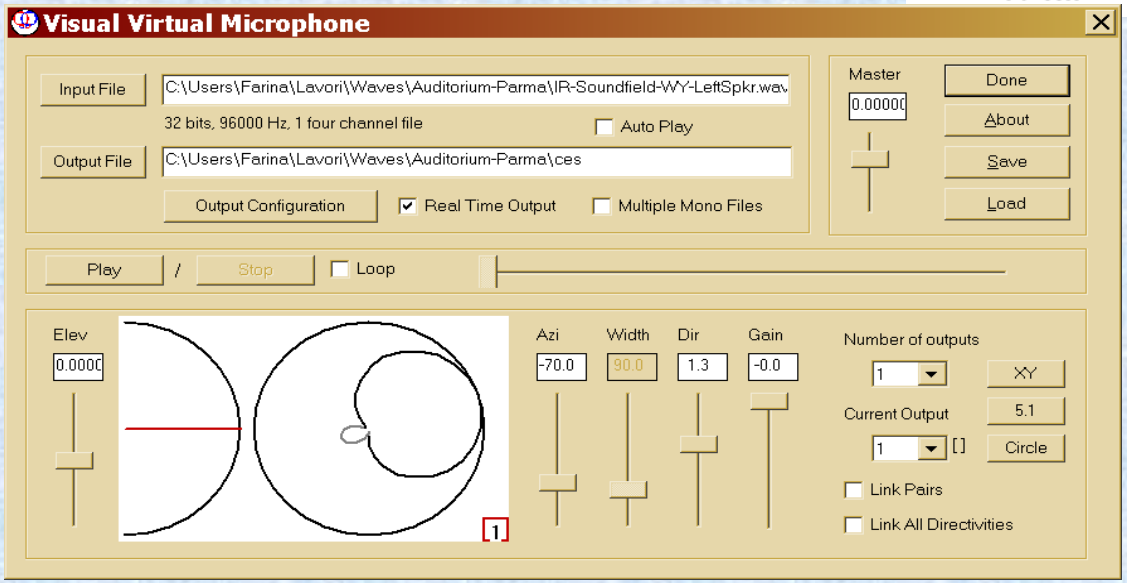
Schematic of the setup

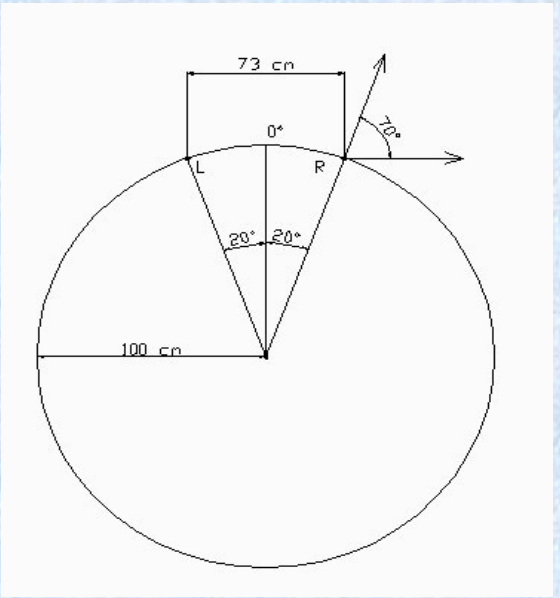
C : Cardioid, 0°

L, R : Super Cardioid, ± 90°

LS, RS : Cardioid, ± 180°

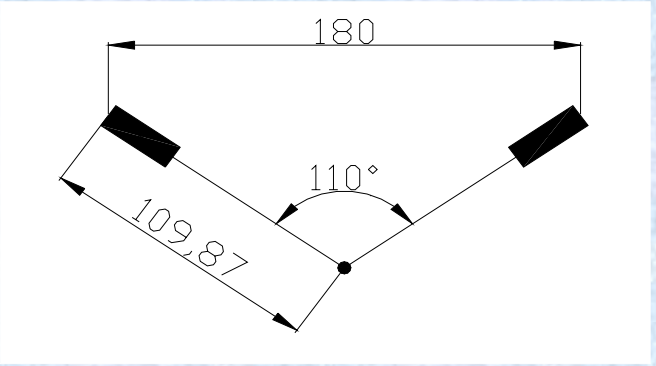




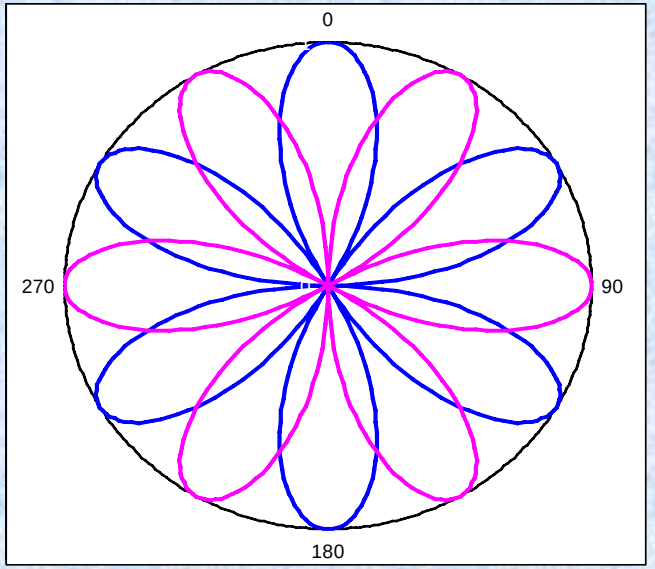


Virtual high-order microphones (M. Poletti)

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

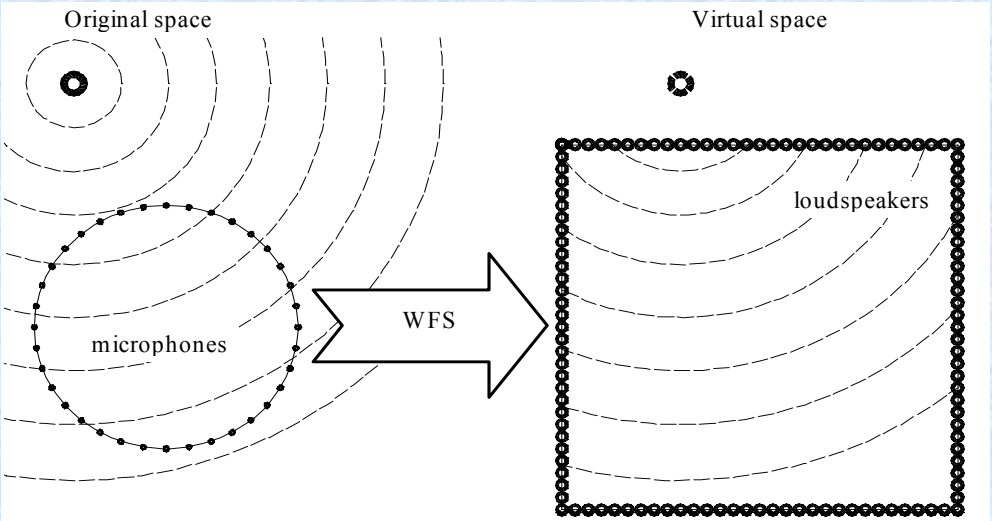


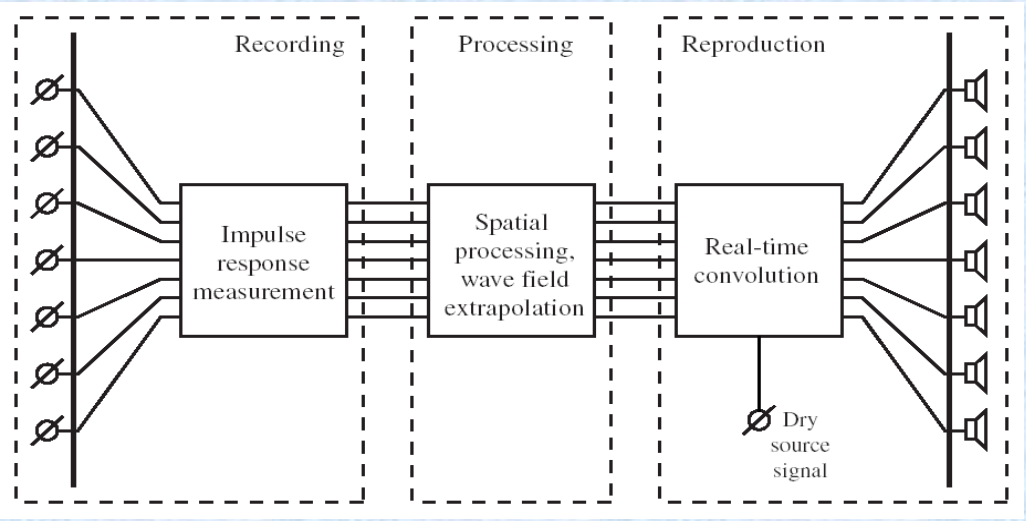
From these 36 impulse responses it is possible to derive the response of cylindrical harmonics microphones (2D Ambisonics) up to 5th order.



Wave Field Synthesis (WFS)

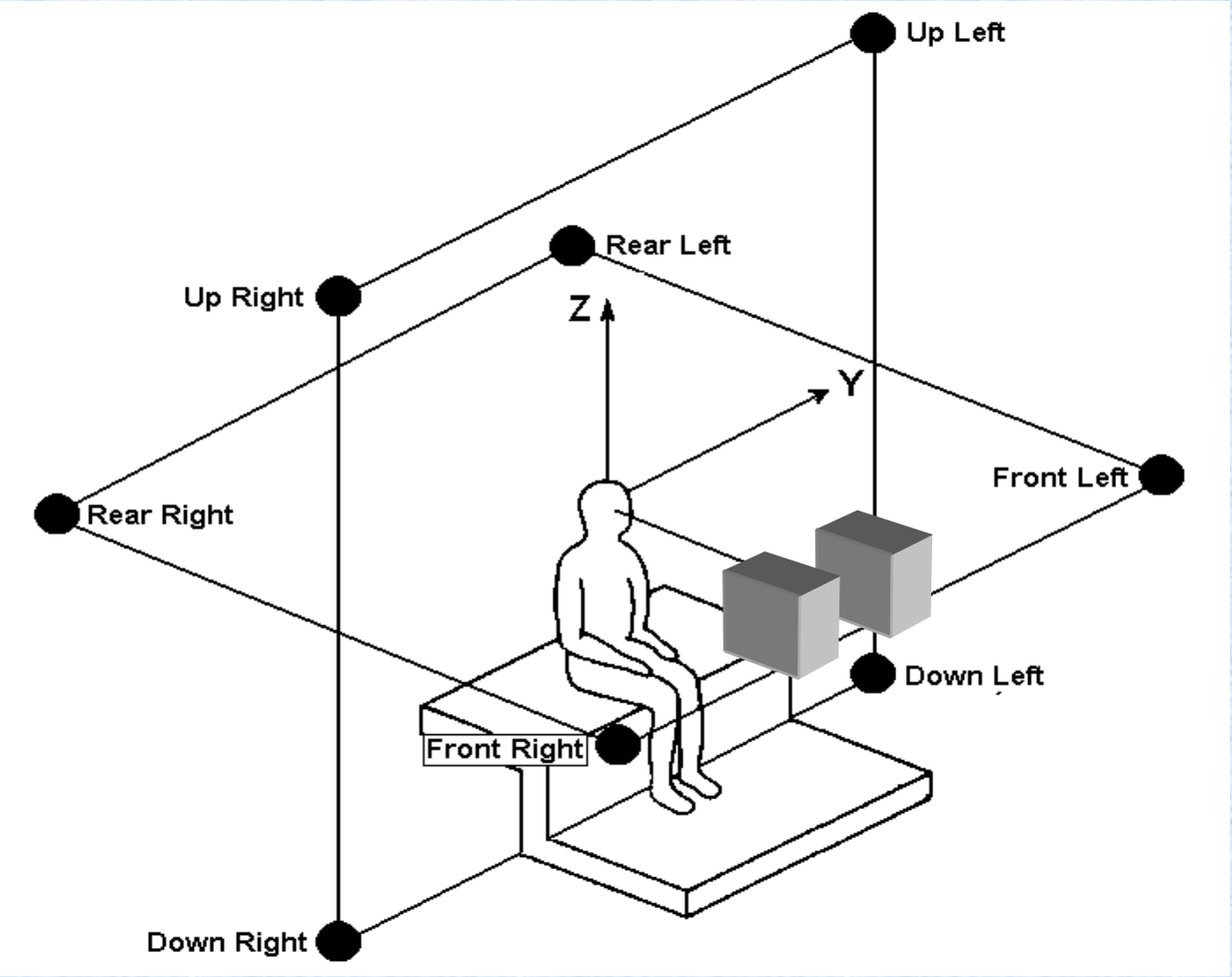
Flow diagram of the process





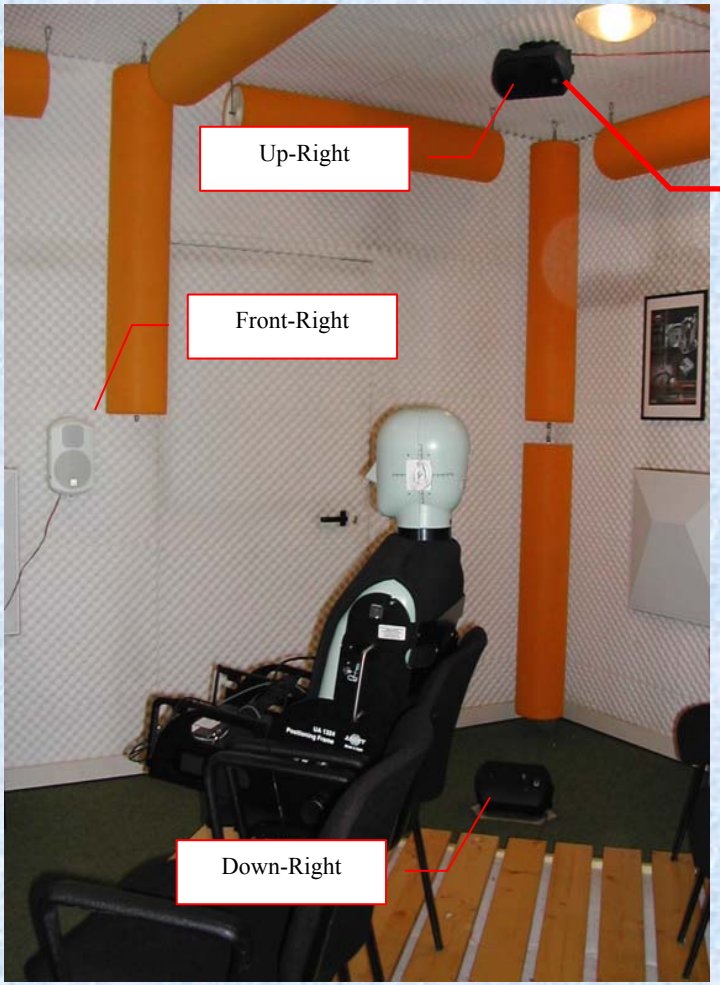
Hybrid methods (Ambiophonics)

Stereo Dipole + Virtual Ambisonics



Listening room for Ambiophonics playback

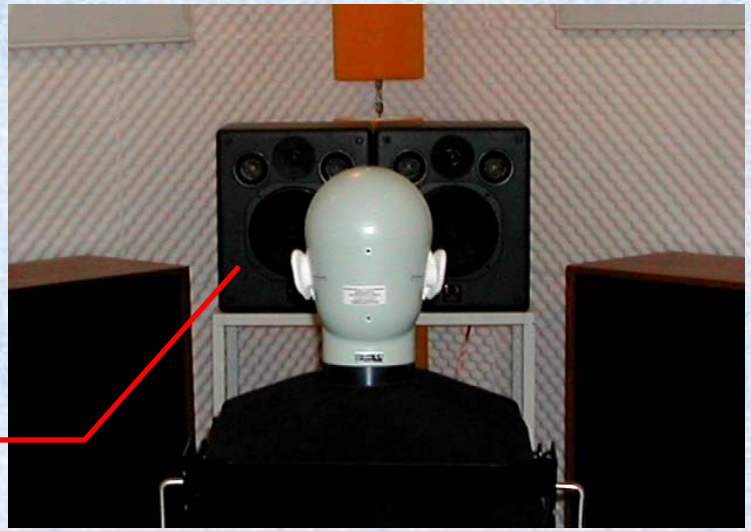
ASK room at Parma University

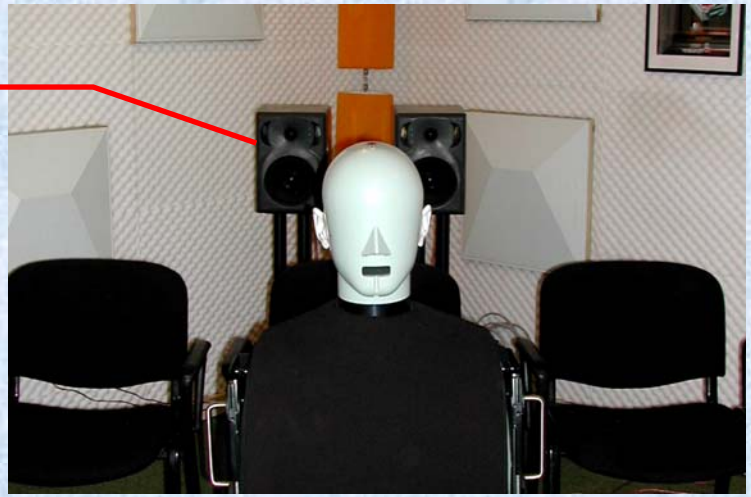


Ambisonics Dual-square array

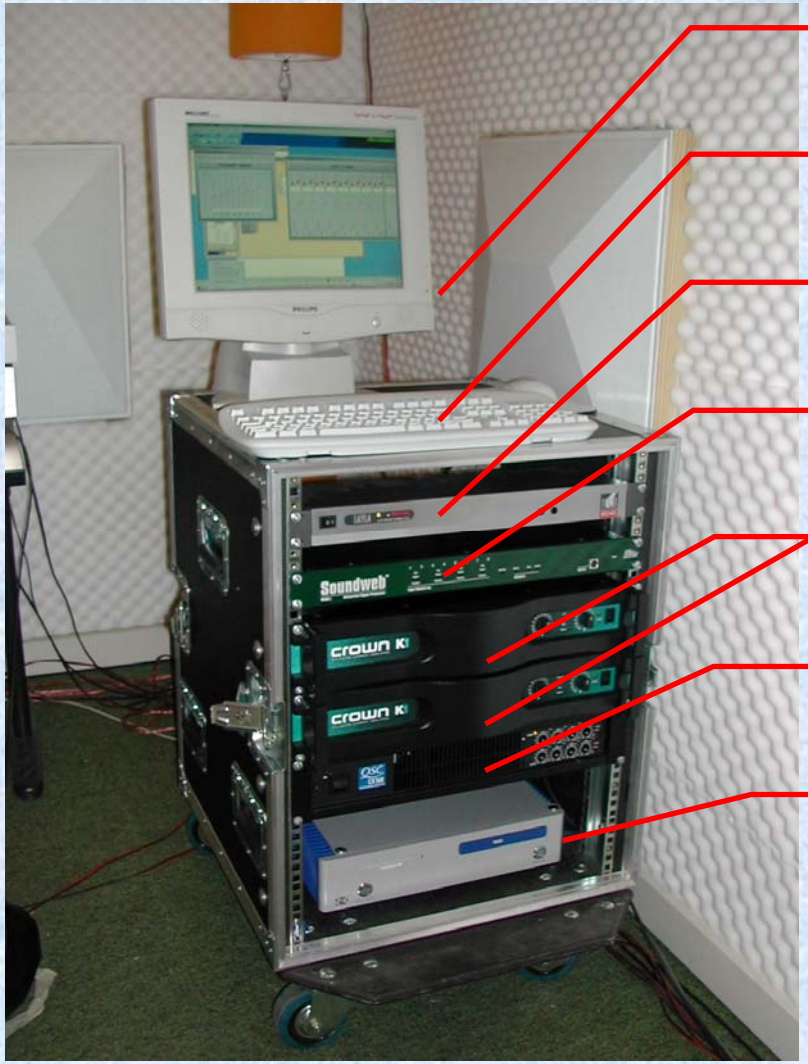
Frontal Stereo Dipole

Rear Stereo Dipole





Noiseless playback system setup



- Philips 15” Brilliance LCD display
- Logitech wireless keyboard & mouse
- Echo Layla Soundboard (8 ins, 10 outs)
- BSS Soundweb digital processor
- 2 Crown K1 amplifiers
- QSC CX168 8-channels amplifier
- Signum Data Futureclient fanless PC

Software for automatic collection of questionnaires

Risposte soggettive

Brano n. 1 2 3 4 | A B C | ▶ || □

File n. 1

Domanda 1	Too much reverberant	○	●	○	○	○	Too much dry
Domanda 2	Too soft	○	●	○	○	○	Too hard
Domanda 3	Treble too weak	○	○	●	○	○	Treble too loud
Domanda 4	Medium too weak	●	○	○	○	○	Medium too loud
Domanda 5	Bass too weak	○	●	○	○	○	Bass too loud
Domanda 6	Poor enveloping	○	●	○	○	○	Good enveloping
Domanda 7	Artificial	○	○	●	○	○	Natural
Domanda 8	Distorted	○	○	●	○	○	Undistorted
Domanda 9	Unpleasant	○	○	○	●	○	Pleasant

Precedente | Successivo | Fine

4 anechoic (dry) music pieces

3 rooms (A,B,C)

The listener is allowed to switch between theatres at will

The listener fills up the questionnaire while listening

Conclusions

- The experimental setup allows for measurements of high-quality mono, binaural and B-format IRs
- A proper listening room is required in order to reproduce sound field with Stereo Dipole and/or Ambisonic methodology
- The sound quality of different theatres can be assessed in real-time in the listening room
- Questionnaires can be collected through an interactive-driven software

Current and future developments

- Thanks to novel zero-latency convolution software, musicians can play a keyboard and listen to a virtual spatial sound environment in real time.
- Spatial auralisation can be imediately switched or morphed while the musician plays the keyboard.
- This technology has been made avaliable also for processing music in recording studios, thanks to plugins developed by Sony, Waves, Voxengo, Tascam, Altiverb
- The next step will be to port this “sampled reverberation” method also for live applications