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on Musical Acoustics
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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, m³/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

**Electrical system**
- Audio electrical Signal \( V(t) \)
- \( i(t) \)
- \( R_e \)

**Mechanical system**
- \( C_m \)
- \( M_m \)
- \( v(t) \)
- \( R_m \)
- \( Bl:1 \)
- \( 1:S_d \)

**Acoustical system**
- \( Q(t) \)
- \( Z_a \)
- Acoustic Pressure signal \( p(t) \)
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.
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).
PCM Converters

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

<table>
<thead>
<tr>
<th>Applications</th>
<th>Resolutions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CDs</td>
<td>44100 Hz</td>
</tr>
<tr>
<td>DAT, DVD Video</td>
<td>48000 Hz</td>
</tr>
<tr>
<td>DVD audio HD rec.</td>
<td>96000 Hz</td>
</tr>
<tr>
<td>Special soundcards</td>
<td>192000 Hz</td>
</tr>
</tbody>
</table>
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
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)
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 than 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)
The FFT contains magnitude and phase. The phase is meaningless for single-channel, but very important during comparison between two channels (e.g.: binaural recordings).
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.).
STFT (short term Fourier transform) for analyzing aperiodical signals

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

Digital Signal Processing 5
Sound propagation in rooms

Point Source

Direct Sound

Reflected Sound

Receiver

Direct Sound

Reflected Sound
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 length of the period.

\[ L = 2^{N-1} \]
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 $m(i)$

$$\tilde{M}(i, j) = m[(i + j - 2) \mod L] - 1$$
MLS example

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

Generate Multiple MLS Sig...

- MLS Order: 15 B
- Amplitude: 1E364
- N. sequences: 15
- Repetitions: 1
- Generate control pulse on right channel
- Control Pulse Event
- At the beginning of each repetition
- At the beginning of odd repetition but first
- At the end of each repetition

User: Andreas Lengholf
Pass key: 

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MLS example
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 \( f_2 \).

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

![Graph showing the sound power level in dB across different frequencies for unequalized and equalized conditions. The graph includes points labeled with Lw,tot = 94.8 dB and Lw,tot = 106.9 dB, indicating the sound power level at total frequencies.]
Transducers (sound source #2)

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

- 3 types of microphones:
  - 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 plays back 36 times the test signal, and simultaneously records the 8 microphonic channels.
Theatres measured

Reverberation Time T20

<table>
<thead>
<tr>
<th>Theatre</th>
<th>Sources/Receivers</th>
</tr>
</thead>
<tbody>
<tr>
<td>Uhara Hall, Kobe, Japan</td>
<td>2/2</td>
</tr>
<tr>
<td>Noh Drama Theater, Kobe, Japan</td>
<td>2/2</td>
</tr>
<tr>
<td>Kirishima Concert Hall, Kirishima, Japan</td>
<td>3/3</td>
</tr>
<tr>
<td>Greek Theater in Siracusa, Italy</td>
<td>2/1</td>
</tr>
<tr>
<td>Greek-Roman Theater in Taormina, Italy</td>
<td>3/2</td>
</tr>
<tr>
<td>Auditorium of Parma, Italy</td>
<td>3/3</td>
</tr>
<tr>
<td>Auditorium of Rome (Sala 700), Italy</td>
<td>3/2</td>
</tr>
<tr>
<td>Auditorium of Rome (Sala 1200), Italy</td>
<td>3/3</td>
</tr>
<tr>
<td>Auditorium of Rome (Sala 2700), Italy</td>
<td>3/5</td>
</tr>
<tr>
<td>Bergamo Cathedral, Italy</td>
<td>2/1</td>
</tr>
<tr>
<td>Teatro Valli, Reggio Emilia, Italy</td>
<td>5/1</td>
</tr>
<tr>
<td>Sydney Opera House, Opera Theatre</td>
<td>4/2</td>
</tr>
<tr>
<td>Sydney Opera House, Concert Hall</td>
<td>3/3</td>
</tr>
<tr>
<td>Sydney Opera House, The Studio</td>
<td>3/1</td>
</tr>
<tr>
<td>Teatro Regio, Parma, Italy</td>
<td>6/1</td>
</tr>
</tbody>
</table>

Frequency (Hz) | T20 (s)
--- | ---
31.5 | 16000
63  | 8000
125 | 4000
250 | 2000
500  | 1000
1000 | 500
2000 | 250
4000 | 125
8000 | 63
16000 | 31.5
Theaters measured (Waves, 2003)
Acoustical Parameters (ISO 3382)

- **Reverberation Time** $T_{20}$:

- **Clarity** $C_{80}$:

- **Definition** $D$:

- **Center Time** $T_S$:
Acoustical Parameters (ISO 3382)

- **Strength:**
  \[ G = \text{SPL} - L_w + 31 \quad \text{dB} \]
  \[ \rho(\tau) = \frac{\int_{-\infty}^{\infty} h_d(\tau) \cdot h_s(\tau+t) \cdot \text{d}\tau}{\sqrt{\int_{-\infty}^{\infty} h_d^2(\tau) \cdot \text{d}\tau \cdot \int_{-\infty}^{\infty} h_s^2(\tau+t) \cdot \text{d}\tau}} \]

- **IACC:**
  \[ LF = \frac{\int_{0ms}^{5ms} h_y^2(\tau) \cdot \text{d}\tau}{\int_{80ms}^{80ms} h_y^2(\tau) \cdot \text{d}\tau} \]
  \[ \text{LFC} = \frac{\int_{0ms}^{5ms} h_y(\tau) \cdot h_w(\tau) \cdot \text{d}\tau}{\int_{80ms}^{80ms} h_w^2(\tau) \cdot \text{d}\tau} \]

- **LF:**

- **LFC:**

---

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Analysis of spatial attributes

Acoustical Parameters v. ...

User Defined Reverberation Time Extremes:
(-5.0, -15.0 dB)

Enable Noise Correction

EDT without linear regression

First Arrival Time Threshold (% of FS): 25

Peak SPL value corresponding to FS: 120.0

Stereo Mode

- 2 Omnidirectional Microphones
- Soundfield Microphone (WY)
- 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: ********

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

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Polar diagrams of IACC and (1-LF)

<table>
<thead>
<tr>
<th>Auditorium</th>
<th>1-LF</th>
<th>IACC</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parma</td>
<td>0.725</td>
<td>0.266</td>
</tr>
<tr>
<td>Roma</td>
<td>0.676</td>
<td>0.344</td>
</tr>
</tbody>
</table>
Auralization is the process for making audible a virtual acoustical environment, obtained by convolving anechoic signal with the room’s impulse response.

![Diagram of auralization process]

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

\[
\begin{align*}
&y := 0; \\
&\text{FOR } n := 0 \text{ TO } N-1 \text{ DO} \\
&\quad y := y + a[n] \cdot x[n];
\end{align*}
\]

On a DSP board this instruction is performed in one cycle

- Clock core = 100 MHz
- Sample frequency \( f_S = 48 \text{ KHz} \)

Upper limit is 2000 MAC per sample
Filtering in the frequency domain:

Could be better operated in the frequency domain

\[ x(n) \xrightarrow{\text{FFT}} X(k) \]

\[ x(n) \otimes h(n) \xrightarrow{} X(k) \cdot H(k) \]

\[ y(n) \xleftarrow{\text{IFFT}} Y(k) \]

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

Solution
- Overlap & Save algorithm.
Overlap & Save algorithm:

Overlap & Save convolution process:

$$x_m(n) \rightarrow \text{FFT} \quad \text{N-point} \quad \text{IFFT} \quad \text{Xm(k)H(k)} \quad \text{Select last N – Q + 1 samples}$$

$$h(n) \rightarrow \text{FFT} \quad \text{N-point} \quad \times$$

Problems

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

Solution

- Uniformly-partitioned Overlap & Save algorithm.
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):

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).

Each block is treated as a separate filter zero-padded with zeros. For every \( i \) 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 results of the multiplications of the \( P \) filters \( S \) with the FFTs of the latest \( P \) input blocks are summed in \( P \) frequency-domain accumulators, and at the end an IFFT is done on the content of first accumulator for producing a block of output data. Only the latest \( L-K \) points of the block have to be 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 1\textsuperscript{st} order (decoding the B-format signal)
- ITU 5.1 (from different 5-mikes layouts)
- 2D Ambisonics 3\textsuperscript{rd} 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
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 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

Signal source (CD)  Filter  Amplifier  Loudspeaker  Microphone

Block diagram

Input signal  Filtering coefficients  System’s Impulse Response (Transfer function)  Output signal
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 millisecond, with no other evident alteration.
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:
  \[ F(\omega) = \frac{\text{Conj}[H(\omega)]}{\text{Conj}[H(\omega)] \cdot H(\omega) + \epsilon(\omega)} \]

- **Step 3** – go back to time domain through an IFFT:
  \[ f(\tau) = \text{IFFT}[F(\omega)] \]

Parametro di regolarizzazione
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

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Inverse filter example

Not filtered system

\[ x(\tau) \rightarrow h(\tau) \rightarrow y(\tau) \]

System

System with filter

\[ x(\tau) \rightarrow f(\tau) \rightarrow z(\tau) \rightarrow h(\tau) \rightarrow y(\tau) \]

Input signal

Filtering coefficients

Filtered signal

Output signal

Input signal

Filtering coefficients

Filtered signal

Output signal

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Binaural (Stereo Dipole)

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

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Binaural (Stereo Dipole#2)
Binaural (Stereo Dipole#3)

\[
\begin{align*}
 f_{ll} &= (h_{rr}) \otimes \text{InvDen} \\
 f_{lr} &= (-h_{lr}) \otimes \text{InvDen} \\
 f_{rl} &= (-h_{rl}) \otimes \text{InvDen} \\
 f_{rr} &= (h_{ll}) \otimes \text{InvDen} \\
 \text{InvDen} &= \text{InvFilter} \left( h_{ll} \otimes h_{rr} - h_{lr} \otimes h_{rl} \right)
\end{align*}
\]

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

\[
\text{InvDen} (\omega) = \frac{\text{Conj} \left[ C(\omega) \right]}{\text{Conj} \left[ C(\omega) \right] \cdot C(\omega) + \varepsilon(\omega)}
\]
Binaural (Dual Stereo Dipole)

**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

Rear

Quested 2108 monitors

Quested F11P monitors

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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.
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.

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

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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

Original space → WFS → Virtual space

Recording
Impulse response measurement
Spatial processing, wave field extrapolation
Real-time convolution
Dry source signal
Hybrid methods (Ambiophonics)

Stereo Dipole + Virtual Ambisonics
Listening room for Ambiophonics playback

ASK room at Parma University

- Frontal Stereo Dipole
- Ambisonics Dual-square array
- Rear Stereo Dipole
- Front-Right
- Up-Right
- Down-Right

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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

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 immediately switched or morphed while the musician plays the keyboard.
- This technology has been made available 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.