HISTORY OF ROOM IMPULSE RESPONSE MEASUREMENTS

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Time Line

The Past

- Traditional time-domain measurements with pulsive sounds and omnidirectional transducers
- Electroacoustical measurements employing special computer-based hardware, a loudspeaker and an omnidirectional microphone

The Present

- Electroacoustical measurements employing standard sound cards, 2 or more loudspeakers and multiple microphones (2 to 8)

The Future

- Microphone arrays for capturing high-order spatial information
- Artificial sound sources employing a dense array of loudspeakers, capable of synthesizing the directivity pattern of any real-world source
Basic sound propagation scheme

![Diagram of sound propagation](image)

- Direct Sound
- Reflected Sound
- Point Source
- Omnidirectional receiver

![Graph of sound propagation](image)

- Direct Sound
- Reverberant tail

Time (seconds) vs. dB
The Past
Traditional measurement methods

- Pulsive sources: ballons, blank pistol
Example of a pulsive impulse response
Test with binaural microphones

- Cheap electret mikes in the ear ducts
A loudspeaker is fed with a special test signal $x(t)$, while a microphone records the room response.

A proper deconvolution technique is required for retrieving the impulse response $h(t)$ from the recorded signal $y(t)$. 
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 non-linear part $K$ and of the background noise $n(t)$. 
Electroacoustical methods

- Different types of test signals have been developed, providing good immunity to background noise and easy deconvolution of the impulse response:
  - MLS (Maximum Length Sequence, pseudo-random white noise)
  - TDS (Time Delay Spectrometry, which basically is simply a linear sine sweep, also known in Japan as “stretched pulse” and in Europe as “chirp”)
  - ESS (Exponential Sine Sweep)

- Each of these test signals can be employed with different deconvolution techniques, resulting in a number of “different” measurement methods

- Due to theoretical and practical considerations, the preference is nowadays generally oriented for the usage of ESS with not-circular deconvolution
The first MLS apparatus - MLSSA

- MLSSA was the first apparatus for measuring impulse responses with MLS
More recently - the CLIO system

- The Italian-made CLIO system has superseded MLSSA for most electroacoustics applications (measurement of loudspeakers, quality control)
The first TDS apparatus - TEF

- Techron TEF 10 was the first apparatus for measuring impulse responses with TDS
- Subsequent versions (TEF 20, TEF 25) also support MLS
The Present
Today’s Hardware: PC and audio interface

Edirol FA-101
Firewire sound card:
10 in / 10 out
24 bit, 192 kHz
ASIO and WDM
Hardware: loudspeaker & microphone

Dodechaedron loudspeaker

Soundfield microphone
The first ESS system - AURORA

Aurora Plugins

- Generate MLS
- Deconvolve MLS
- Generate Sweep
- Deconvolve Sweep
- Convolution
- Kirkeby Inverse Filter
- Speech Transm. Index

- Aurora was the first measurement system based on standard sound cards and employing the Exponential Sine Sweep method
- It also works with traditional TDS and MLS methods, so the comparison can be made employing exactly the same hardware
**MLS method**

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

$$L = 2^N - 1$$
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 $m(i)$:

$$\tilde{M}(i, j) = m[(i + j - 2) \mod L] - 1$$
MLS example
Example of a MLS impulse response
Exponential Sine Sweep method

- $x(t)$ is a band-limited sinusoidal sweep 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 – $x(t)$
The not-linear behaviour of the loudspeaker causes many harmonics to appear.
The deconvolution of the IR is obtained convolving the measured signal $y(t)$ with the inverse filter $z(t)$ [equalized, time-reversed $x(t)$].
Deconvolution of Exponential 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 $z(t)$
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. Exp. Sine Sweep

Aurora - Logarithmic Sine Sweep

Time (seconds)
Example of an ESS impulse response
Spatial analysis by directive microphones

- The initial approach was to use directive microphones for gathering some information about the spatial properties of the sound field “as perceived by the listener”
- Two apparently different approaches emerged: binaural dummy heads and pressure-velocity microphones:

  - Binaural microphone (left)
  - and
  - variable-directivity microphone (right)
IACC “objective” spatial parameter

- It was attempted to “quantify” the “spatiality” of a room by means of “objective” parameters, based on 2-channels impulse responses measured with directive microphones.
- The most famous “spatial” parameter is IACC (Inter Aural Cross Correlation), based on binaural IR measurements.

\[ \rho(t) = \frac{\int_{0}^{80 \text{ms}} p_L(\tau) \cdot p_R(\tau + t) \cdot d\tau}{\sqrt{\int_{0}^{80 \text{ms}} p_L^2(\tau) \cdot d\tau \cdot \int_{0}^{80 \text{ms}} p_R^2(\tau + t) \cdot d\tau}} \]

\[ \text{IACC}_E = \text{Max}[\rho(t)] \quad t \in [-1 \text{ms}...+1 \text{ms}] \]
LF “objective” spatial parameter

- Other “spatial” parameters are the Lateral Energy ratio LF
- This is defined from a 2-channels impulse response, the first channel is a standard omni microphone, the second channel is a “figure-of-eight” microphone:

\[ \text{LF} = \frac{\int_{5\text{ms}}^{80\text{ms}} h_8^2(\tau) \cdot d\tau}{\int_{0\text{ms}}^{80\text{ms}} h_o^2(\tau) \cdot d\tau} \]
Are binaural measurements reproducible?

- Experiment performed in anechoic room - same loudspeaker, same source and receiver positions, 5 binaural dummy heads
Are binaural measurements reproducible?

- Diffuse field - huge difference among the 4 dummy heads

![Graph](image)
Are LF measurements reproducible?

- Experiment performed in the Auditorium of Parma - same loudspeaker, same source and receiver positions, 4 pressure-velocity microphones
Are LF measurements reproducible?

- At 25 m distance, the scatter is really big
3D Impulse Response (Gerzon, 1975)

- **Original Room**
  - Sound Source
  - SoundField Microphone

- **Measurement of B-format Impulse Responses**
  - B-format 4-channels signal (WXYZ)

- **Portable PC with 4-channels sound board**
  - MLS or sweep excitation signal

- **B-format Imp. Resp. of the original room**

- **B-format 4-channels signal (WXYZ)**

- **Convolver**
  - Convolution of dry signals with the B-format Impulse Responses

- **Ambisonics decoder**

- **Speaker array in the reproduction room**
3D extension of the pressure-velocity measurements

- The Soundfield microphone allows for simultaneous measurements of the omnidirectional pressure and of the three cartesian components of particle velocity (figure-of-8 patterns)
The Waves project (2003)

- The original idea of Michael Gerzon was finally put in practice in 2003, thanks to the Israeli-based company WAVES.
- More than 50 theatres all around the world were measured, capturing 3D IRs (4-channels B-format with a Soundfield microphone).
- The measurements did also include binaural impulse responses, and a circular-array of microphone positions.
- More details on WWW.ACOUSTICS.NET.
The Future
The Future

- Microphone arrays capable of synthesizing arbitrary directivity patterns
- Advanced spatial analysis of the sound field employing spherical harmonics (Ambisonics - 1° order or higher)
- Loudspeaker arrays capable of synthesizing arbitrary directivity patterns
- Generalized solution in which both the directivities of the source and of the receiver are represented as a spherical harmonics expansion
Directivity of transducers

Soundfield ST-250 microphone

125 Hz

2000 Hz
How to get better spatial resolution?

- The answer is simple: analyze the spatial distribution of both source and receiver by means of higher-order spherical harmonics expansion.
- Spherical harmonics analysis is the equivalent, in space domain, of the Fourier analysis in time domain.
- As a complex time-domain waveform can be though as the sum of a number of sinusoidal and cosinusoidal functions, so a complex spatial distribution around a given notional point can be expressed as the sum of a number of spherical harmonic functions.
Higher-order spherical harmonics expansion

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3°-order microphone (Trinnov - France)

- Arnoud Laborie developed a 24-capsule compact microphone array - by means of advanced digital filtering, spherical ahrmonic signals up to 3° order are obtained (16 channels)
4°-order microphone (France Telecom)

- Jerome Daniel and Sebastien Moreau built samples of 32-capsules spherical arrays - these allow for extractions of microphone signals up to 4° order (25 channels)
Verification of high-order patterns

- Sebastien Moreau and Olivier Warusfel verified the directivity patterns of the 4°-order microphone array in the anechoic room of IRCAM (Paris)
What about source directivity?

- Current 3D IR sampling is still based on the usage of an “omnidirectional” source.
- The knowledge of the 3D IR measured in this way provides no information about the soundfield generated inside the room from a directive source (i.e., a musical instrument, a singer, etc.).
- Dave Malham suggested to represent also the source directivity with a set of spherical harmonics, called O-format - this is perfectly reciprocal to the representation of the microphone directivity with the B-format signals (Soundfield microphone).
- Consequently, a complete and reciprocal spatial transfer function can be defined, employing a 4-channels O-format source and a 4-channels B-format receiver:

\[ \text{O-format 4-channels source} \rightarrow \text{B-format 4-channels microphone (Soundfield)} \rightarrow \text{Portable PC with 4in-4out channels sound board} \]
Directivity of transducers

LookLine D200 dodecahedron

- **250 Hz**
- **1000 Hz**
- **2000 Hz**
- **4000 Hz**
- **8000 Hz**
- **16000 Hz**

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High-order sound source

- Adrian Freed, Peter Kassakian, and David Wessel (CNMAT) developed a new 120-loudspeakers, digitally controlled sound source, capable of synthesizing sound emission according to spherical harmonics patterns up to 5° order.
Technical details of high-order source

- Class-D embedded amplifiers
- Embedded ethernet interface and DSP processing
- Long-excursion special Meyer Sound drivers
Accuracy of spatial synthesis

- The spatial reconstruction error of a 120-loudspeakers array is frequency dependant, as shown here:

- The error is acceptably low over an extended frequency range up to 5°-order
Complete high-order MIMO method

- Employing massive arrays of transducers, it will be feasible to sample the acoustical temporal-spatial transfer function of a room.
- Currently available hardware and software tools make this practical only up to 4° order, which means 25 inputs and 25 outputs.
- A complete measurement for a given source-receiver position pair takes approximately 10 minutes (25 sine sweeps of 15s each are generated one after the other, while all the microphone signals are sampled simultaneously).
- However, it has been seen that real-world sources can be already approximated quite well with 2°-order functions, and even the human HRTF directivities are reasonably approximated with 3°-order functions.
Conclusions

- The sine sweep method revealed to be systematically superior to the MLS method for measuring electroacoustical impulse responses.
- Traditional methods for measuring “spatial parameters” (IACC, LF) proved to be unreliable and do not provide complete information.
- The 1°-order Ambisonics method can be used for generating and recording sound with a limited amount of spatial information.
- For obtaining better spatial resolution, High-Order Ambisonics can be used, limiting the spherical-harmonics expansion to a reasonable order (2°, 3° or 4°).
- Experimental hardware and software tools have been developed (mainly in France, but also in USA), allowing to build an inexpensive complete measurement system.