IMPULSE RESPONSE MEASUREMENTS

Angelo Farina
Industrial Engineering Dept.
University of Parma - ITALY
HTTP://www.angelofarina.it
Time Line

The Past

- Traditional time-domain measurements with pulsed 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

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

Graph showing sound propagation with dB level over time.
The Past
Traditional measurement methods

- Pulsive sources: ballons, blank pistol
Example of a pulsive impulse response

Note the bell-shaped spectrum
Test with binaural microphones

- Cheap electret mikes in the ear ducts
Loudspeaker as sound source

- 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 (1998)
- 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.

\[
L = 2^N - 1
\]
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.

\[ h = \frac{1}{L + 1} \cdot M \cdot y \]

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 measurement procedure

Deconvolve Multiple MLS Sequ...
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.
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 \( x(t) \)]
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.
Result of the deconvolution

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

- After the sequence of impulse responses has been obtained, it is possible to select and extract just one of them (the 1°-order - Linear in this example):
Example of an ESS impulse response

Octave-Band Spectrum

Acoustical Parameters according to ISO3382 (v. 4.2)

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Maximum Length Sequence vs. Exp. Sine Sweep

Aurora - Logarithmic Sine Sweep
Post processing of impulse responses

- A special plugin has been developed for the computation of STI according to IEC-EN 60268-16:2003

**STI & Octave Band Analysis**

- Calibration (Octave Analysis):
  - Full Scale
  - Calibration value (dB): 120.00
- Compute Octave Band Spectrum
- Load SPL Values from File...
- Save SPL Values to File...

**Octave-Band Spectrum**

- Frequency bands: 31.5 Hz, 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz, 16 kHz
- Level values for each band

**STI (according to IEC-EN 60268-16:2003)**

- STI values for different frequency bands
- STI Male, STI Female, flat-STI, STI2, STI+1

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Post processing of impulse responses

- A special plugin has been developed for performing analysis of acoustical parameters according to ISO-3382
The new AQT plugin for Audition

- The new module is still under development and will allow for very fast computation of the AQT (Dynamic Frequency Response) curve from within Adobe Audition.
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.

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

\[
\text{IACC}_E = \text{Max}[\rho(t)] \quad t \in [-1\text{ms}...+1\text{ms}]
\]
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:

\[
LF = \frac{\int_{0}^{5\text{ms}} h_{\text{o}}^2(\tau) \cdot d\tau}{\int_{0}^{80\text{ms}} h_{\text{o}}^2(\tau) \cdot d\tau}
\]

\[
h_{\text{o}}(\tau) \quad \text{omni}
\]

\[
h_{\text{8}}(\tau) \quad \text{figure-of-eight}
\]
Are IACC measurements reproducible?

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

- Diffuse field - huge difference among the 4 dummy heads
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

![Comparison LF - measure 2 - 25m distance graph](image-url)
3D Impulse Response (Gerzon, 1975)

Original Room
Sound Source
SoundField Microphone

Measurement of B-format Impulse Responses
Portable PC with 4-channels sound board
B-format Imp. Resp. of the original room

B-format 4-channels signal (WXYZ)
MLS or sweep excitation signal

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

B-format 4-channels signal (WXYZ)
Ambisonics decoder

Sound Source
Mono Mic.

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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Arnoud Laborie developed a 24-capulse 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)
4°-order microphone (Italy)

- Angelo Farina’s spherical mike (32 capsules)
5°-order microphones (University of Sydney)

- Chris Craig’s dual-sphere concentrical mike (64 capsules)
- And his 32-capsules cylindrical mike
Verification of high-order patterns

- Sebastien Moreau and Olivier Warusfel verified the directivity patterns of their 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 provide 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:
Directivity of transducers

LookLine D-300 dodechaedron

250 Hz

1000 Hz

2000 Hz

4000 Hz

8000 Hz

16000 Hz
Directivity of transducers

LookLine D-200 dodecahedron

250 Hz

1000 Hz

2000 Hz

4000 Hz

8000 Hz

16000 Hz
Directivity of transducers

Omnisonic 1000 dodecahedron

250 Hz

1000 Hz

2000 Hz

4000 Hz

8000 Hz

16000 Hz
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 directivites are reasonably approximated with 3°-order functions.
Improving the ESS method
Topics

- Pre-ringing at high and low frequency before the arrival of the direct sound pulse
- Pre/post equalization of the test signal performed in a way which avoids time-smearing of the impulse response
- Sensitivity to abrupt pulsive noises during the measurement
- Skewing of the measured impulse response when the playback and recording digital clocks are mismatched
- Cancellation of high frequencies in the late part of the tail when performing synchronous averaging
Pre-ringing at high and low frequency

- Pre-ringing at high frequency due to improper fade-out

This picture shows the preringing obtained deconvolving directly the test signal, without passing through the system under test.
Pre-ringing at high and low frequency

- Perfect Dirac’s delta after removing the fade-out

This picture shows the result obtained deconvolving directly the test signal, without passing through the system under test, and employing a sine sweep going up to the Nyquist frequency.
Pre-ringing at high and low frequency

- Pre-ringing at low frequency due to a bad sound card featuring frequency-dependent latency

This artifact can be corrected if the frequency-dependent latency remains the same, by creating a suitable inverse filter with the Kirkeby method.
Kirkeby inverse filter

- The Kirkeby inverse filter is computed inverting the measured IR

1) The IR to be inverted is FFT transformed to frequency domain:

\[ H(f) = \text{FFT}[h(f)] \]

2) The computation of the inverse filter is done in frequency domain:

\[ C(f) = \frac{\text{Conj}[H(f)]}{\text{Conj}[H(f)] \cdot H(f) + \varepsilon(f)} \]

Where \( \varepsilon(f) \) is a small, frequency-dependent regularization parameter.

3) Finally, an IFFT brings back the inverse filter to time domain:

\[ c(t) = \text{IFFT}[C(f)] \]
Pre-ringing at high and low frequency

- Convolving the time-smeared IR with the Kirkeby compacting filter, a very sharp IR is obtained.

The same method can also be applied for correcting the response of the loudspeaker/microphone system, if an anechoic preliminary test is done.
- Pre-ringing at high and low frequency before the arrival of the direct sound pulse
- Pre/post equalization of the test signal performed in a way which avoids time-smearing of the impulse response
- Sensitivity to abrupt pulsive noises during the measurement
- Skewing of the measured impulse response when the playback and recording digital clocks are mismatched
- Cancellation of high frequencies in the late part of the tail when performing synchronous averaging
Equalization of the whole system

- An anechoic measurement is first performed
Equalization of the whole system

- A suitable inverse filter is generated with the Kirkeby method by inverting the anechoic measurement
Equalization of the whole system

- The inverse filter can be either pre-convolved with the test signal or post-convolved with the result of the measurement.
- Pre-convolution usually reduces the SPL being generated by the loudspeaker, resulting in worst S/N ratio.
- On the other hand, post-convolution can make the background noise to become “coloured”, and hence more perceptible.
- The resulting anechoic IR becomes almost perfectly a Dirac’s Delta function.
Topics

- Pre-ringing at high and low frequency before the arrival of the direct sound pulse
- Pre/post equalization of the test signal performed in a way which avoids time-smearing of the impulse response
- Sensitivity to abrupt pulsive noises during the measurement
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- Cancellation of high frequencies in the late part of the tail when performing synchronous averaging
Sensitivity to abrupt pulsive noises

- Often a pulsive noise occurs during a sine sweep measurement
Sensitivity to abrupt pulsive noises

- After deconvolution, the pulsive sound causes untolerable artifacts in the impulse response.

The artifact appears as a down-sloping sweep on the impulse response. At the 2 kHz octave band the decay is distorted, and the reverb. time is artificially increased from 2.13 to 2.48 s.
Sensitivity to abrupt pulsive noises

- Several denoising techniques can be employed:
  - Brutely silencing the transient noise
  - Employing the specific “click-pop eliminator” plugin of Adobe Audition
  - Applying a narrow-passband filter around the frequency which was being generated in the moment in which the pulsive noise occurred

- The third approach provides the better results:
Topics

- Pre-ringing at high and low frequency before the arrival of the direct sound pulse
- Pre/post equalization of the test signal performed in a way which avoids time-smearing of the impulse response
- Sensitivity to abrupt pulsive noises during the measurement
- Skewing of the measured impulse response when the playback and recording digital clocks are mismatched
- Cancellation of high frequencies in the late part of the tail when performing synchronous averaging
Clock mismatch

- When the measurement is performed employing devices which exhibit significant clock mismatch between playback and recording, the resulting impulse response is “skewed” (stretched in time):

The pictures show the results of an electrical measurement performed connecting directly a CD-player with a DAT recorder.
Clock mismatch

- It is possible to re-pack the impulse response employing the already-described approach based on the usage of a Kirkeby inverse filter:

However, this is possible only if a “reference” electrical (or anechoic) measurement has been performed. But, in many cases, one only gets the re-recorded signals, and no reference measurement is available, so the Kirkeby inverse filter cannot be computed.
Clock mismatch

- However, it is always possible to generate a pre-stretched inverse filter, which is longer or shorter than the “theoretical” one - by proper selection of the length of the inverse filter, it is possible to deconvolve impulse responses which are almost perfectly “unskewed”:

The pictures show the result of the deconvolution of a clock-mismatched measurement, in which a pre-stretched inverse filter is employed, 8.5 ms longer than the theoretical one.
Topics

- Pre-ringing at high and low frequency before the arrival of the direct sound pulse
- Pre/post equalization of the test signal performed in a way which avoids time-smearing of the impulse response
- Sensitivity to abrupt pulsive noises during the measurement
- Skewing of the measured impulse response when the playback and recording digital clocks are mismatched
- Cancellation of high frequencies in the late part of the tail when performing synchronous averaging
High-frequency cancellation due to averaging

- When several impulse response measurements are synchronously-averaged for improving the S/N ratio, the late part of the tail cancels out, particularly at high frequency, due to slight time variance of the system.

Comparison of a single sweep 50 s long with the synchronous average of 50 sweeps, 1 s long each.

Spectrum of a single sweep of 50s (above) versus 50 sweeps of 1s (below) short-FFT spectrum at 200 ms after direct sound.
High-frequency cancellation due to averaging

- However, if averaging is performed properly in spectral domain, and a single conversion to time domain is performed after averaging, this artifact is significantly reduced.
- The new “cross Functions” plugin can be used for computing $H_1$:

$$H_1(f) = \frac{G_{LR}}{G_{LL}}$$
Conclusions (I of II)

- The ESS 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.
Conclusions (II of II)

- ESS is now employed in top-grade measurement systems: Audio Precision (TM), Rhode-Schwartz and B&K / DIRAC
- However, these completely-packaged measurement systems often do not allow to play “tricks” and to adjust the signals for solving problems, which have been shown here
- Workarounds have been found for the problems occurring when performing ESS measurements
- These workarounds are easily applied by working with a general purpose sound editor (Adobe Audition)
- A number of additional plugins have been developed, making easy to generate the test signal, to deconvolve and process impulse responses, to compute inverse filters and to perform advanced processing (STI, AQT, etc.)
- These plugins are freely downloadable at the AURORA web site: www.aurora-plugins.com