La misura della risposta all'impulso per la caratterizzazione di sistemi acustici e vibrazionali

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Time Line

The Past

- Traditional time-domain measurements with pulsive sounds and omnidirectional transducers
- MLS and TDS methods for electroacoustical measurements

The Present

- Electroacoustical measurements employing the Exponential Sine Sweep method (ESS)
- Capturing spatial information with multichannel microphones
- Post processing of measured IRs for computing acoustical parameters and for auralization
The Past
Starting point: room impulse response

Point Source -- Direct Sound

Point Source -- Reflected Sound

Omnidirectional receiver

Graph showing time (seconds) vs. dB with labels for Direct Sound and Reverberant tail.

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Traditional measurement methods

- Pulsive sources: ballons, blank pistol
Example of a pulsive impulse response
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 $y(t)$.
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- The Italian-made CLIO system has superseded MLSSA for most low-cost electroacoustics applications (measurement of loudspeakers, quality control)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
Theory of 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

Portable PC with 4-channels sound board

Room

Loudspeaker

Microphone

Measurement of Room Impulse Response

Output signal y

MLS test signal x

Portable PC with additional sound card

Generate Multiple MLS Sig...

| MLS Order | 15 B | OK |
| Amplitude | 16394 | Cancel |
| N sequences | 16 | |
| Repetitions | 1 | 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: ********

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MLS example

Deconvolve Multiple MLS Sequences

Input Data:
- MLS Order: 15
- N. of measurements: 1
- N. of sequences/measurement: 16
- N. of first sequences to skip: 1

Output Results:
- N. of samples for each sequence: 32767
- N. of samples to skip: 0

- Scale each response separately
- Remove DC component

User: Andreas Langhaff
Reg. key: 

[OK], [Cancel], [Help]
Example of a MLS impulse response

Octave-Band Spectrum

Acoustical Parameters according to ISO3382 (v. 4.2)

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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
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 $f_2$.

$$x(t) = \sin \left[ \frac{2 \cdot \pi \cdot f_1 \cdot T}{\ln \left( \frac{f_2}{f_1} \right)} \cdot \left( e^\frac{t \cdot \ln \left( \frac{f_2}{f_1} \right)}{T} - 1 \right) \right]$$
Test Signal – x(t)
The non-linear behaviour of the loudspeaker causes many harmonics to appear.
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.

![Test Signal \( x(t) \)](image1)

![Inverse Filter \( z(t) \)](image2)
Deconvolution = rotation of the sonograph

- Convolving with the inverse filter rotates the time-log(f) plane counter clockwise
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 insulate just one of them:
Maximum Length Sequence vs. Exp. Sine Sweep

Aurora - Logarithmic Sine Sweep
Example of an ESS impulse response
Post processing of impulse responses

- A special plugin has been developed for the computation of STI according to IEC-EN 60268-16:2003
The STI method is based on the MTF concept: a carrier signal (one-octave-band-filtered noise) is amplitude modulated at a given modulation frequency with 100% modulation depth. At the receiver, the modulation depth is reduced, due to noise, reverb, echoes, etc.
MTF from Impulse Response

- **It is possible to derive the MTF values from a single impulse response measurement:**

To compute each value of $m(F)$ from the impulse response $h(t)$, an octave-band filter is first applied to the impulse response, in order to select the carrier’s frequency band $f$. Then $m(F)$ is obtained with the formula

$$m(F) = \frac{\int_{0}^{\infty} h_f^2(\tau) \cdot \exp(-j \cdot 2 \cdot \pi \cdot F \cdot \tau) \cdot d\tau}{\int_{0}^{\infty} h_f^2(\tau) \cdot d\tau}$$
Background noise

- If the background noise is superposed to the impulse response, the previous method already takes care of it, and the MTF values are measured correctly.
- However, in some cases, it is advisable to perform a noise-free measurement of the IR, and then insert the effect of the noise with the following expression:

\[
m(F) = m'(F) \cdot \frac{1}{1 + 10^{\left(\frac{L_{noise} - L_{signal}}{10}\right)}}
\]

- This makes it possible to measure a “clean” impulse response, and then to perform separately the signal and noise recordings.
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.
Distortion measurements

A headphone was driven with a 1 V RMS signal, causing severe distortion in the small loudspeaker.

The measurement was made placing the headphone on a dummy head.

Measurements: ESS and traditional sine at 1 kHz
Distortion measurements

- Comparison between:
  - traditional distortion measurement with fixed-frequency sine (the black histogram)
  - the new exponential sweep (the 4 narrow, coloured lines)
Spatial analysis by directive impulse responses

- 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

Pressure-velocity 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.

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

\[
\text{IACC}_E = \text{Max}[\rho(t)] \quad t \in [-1\text{ms}, +1\text{ms}]
\]
Lateral Fraction (LF) spatial parameter

- Another “spatial” parameter is the Lateral Fraction 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:

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

- Experiment performed in anechoic room - same loudspeaker, same source and receiver positions, 5 binaural dummy heads
Are 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
3D Impulse Response (Gerzon, 1975)

Portable PC with 4-channels sound board

Measurement of B-format Impulse Responses

B-format 4-channels signal (WXYZ)

MLS or sweep excitation signal

Convolution of dry signals with the B-format Impulse Responses

B-format Imp. Resp. of the original room

Original Room

Sound Source

SoundField Microphone

Sound Source

Mono Mic.

Convolver

B-format 4-channels signal (WXYZ)

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)
Directivity of transducers

Soundfield ST-250 microphone

125 Hz

2000 Hz
A-format microphone arrays

Today several alternatives to Soundfield microphones do exist. All of them are providing “raw” signals from the 4 capsules, and the conversion from these signals (A-format) to the standard Ambisonic signals (B-format) is performed digitally by means of software running on the computer.
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 Ciresa project (2005)

- Measurements of the vibrations and radiated sound from wood panels
- Mapping of harmonic tables by means on an XY scanner
- Pressure measured by means of a linear microphone array
- Velocity measured by means of a laser vibrometer

Radiated sound pressure level (Left) and vibration velocity (Right)
Conclusions

- The sine sweep method revealed to be systematically superior to the MLS & TDS methods for measuring electroacoustical impulse responses.

- The ESS method also allows for measurement of non-linear devices and to assess harmonic distortion.

- Current limitation for spatial analysis in room acoustics is due to transducers (loudspeakers and microphones).

- A new generation of loudspeakers and microphones, made of massive arrays, is under development.

- The “harmonic orders” impulse responses obtained by the exponential sine sweep method can be used for non-linear convolution, which yields more realistic auralization.