September 29

VIRTUAL ROOM ANALYSIS

* Mic: PCB piezoelectronics, omnidirectional microphone
* The sweep has been sent by each loudspeaker one after the other
* The sum of the single impulse responses is being computed

GOAL

* Find T20, clarity, definition

SET UPS

Same set of the morel compensation procedure.

Low pass 40 Hz, switch frequency between woofer and tweeter: 2500 Hz

First measure: the headrest has been removed in order to avoid sound absorptions or reflections, the mic has been placed in the driver position, at the height of the ears, pointing at the ceiling.

TO BE SOLVED

* Microphone calibration: the mic has been calibrated but the post gain not applied yet because of doubts about the procedure to do it for the reason that the IR is obtained throught convolution passages. 

NEXT STEPS:

* Let all the loudspeakers play once all together to get the whole impulse response and compare with the sum of the single ones 
* Measure from different points

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October 19

Loudspeakers were simultaneously activated, and the two responses, this last one and the one obtained by summing the measurements taken on September 29th, were compared.

Is it possible to identify a similar trend in the two impulse responses, specifically noting a gap around 1600 Hz.

Discussion on the found parameters:

* T20 very low
* Clarity very high
* Definition not interesting because gives the same information given by the clarity
* Same results obtained with audition if the window is set as indicated in the matlab code, 5051 samples
* Use of the standard ISO 3382 to get the parameters
* Use of the Toolbox ITA

NEXT STEPS:

* Write details about the decay curve and the Schroeder’s integral and the extraction of the parameters 🡪 state of art 
* Add details on the ITA Toolbox procedure 

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ITA TOOLBOX

* Freq. Range: 100-5000
* 1/3 or 1 octave band
* Lunderby algorithm for noise detection
* Send to the function the signal properly cut, cutting the tail, 5001 samples

EDC:Early decay curve, or called Schroeder’s curve, obtained by calculating the reverse time integrated impulse response: Generate for each octave band the decay curve by a backward integration of the squared impulse response.

Lunderby algorithm in order to minimize the background noise, it gives a integration starting point value from which start the backward integration, as suggested by ISO 3381

The Schroeder Integral is a curve obtained by backwards integration of the squared impulse response, ideally starting from a point where the response falls into the noise and applying a correction (a starting value for the integral) which assumes the rate at which the Schroeder curve is falling continues for the whole response. It is used a procedure to estimate the best starting point for the integration, based on "Lundeby's Method" (from the paper by A. Lundeby, T. E. Vigran, H. Bietz, and M. Vorländer, “Uncertainties of Measurements in Room Acoustics,” Acustica, vol. 81, pp. 344–355 (1995))

C50 clarity factor for speech= 10log(D50/(1-D50)) dB

D50=(1-edc(0.05, :))\*100 e tende a 100 per alta clarity

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October 20

NEXT STEPS:

* Do measures mantaining a cental positioned mic and trying to cover/remove the tv, compare the results focusing on the 1600 Hz zone 
* Decide if remove the tv 
* Set up measurements using more microphone’s positions
* Sub compensation
* Sub positioning through real time measurements or other method to be defined

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October 23

SMOOTHING of the FFT signal’s plots

The function smoothSpectrum offered by the Matlab toolbox IoSR, containing functions and classes for various purpose as auditory modelling and signal processing was used to visualize a smoother spectrum, neglecting the swing trend. The used function applies 1/N-octave smoothing to the frequency spectrum using a Gaussian window whose centre frequency is f(i), and whose standard deviation is proportional to f(i)/N, where N is chosen to be equal to 5.

ANALYSIS OF THE IR WITH COVERED TV

An absorbing panel has been placed in front of the TV screen to analyze the spectrum of the impulse response under these altered conditions.

The microphone was placed at the listening position, using the same setup as on September 29, which also involved removing the headrest.

The behavior is analyzed individually for each loudspeaker response, for the sum of their responses, and for the impulse response obtained by activating all the loudspeakers simultaneously.

It has been observed that the trend in consistent in both configurations, hence the decay of amplitude in the spectrum around 1600 Hz persists. Consequently, it can be concluded that the TV screen is not the source of this disruptive reflection or absorption.

NEXT STEPS:

* Understand Lunderby method. 
* Gap around 250 Hz in LS L4-L3, shall we investigate on this? No
* Gap around 200 Hz in LS L1-L2-L23, shall we investigate on this? In Ls1 the gap is less present with tv No
* Possible solutions to try: cover with absorbing material the ribs on the ceiling, otherwise try different measurements positions, it could be the seat

The Lundeby-Algorithm is used for noise detection. The algorithm uses the last part of the RIR to estimate the noise level, the determination of integration limit and noise compensation of the impulse responses through an iterative procedure.

October 26

More measurements where done in order to analyse the acoustical behaviour where the TV screen is removed and the door covered by absorbing panel.

The microphone was set in the driving central point and the headrest removed.

* Add material of the absorbing panel and thickness (10 cm?) (sono dietro la TV)

The loudspeakers are activated singularly and then all at once, as previously done. In each situation the behaviour is similar, preserving the decay of amplitude in the 1600 Hz zone.

It can be state that nor the door, nor the Tv screen are the accountable for the problem stated.

NEXT STEPS:

1. perform the measurements in different positions to ensure that the decay around 1600 is caused by the seat and not by other characteristics(room itself, other objects).
2. Solve the calibration problem

October 30

Problems:

* Correction factor after the windowing: fftfactor=20log10(sqrt(ECF)\*4\*nfft/lunghezza\_filewav) what does this means?
* It is noticed that the amplitude of the total IR obtained by the sum of the single morel IRs was not correct. In fact the single signals were firstly cut and then summed together, if the sum is firstly obtained using the whole signal and therefore cut, the amplitude matches the one presented by the AllAtOnce file. 

Energy correction factor

<https://community.sw.siemens.com/s/article/window-correction-factors>

The proper procedure should be the multiplication of the linear spectrum (pascal) by the amplitude correction factor of the hanning window(1/mean(w)= about 2) and then translate into dB.

My suggestion is to multiply the linear spectrum by winfactor.

TO DO:

CODE DETAILS on calibration

In general, to return a FFT amplitude equal to the amplitude signal which you input to the FFT, you need to normalize FFTs by the number of sample points you're inputting to the FFT:



Note that doing this will divide the power between the positive and negative sides, so if you are only going to look at one side of the FFT, you can multiply the FFT by 2



October 31

More measurements are effectuated: in particular the seat has been removed and the pedal too.

The mic has been placed exactly in the same position that had when the measures were done on the chair.

In this case the TV is kept in the room.

The results seems to be quite similar to the previous ones.

The gap in 1600 Hz is a bit attenuated.

In order to plot the correct sum of the impulse responses the cut of the single signals is done after the summation, additionally the single signals are aligned together in order to compensate the delta caused by the different distances (in particular the ceiling loudspeaker).

TO DO:

* Analyze room structure/building/ position of the loudspeakers
* Last measure: no TV, no chair
* Could we compare the properties we got with an existing ambisonics room

Discussion about the predistortion of the loudspeakers basing on the room characteristics.

One last suggestion is to try the measurements with no chair and no TV.

Chain in the code:

* Load sweep
* Calibrate sweep
* Convolve and obtain IR
* Align all the impulse responses
* Sum in the total
* Cut the total
* Window
* Fft and fix the amplitude
* Smooth 5 octave band
* Plot

Immersive sound reproduction Rooms:

<https://www.ircam.fr/article/connaissez-vous-lespace-de-projection>

\*<file:///C:/Users/User/Downloads/THREE-DIMENSIONAL_ACOUSTIC_DISPLAYS_IN_A_MUSEUM_EM.pdf>

<https://pesaromusei.it/sonosfera/>

<https://www.conservatoriorossini.it/lems-space/>

Farina, in \*, suggests a range of values for T20, but the provenience of these values are not known.

OBSERVATION:

The allAtOnce file is not used anymore because the morel above the driver is not aligned.

02 November

The measurements are carried out by utilizing the non predistorted sweep.

* PCB microphone
* Configuration 1(TV and Door not covered, seat present)
* Two lateral microphones head distanced

The trend is still the same on 1600 Hz

Used sweep: 5 seconds

ATTENTION: the amplitude differs because of a different sweep was used so that it could present a different sound intensity.

Therefore it is not a problem of the predistortion, nor of the microphone.

03 November

The cases number 7 and 9 have been analyzed with and without a grid, with pre-distorted and non-pre-distorted signals.

Changes are noticeable only at low frequencies, where the speaker has been compensated.

06 November

With the use of real-time analysis with Smaart, all the speakers were reanalyzed, and it was noticed that the drop at 1600 Hz was due to the fact that the tweeter's phase had been inverted in the past.

Microphone gain applied: +35 dB.

The speakers were all phase-aligned.

Volume imbalance gain adjustments were made, for both tweeter and woofer:

* -4 dB Morel n. 25
* +4 dB Morel n.13

07 November resume of the previous week

* It has been verified that the 1600 Hz gap is present in numerous speakers, with a uniform distribution. Number 19 is definitely flat.
* The different configurations do not significantly improve or alter this behavior.
* The summed signals were first aligned to balance the incorrectly maintained distances in the speaker placement.

1600 HZ PROBLEM: SOLUTION OBTAINED :

All the tweeter except n. 19 had the tweeter reversed of phase, this leads to a loss of amplitude in the frequencies under the crossover zone.

All the cables of the tweeters have been set to the correct position.

NEXT STEPS

Analyze the subwoofer> use the 5 seconds inv sweep, just the spectrum, not t20 clarity

Could be the room improved?

Position the subwoofer

Descrive the problems of the room and apply final equalization

8 november

The REW software and RePhase were tried on test impulse responses.

The procedure seems to work but the procedure to fix the phase should be defined.

The obtained file is a wav and should be sent to the amplifiers to fix the single morels.

The created FIR is optimized for the equalization of the spectrum amplitude.

Firstly the measurement is loaded in REW, smoothed and exported as .txt, therefore loaded in RePhase where the equalization of the spectrum could be done. The exported file is, then, a wav file and it will be the FIR.

10 November

About the correction of the phase:

It has been noted that the time di\_erences between

loudspeaker signal arrivals are an important factor in

the perception of spatial images [14], due to psychoa-

coustic factors such as the precedence e\_ect [15, 16]

where late arriving signals are given less prominence

than earlier ones when evaluating direction. When two

identical stimuli are presented from di\_erent loud-

speakers with a short time delay, precedence e\_ect

experiments have shown that as the time delay in-

creases to 1 ms the image shifts to the direction of

the leading sound. (Off-centre Localisation Performance of

Ambisonics and HOA For Large and Small Loudspeaker Array Radii

Proposals:

* use alignment tool to find the value of delay to align tweeter and woofer
* flat the phases of the loudspeaker as RCF suggests
* equalize the frequencies gains

Notice:

* is has been noticed that tweeter and woofer are already aligned in phase and the spectrum is optimized from this alignment.
* The analysis was conducted exciting with the predistorted sweep being now every tweeter phase-fixed.
* To test the room itself excluding the contribution of the Morels 6 balloons were popped:
  + Three balloons covering the door and tv with absorbing material (5 cm for the TV, 2.5 cm for the door),
  + Three balloons preserving the tv and door not covered.

Balloons positioning:

* + - Top, 70 cm upon the microphone in driver position
    - Back, 1,20 m distant from the mic in the door direction
    - Front, 1,40 m distant from the mic, in the TV direction

These last tests can be use exclusively to analyse the room in terms of acoustics parameters, in the case all the frequency bands are detected for the identification of the values needed.

In particular the driving position is of our interest, the other two positions are useful just for a comparison, to describe the acoustic quality even in the limit of the loudspeaker sphere.

November 14

The different methods for the estimation of the acoustic parameters are again under study because of differences present in the various softwares/applications:

REW

The values are without doubts too high if the time reversed filtering is not applied.

The Time reversed filtering control applies the octave band filters backwards in time, this greatly reduces the filter's own contribution to the measured decay. When using 1/3 octave filters at low frequencies the filter decay time can be significant, over 200 ms for a 100Hz 1/3 filter, for example. Applying the filter in reverse reduces this decay to less than 50 ms, but it does affect the response somewhat, such that Early Decay Time (EDT) figures using Time-Reversed filters may not be valid.

15 November

There has been found a decisive problem in the use of ITA toolbox. Where the wav file is saved with the proper rescale [-1 1], to avoid clipping of the data caused by the audiowrite function, the Shroeder’s curve for the needed parameters is not detected well, as the impulse is too short to be detected by the application.

The solution applied is the use of irStat Matlab toolbox, with its use the rt matches sufficiently with audition, balloon analysis and sweep presents similar trends.

By inserting ‘mean’ in ‘spec’ options in irStat, the mean of rt is calculated

Returns the reverberation time using a method based on ISO 3382-1:2009. The function uses reverse cumulative trapezoidal integration to estimate the decay curve, and a linear least-square fit to estimate the slope between 0 dB and -60 dB or the range defined iny\_fit. Estimates are taken in octave bands and the overall figure is an average of the 500 Hz and 1 kHz bands, if the mean is requested

The function determines the direct sound as the peak of the squared impulse response.

With spec='full', the function returns the RT and EDT as calculated for each octave band.

y\_fit [-5 -25] for the t20

Octave-band filters are calculated according to ANSI S1.1-1986 and IEC standards.

20 November

TUNING

The FIR to fix the amplitude spectrum trend will be created on REW with the use of the automatic filters on the equalization tool.

It is supposed to be 1 unique FIR both for the tweeter and woofer because of the signal split in digital at the beginning of the chain, where is actually the crossover located🡪 to check in experimentation.

To do:

Fix the report and finish it.

Try to get the parameters from the ambisonics data.

DOMANDE

C50, T20, EDT in octave bands is ok?

Better visualize T30 than T20?

Confirm cumulative trapezoidal integration instead of backward time reverse integration and not use of noise detection.