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TUverb

Report TUB Soundscape project: measurement and intervention

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1 Introduction and Objective

As part of the measurement and intervention group, TUverb aims at taking an experimental approach to analysis of soundscapes in the TU Campus, through breaking them down into their acoustical components.

The three main components for this project are:

- A **static soundscape** - captured as field recording
- A **dynamic soundscape** - recording of single events for later insertion
- The **decay characteristics** - captured as an impulse response

The probable evaluation approach is going to be through qualitative research, in the form of a free sound listening session, where the participant will be able to assess existing TU spaces, synthetic spaces created with TU existing components, as well as new spaces that each participant can customize to its personal preference.

This process will be supported by a software framework [1] that allows the user to select different spaces and customize them with preset parameters, all through a graphic user interface.

The final goal is to provide a base of 'spirits' of different rooms of the TU and being able to set every input sound in the context of an TU space in real-time and therefore convert it into a new soundscape. A list of specific and endless[2] field recordings which can be underlining the input sound accomplishes this approach.

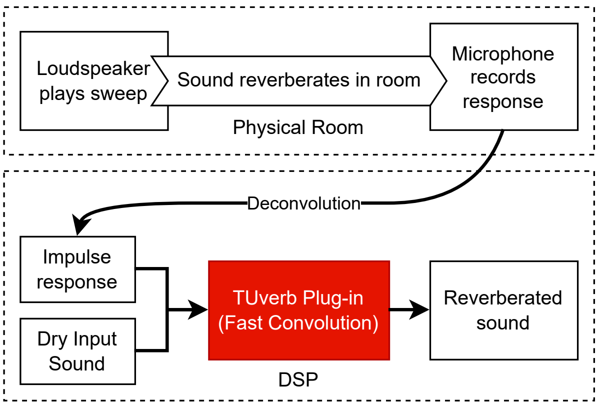

Figure 1: The process from collecting IRs to the resulting reverberated output

[1] in the form of VST Plugin made with JUCE

[2] They will be edited in a way, that they can endlessly be chained together with no hearable disturbance.

2 Methodology

The acoustical characteristics of a space will be recorded in two steps:

- The reverb characteristics will be sampled in the form of an impulse response (section 2.3).
- The original soundscape will be recorded as a field recording, to be used later as a foundation for the design of new experimental spaces (section 2.4).

2.1 Recording Setup

Three main elements are necessary for a proper impulse response recording:

- A full-range loudspeaker[3]
- A microphone array[4]
- A digital recording device[5]

Figure 2: The three main elements of the recording setup

[3]with known frequency response

[4]they should have matched directional characteristic. Sensitivity and Frequency response should at least be known for compensation

[5]which can interface all used microphones at once

2.1.1 Loudspeaker

For our practical recording setup, one IK Multimedia Micro Monitor will be used. With its fairly linear frequency response [6] from 45Hz to 22kHz, high SPL capabilities with low THD and portability, it is a good choice for this process.

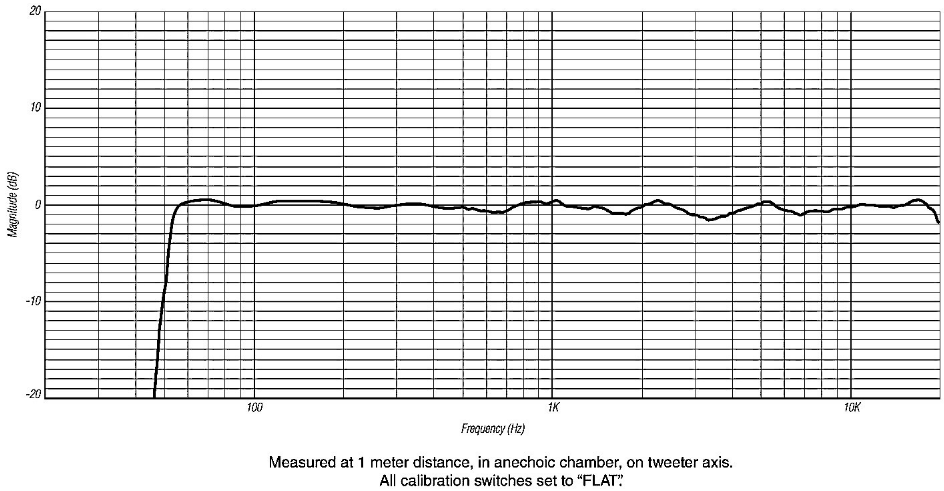

Figure 3: iLoud MM Frequency Response[7]

2.1.2 Microphone Array

The sweep signal played through the loudspeaker will be captured with a stereo ORTF array [9]. This allows for a good spatial localization that works both with headphones and stereo reproduction with loudspeakers [2].

The main microphone set will be a matched pair of Beyerdynamic MC930 cardioid small diaphragm condenser, which have a quite flat frequency response (See Figure 4).

[6] up to -10 dB

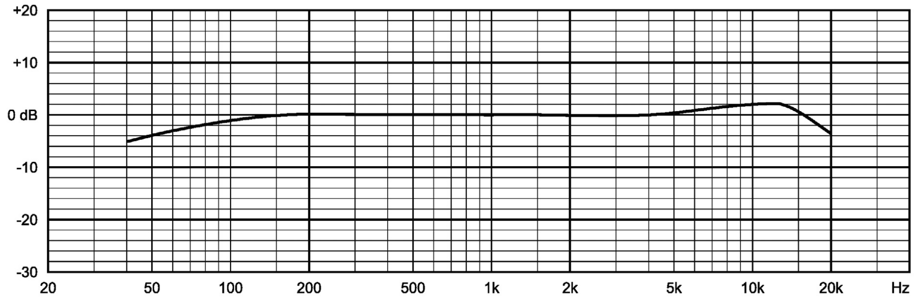

Figure 4: MC930 Frequency Response with ±2dB max. Ripple, Sensitivity of 30mV/Pa

2.1.3 Digital Recording Device

The microphone will be powered and recorded through an RME Babyface Pro FS[1], a high-end audio interface with extremely flat frequency response characteristics (See Figure 5).

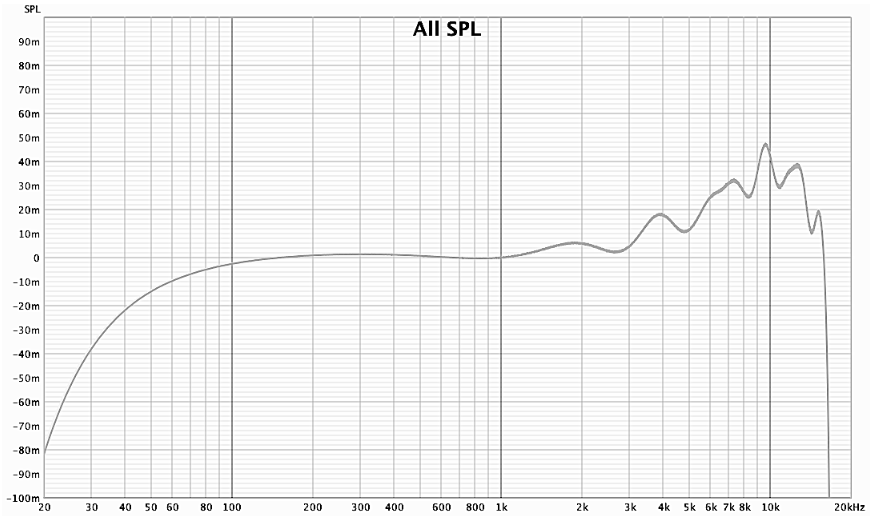

Figure 5: RME Babyface Pro FS I/O Loopback Plot with +0.048bB and -0.08dB max. Ripple

2.2 Equipment Calibration

For a proper and absolute measurement from the TU spaces, some calibration and knowledge of the equipment is necessary. In the end, we want to draw a measurement chain where every part's frequency influence is mapped. This way we will be able to provide absolute measurements of the room's acoustical behaviors and therefore comparative sounds can be created. This has a great impact on the perception of a possible soundscape.

The focus in this chapter will be the theory of device calibration.

In general, microphones are transducers which convert the change in air pressure into electrical signals. In the externally viewed mode of operation, they are then distinguished by their directional characteristics, as well as the sensitivity. The directional characteristic describes the directional and frequency-dependent sound conversion. For a uniform and spatially independent recording of acoustic signals, a microphone with an omnidirectional characteristic, i.e. one that is spatially and frequency independent, is therefore required. Condenser microphones with a closed capsule fulfill this criterion. These come from the category of transducer microphones. This has the further advantage that they convert the sound evenly, regardless of the distance to the sound source, so that no differentiation of the measuring ranges, i.e. a differentiation between near, far or diffuse field, has to be made. Sensitivity is a constant measure of the microphone's ability to convert sound pressure into a voltage level. It is usually specified for a specific frequency, so that an absolute sound pressure and thus an absolute loudness can also be assigned to a measurement.

In our case, loudspeakers are sound sources that are directed and can be controlled from outside as an excitation source. The demands on these are much lower and more practical, such as a low weight and a compact size. Nevertheless, a measurement is absolutely necessary in order to obtain the frequency response of the loudspeaker and also to be able to take into account any non-linearities (i.e. frequency differences at different volumes) in the transmission. When measuring, it is only important to direct the microphone towards the tweeter of the loudspeaker, since the high-frequency sound waves propagate in a more directional manner.

The same applies to the measurement of the microphone if a loudspeaker with a separate tweeter is used for excitation from it.

Finally, we now come to the concrete measurement. For this we went to the anechoic room of the TU, a room equipped with various sound absorbers, in which there are practically no reflections for all airborne sound frequencies above 60Hz. This allows the acoustic properties of the loudspeakers and microphones to be isolated and measured. To simplify the process and have a fixed reference, we also borrowed a measurement microphone with a known frequency response and sensitivity. The measurement itself then proceeds very stringently along the measurement chain. We start at the audio interface, continue via the loudspeaker and finally to our own microphones. The interface is initially fed back from its output to its input, so that the absolute input level sensitivity can be determined and a frequency response of the input analog-to-digital converter can be created.(The measurement is performed by a frequency sweep of 20kHz-20Hz) Subsequently, the frequency responses from the loudspeaker to the borrowed measurement microphone are recorded. It is important to record and document different volume levels in order to record non-linear behavior. In addition, several recordings are made per setting in order to filter out random variables. Finally, the own microphones are measured using a fixed loudspeaker dynamic. The data is recorded raw, and the process is documented. This way, all frequency responses can later be averaged and correctly offset against each other, so that the equipment is finally ready for the creation of absolute measurement recordings.

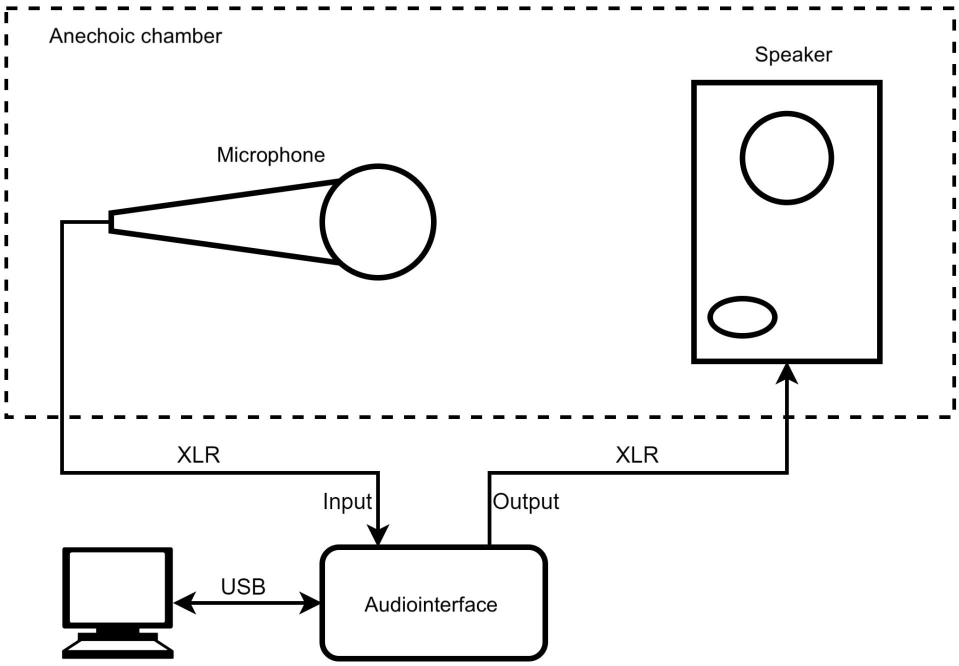

Figure 6: schematic structure of a calibration setup

2.3 Impulse Response Recording

A loudspeaker is necessary for the reproduction of a logarithmic frequency sweep signal used to excite the room. This method is more robust than a short transient test signal, and therefore leads to better results.[4]

The test signal will be played through the speaker and recorded with the ORTF microphone array.

The impulse response will be created through deconvolution of the test signal with the recorded signal.

These impulse responses can be used with a convolver to add space characteristics to additional sound content, which will be added to the field recordings, thereby creating new, synthetic soundscapes that maintain the original characteristic of such space.

2.4 Field Recording

A set of field recordings of the sampled environment will be recorded with the setup described in section 2.1, for the purpose of being played as a soundbed together with added sounds, to increase the authenticity of the soundscape. Having a separate soundbed element further allows for experimentation in the form of mix and matching with other sampled impulse responses. All of these tools allows for an extensive creation of new, undiscovered sound environments.

2.5 Software Development

2.5.1 Framework

In an attempt to make the software usable for as many people as possible and regardless of type of computer or operating system they might use, it was decided to write an audio plugin with help of the "JUCE" framework [6], which is written in the C++ programming language. It allows for building a standalone application or a plugin in one of the most prominent formats (VST3, AU, ...) from the same source code files.

This will also provide a lot of flexibility in setting up a possible intervention on the campus in the near future, as any audio workstation will probably be compatible. The plugin can even be run on a small single board computer like the "Raspberry Pi" [8] and works in real time.

The whole software project is version controlled through means of *git*. FRUT[5] is used to convert the JUCE projects to a CMake[3] project, which in turn can be built almost everywhere.

2.5.2 Features

The software processes input samples and simulates the acoustics of some of the rooms in TU Berlin by means of convolution with a prerecorded impulse response. A drop-down menu facilitates room choice for the user. As of yet, it features two more controls that can be set within the plugin. The first one is a "Volume" parameter, that has a range of $-24\,\mathrm{dB}$ to $12\,\mathrm{dB}$. Secondly there is a "Dry / Wet" parameter, that sets a mixing ratio of processed samples to their unprocessed counterparts. This makes it possible to dial in the amount of artificial reverberation applied to any input signal. All parameters are fully automatable, so the plugin could even be useful in other contexts.

Figure 7 shows the current plugin user interface as rendered in a digital audio workstation.

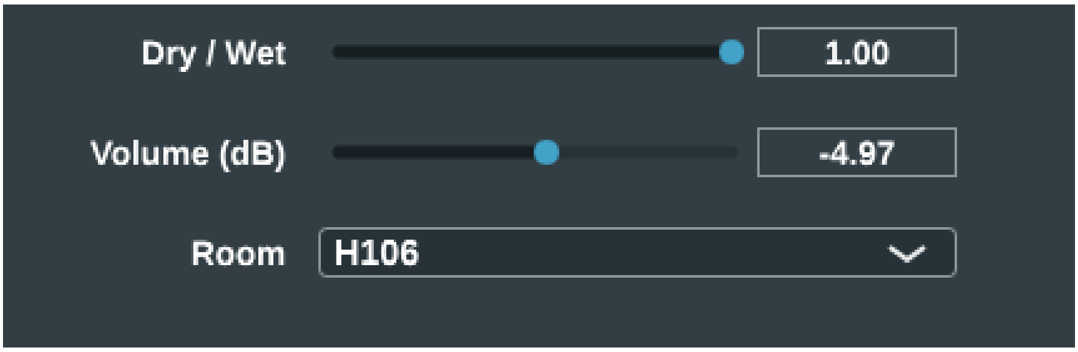

Figure 7: TUverb plugin user interface

It is planned that some more features might be added to the plugin in the coming two weeks. One of which may be the addition of field recordings, which could be mixed in with the output signal. Also the software will feature more impulse responses than currently available.

2.5.3 Technology

Internally the plugin is using the method of *uniformly partitioned fast convolution*[10]. In this method signals are processed blockwise, which decreases the processor's computational load (as compared to processing individual samples). The fast convolution algorithm corresponds to a frequency domain multiplication. First the Fourier transform of both the input signal and the impulse response are found. They are then multiplied and inversely Fourier transformed to find the simulated room response. There are some other technicalities, that make it possible for an audio plugin to do all the calculations efficiently in real-time and without audible glitches in the audio, but explaining all of those would presumably be out of scope for this report. Luckily, the JUCE framework abstracts a lot of this away and deeper need of the techniques used is not necessary to use or even extend the plugin.

3 Sound Art and Interventions

The TUverb is a useful tool to compare the sonic character of different spaces in the TU campus. Furthermore it can be used for sound art and interventions. One possible project could be to recreate Alvin Lucier's sound art piece "I Am Sitting in a Room". Instead recording multiple iterations of feeding back the sound of a loudspeaker into a microphone, always with the sound of the room, one can take the TUverb and apply it to the recorded voice multiple times.

A demo can be found here: https://soundcloud.com/schygiol/i-am-sitting-in-h106

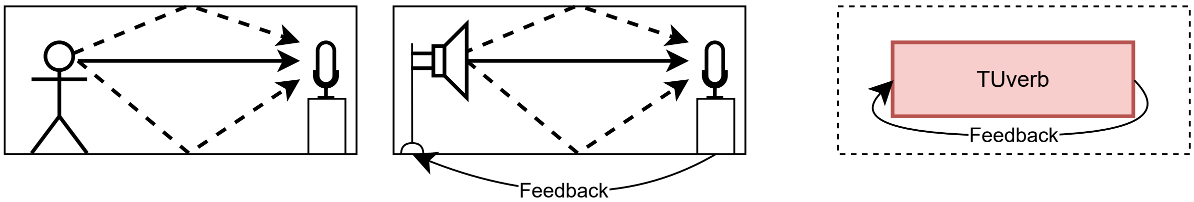

Figure 8: ("I'm sitting in a room")

We are also planing an intervention where we gonna use the TUverb plugin to different TU spaces where one would not expect them. By that we would like to show how the sound of a room can change the perceived environment. For example by giving an dry sounding room like a student café the reverb of a lecture hall might create interesting reactions, like discomfort or raised concentration.

References

[1] RME Audio. *Babyface Pro FS Datasheet*. 2022. URL: https://www.rme-audio.de/babyface-pro-fs.html (visited on 06/20/2022).

[2] Andrea Azzali et al. "Reproduction of Auditorium Spatial Impression with Binaural and Stereophonic Sound Systems". In: (2005), p. 12.

[3] *CMake Homepage*. URL: https://cmake.org/ (visited on 07/07/2022).

[4] Angelo Farina. "Advancements in Impulse Response Measurements by Sine Sweeps". In: Audio Engineering Society Convention 122. Audio Engineering Society, May 1, 2007. URL: https://www.aes.org/e-lib/online/browse.cfm?elib=14106 (visited on 06/20/2022).

[5] *FRUT Project Repository*. URL: https://github.com/McMartin/FRUT (visited on 07/07/2022).

[6] *JUCE Framework Homepage*. URL: https://juce.com.com/ (visited on 07/07/2022).

[7] IK Multimedia. *iLoud Micro Monitor Datasheet*. 2022. URL: https://www.ikmultimedia.com/products/iloudmm/index.php?p=specs (visited on 06/20/2022).

[8] *Raspberry Pi 4, Single Board Computer*. URL: https://www.raspberrypi.com/products/raspberry-pi-4-model-b/ (visited on 07/07/2022).

[9] eberhard Sengpiel. *Das Stereo Mikrofonsystem ORTF*. Aug. 1998. URL: http://www.sengpielaudio.com/DasStereo-MikrofonsystemORTF.pdf (visited on 06/20/2022).

[10] Jan Wilczek. *Fast Convolution: FFT-based, Overlap-Add, Overlap-Save, Partitioned*. URL: https://thewolfsound.com/fast-convolution-fft-based-overlap-add-overlap-save-partitioned/ (visited on 07/07/2022).