Measuring Reverberation Times in Active Acoustic MIMO Systems

Clemens Frischmann[1], Fabio Kaiser[2]

[1] TU Graz, 8010 Graz, Austria, Email: clemens.frischmann@student.tugraz.at

[2] Rohde Acoustics, 1070 Vienna, Austria, Email: kaiser@rohde.at

Introduction

In room acoustics a versatile approach becomes more and more mandatory due to the variety of events in one venue. Adjustable acoustic conditions can be accomplished with traditional methods by, for instance, adjustable absorbers and coupled rooms, or in a more modern approach by active acoustics systems. To implement, calibrate and maintain such active systems, room acoustic and electroacoustic measurements have to be conducted. Therefore engineers and measurement equipment have to be brought in, to carry out these essential measurements. This is both time- and cost-intensive. So why not use the existing infrastructure of such systems and conduct measurements with the existing hardware?

The aim of this work is to verify, if measurements conducted system internally reach results similar to measurements performed compliant to ÖNORM EN ISO 3382-1 [1]. There are many differences in these measurement setups, which are discussed in the section below. Therefore, it has to be verified, if satisfactory results can be achieved. To do so a case study has been carried out. In a room with installed active acoustics, impulse responses from all acoustic signal paths, so all loudspeaker-microphone-paths, have been measured for different system settings. Reverberation times (T_{30}) have been calculated in octave bands from 63Hz to 8kHz. By comparing the results with the ISO conform measurements, conclusions could be drawn and the number of mandatory measurements (loudspeaker-microphone-pairs) could be minimized.

Both, ISO conform and system internal measurements have been conducted using the ITA-toolbox[2] for MATLAB. Reverberation times were calculated with the command `ita_roomacoustics()` from the same toolbox.

Difference to ISO 3382-1

The main differences are found in loudspeaker types, microphone types and measurement positions.

Loudspeakers

In ISO 3382-1 [1] the measurement loudspeaker is defined to be omnidirectional (to a certain degree) for the investigated frequency range. This is not the case for AMS measurements. The loudspeakers used in active acoustic systems are directional for mid and high frequencies. In most cases the main speakers of active acoustic systems have a lower boundary frequency of about $f_u \approx 100Hz$. This can differ between systems, dependent on the loudspeakers used. All active acoustic systems under investigation have LFE channels to compensate the main speakers for low frequencies. These additional channels have not been used for measurements in this work.

Microphones

As stated in ISO 3382-1 [1] measurement microphones must be omnidirectional. To reach a certain level of feedback stability (gain before feedback), directional microphones are often used in active acoustic systems. The systems investigated in this work incorporate microphones with super cardioid polar patterns. The frequency response (for the 0° elevation and azimuth direction) is flat for all frequencies of interest with a +3dB boost at $f = 12$kHz

Measurement Positions

In ISO 3382-1 [1] loudspeaker and microphone positions are described. It is stated, that microphones must not be placed in the vicinity ($\frac{\lambda}{4}$ of the lowest frequency of interest) of reflecting surfaces. Microphone positions must be $\frac{\lambda}{2}$ (of the lowest frequency of interest) apart. It is also written that microphones and loudspeakers must not be „too near“ to each other, to prevent strong impact of direct sound. These requirements may not be met for all positions of active acoustic systems. Dependent on the system, microphones are mounted on the ceiling or walls or be suspended from the ceiling. For the case of ceiling and wall mounted microphones, the requirements for a minimum distance to reflecting surfaces can not be met. Microphones and loudspeakers are placed equally distributed, therefore minimum distances between microphones (and loudspeakers) can not be fulfilled for all positions. Positioning requirements in terms of hight can also not be met. ISO 3382-1 [1] states a hight of $h_S = 1,5$m for sources and $h_R = 1,2$m for receivers (seated).

Case-study

Test measurements have been conducted at the PKE-Showroom located at PKE Austria headquarters in Vienna. Here test and demonstrations are performed for media installations, security installations and Amadeus Active Acoustics. The installed system is a rather small, with 24 loudspeakers and 12 microphones. In addition four sub-woofer-loudspeakers are installed, which have not been used in this test. The venues layout is shown in Figure 2. With a footprint of $A \approx 135\text{m}^2$ and a mean hight of $\bar{h} \approx 3,3$m a volume of $V \approx 446\text{m}^3$ results.

As always with active acoustic systems, a dry initial acoustic situation is intended, as such systems are only able to extend reverberation. To describe the initial state of the room, reverberation times in octave bands are shown for the inactive system (measured ISO 3382-1 [1] compliant) in Figure 1. We can see a maximal mean reverberation of $T_{30}\mid_{f=250\text{Hz}} = 0,85$s for the 250Hz octave band. This room is also used for presentations and meetings, and so has a reverberation time to meet the requirements of

ÖNORM B 8115-3 [3] for the class *Speech*.

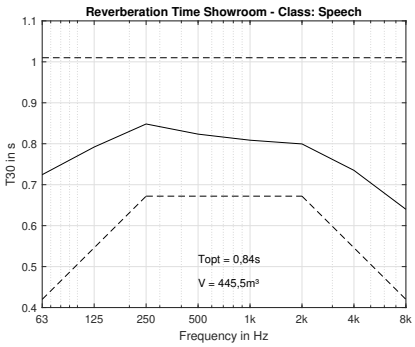

Abbildung 1: Reverberatin time T30 in octave bands of the Amadeus showroom (system inactive) with the standard bounds for the class *Speech* from ÖNORM 8115-3 [3]

Measurement compliant with ÖNORM EN ISO 3382-1

ISO 3382-1 [1] conform measurements were conducted for two extreme system states: System Off and the system setting producing the longest reverberation time. For each of the measurements the same measurement positions, two source positions and six microphone positions, were used. In Figure 2 all positions and reference planes are marked. Distances to record positions were measured perpendicular to reference planes and are listed in Table 1. Source positions are located in an area where orators would talk. Microphone positions were chosen to sample the room homogeneously. Due to the room being equipped to hold presentations, tables and chairs are arranged in U-form oriented to the x-basis shown in Figure 2.

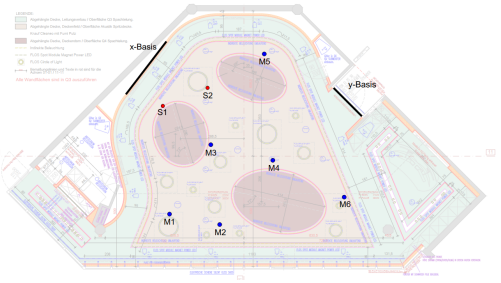

Abbildung 2: Measurement positions of ISO conform measurements for all pre-sets [4]

Tabelle 1: Measurement position coordinates of ISO conform measurements for all pre-sets in [m]

	S1	S2	M1	M2	M3	M4	M5	M6
x	1, 6	2, 8	4, 8	6, 9	4, 4	7, 0	3, 9	10, 6
y	6, 0	4, 0	9, 3	8, 0	5, 7	4, 2	1, 1	3, 1

An exponential sweep was used as measurement signal. With a sampling rate of $f_s = 44100$Hz and an FFT-order of 19, a sweep duration of $T = 11,9$s resulted. The results will be shown in Section „Results“, when comparing measurements.

System internal measurement

For the system internal measurement, all 12 microphone channels and all 24 loudspeaker channels were selected, so 288 impulse responses have been measured. Contrary to the ISO measurement before, measurement signals were led through a digital audio network, DANTE. In Figure 3 a block diagram of the measurement set-up for the system internal measurement is shown. The processing unit in the center is connected to in-going and out-going amplifiers via DANTE, which is part of the installed infrastructure. From there on analogue signals are led to loudspeakers and from microphones through copper cables (standard speaker cables and symmetric microphone cables with XLR connectors). The measurement computer is incorporated into the DANTE network through the DANTE-Virtual-SoundCard. Measurement signals (to loudspeakers) and response signals (from microphones) are sent to and received from the active acoustic processor. Therefore, patches were inserted into the processing software which bypass all equalizers and all other effects by virtual Y-connectors. So, measurement signals play directly to loudspeakers and response signals record directly from microphones, without disabling the active acoustic algorithms. This enables measurements in an active system state. The sampling frequency was set to $f_s = 48000$Hz. With an FFT order of 19 an exponential sweep length of $T = 10,9$s results. Figure 4

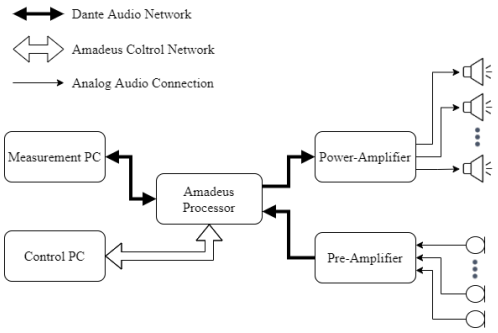

Abbildung 3: Inclusion of external measurement and control computer in the Amadeus Active Acoustic system

shows loudspeaker and microphone positions of the active acoustic system. Due to the approximation of round boundaries by straight lines, loudspeaker position twelve appears to be outside. All positions are available in three dimensional vectors. Thus, for loudspeaker-microphone paths characteristic lengths can be calculated.

Results

Before discussing reverberation times some quality measures to compare reverberation times have to be introduced. The goal is to reach similar reverberation times like the mean reverberation times from the ISO compliant measurements. To quantify similarity the just noticeably differences (JND) was chosen. ISO 3382-1 [1] states this JND to be $JND_{ISO} = 5\%$. Blevins at. al. [5] on the other hand conclude a JND of $JND_{blev} = 24,5\%$ in their paper from 2013. So, values quantifying JNDs for reverberation times tend to differ sharply. To judge similarity of the reverberation times the ISO compliant

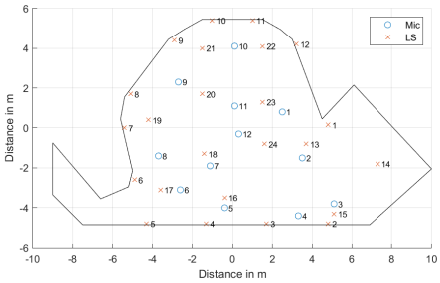

Abbildung 4: Active acoustic system: Microphone positions, wall mount loudspeaker positions (1 to 12) and ceiling mount loudspeaker positions (13 to 24)

JND of $JND_{ISO} = 5\%$ is used. The system internal measurement works fine, if mean values lay inside the JND boundary around the ISO measured reverberation time mean. So, a maximal error of 5% must not be exceeded. This means reverberation times tuned to fit both measurement types would not differ audibly.

Figure 5 and 6 show reverberation times from both measurement methods, compared for each octave band. In Figure 5 the full graph is depicted whereas Figure 6 shows a zoomed version of the same graph. In green the statistical evaluation of the ISO 3382-1 [1] compliant measurement procedure can be seen. In blue the evaluation of the *full set* is drawn, so all 288 measurements carried out system internally. Each box represents the interval from the 25% percentile to the 75% percentile. The whiskers drawn as dashed lines from each box depict the upper and lower quartile. Outliers are drawn as red x, medians as red lines and arithmetic mean values as black + signs. Outliers are defined as data points 1, 5 times the interquartile range above or below the interquartile box. The horizontal dashed line represents the JND according to ISO 3382-1 [1] around the mean value of the ISO conform measurements for each octave band. The white area marks the JND as described by Blevins at. al. [5]. As seen in Figure 6 system internally measured arithmetic means lay inside the JND boundaries (ISO compliant JND of 5%) for octaves from 250Hz to 4kHz. Therefore we meet our set condition for this frequency range. In Figure 5 many extreme outliers can be seen.

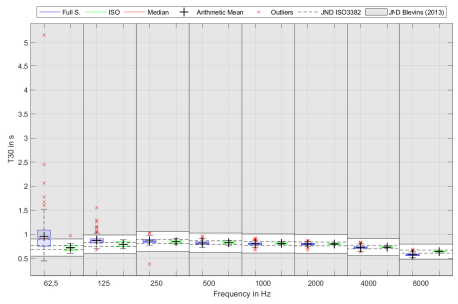

Abbildung 5: Statistical evaluation of reverberation times measured ISO conform and system internally; full set of 288 AMS measurements

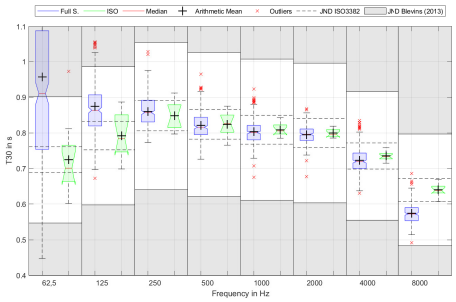

Abbildung 6: Statistical evaluation of reverberation times measured ISO conform and system internally (AMS); full set of 288 AMS measurements; zoomed

In the next step these extreme outliers should be excluded from the evaluation process without worsening the results. Too long measurement paths could be a problem due to insufficient SNR. In this case study it was found, that all measurements have a peak to noise ratio (PNR according to Lundeby [1]) grater than 45dB, which means no long distances have to be excluded from the evaluation process. So, an exclusion of extreme outliers is done by excluding short measurement paths. The lowest considered frequency is the lower boundary of the 63Hz octave band, which is 44Hz. It was found to be best practice to limit the distance between microphones and loudspeakers to be grater than half the wavelength $l > \frac{\lambda}{2}$. With a frequency of 44Hz this results in a minimum distance of $l_{min} = 3,88\text{m} \approx 4\text{m}$. After excluding short measurement paths less outliers occur as seen in Figure 7. Especially extreme outliers have been successfully excluded.

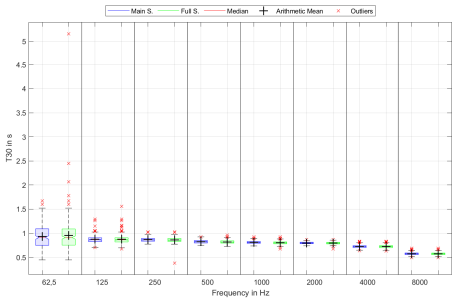

Abbildung 7: Statistical evaluation of reverberation times measured system internally (AMS); full set of 288 AMS measurements compared to main set of 208 AMS measurements

There are still 208 impulse responses and all loudspeakers included in this *main set*, which results in long measurement times (no time reduction compared to the *full set*). Therefore, in the next step it should be investigated, how many impulse responses are necessary to fully represent the main set. This sub-quantity shall be called *subset*. First a *subset* size N is chosen. This size is given as a percentage of the *full set*, as well as a total number of impulse responses. Then N impulse responses are chosen randomly from the *main set*, so the set restricted by measurements path length. After the mean reverberation times of this *subset* are calculated and stored, another

subset is chosen randomly. This operation is repeated until the total number of chosen impulse responses exceeds the size of the *full set*. For example, a *subset* size chosen to be 10% of the *full set* would correspond to 29 impulse responses. The operation of randomly choosing from the *main set* would so be repeated 10 times to result in 290 impulse responses and 10 mean values. This way about the same number of impulse responses form a basis for the statistical evaluation of the *subset* mean values, and no statistical bias is expected. The resulting *subset* mean values are then compared to the *main set* and ISO compliant measurements. If all *subset* means lay inside the JND boundary around the *main set* mean, and all *subset* means lay inside the JND boundary around the ISO compliant measurements for octave bands from 250 Hz to 4 kHz, the *subset* is considered to fully represent the *main set*. A *subset* size is altered until a size is found which just satisfies the mentioned conditions. The fitting *subset* size was found to be $N = 29$ impulse responses, or 10% of the *full set*. In Figure 8 a comparison between the *subset* means and the *main set* is shown. A comparison between the *subset* means and the ISO compliant measurement is drawn in Figure 9. So, for the inactive system the chosen procedure works well for octave band from 250Hz to 4kHz.

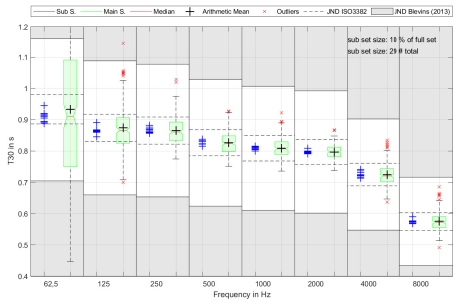

Abbildung 8: System inactive: Comparison between mean values of subsets containing 29 randomly chosen impulse responses and the statistically evaluated main set

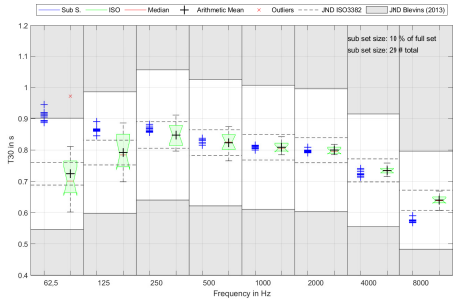

Abbildung 9: System inactive: Comparison between mean values of subsets containing 29 randomly chosen impulse responses and the statistically evaluated ISO conform measured reverberation times

The same evaluation has been performed for the active system. Very long reverberation times, exceeding $T = 10$s in low frequencies, could be reached with this system setting. Figure 10 shows a comparison between *subset* mean values and the statistically evaluated ISO compliant measurements. The *subset* size displayed is $N = 29$ impulse responses, as before. For octave bands from 250Hz to 4kHz all *subset* mean values lay inside the 5% boundary. Therefore, the chosen measurement procedure works for both, an active and an inactive system sate.

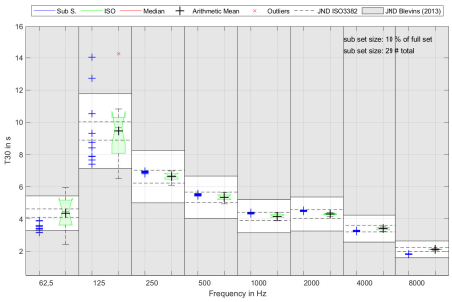

Abbildung 10: System active: Comparison between mean values of subsets containing 29 randomly chosen impulse responses and the statistically evaluated ISO conform measured reverberation times

Conclusion

For an active acoustic system room acoustic measurements have been conducted system internally and compliant to ÖNORM EN ISO 3382-1 [1]. Reverberation times have been evaluated and compared statistically. For system internal measurements, signal path length exceeding $l = 4$m have been excluded. With a set of 29 randomly chosen measurement paths, mean values in octave bands differ less than 5% from reverberation time mean values measured ISO 3382-1 [1] conform. This has been shown for an active and an inactive system state.

Literatur

[1] ÖNORM EN ISO 3382-1: Acoustics - measurement of room acoustic parameters - part 1: Performance spaces (2009)

[2] M. Berzborn, R. Bomhardt, J. Klein, J.-G. Richter, and M. Vorländer: The ITA-toolbox: An open source matlab toolbox for acoustic measurements and signal processing, vol. 2017, pp. 6–9, 2017.

[3] ÖNORM B 8115-3: Schallschutz und raumakustik - teil 3: Raumakustik (2005)

[4] C. P. GMBH: Polierplan, PKE showroom, Plannummer: 60/01, 09.09.2019.

[5] M. G. Blevins, A. T. Buck, Z. Peng, and L. M. Wang: Quantifying the just noticeable difference of reverberation time with band-limited noise centered around 1000 Hz using a transformed up-down adaptive method, 2013.