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Room acoustic evaluation of active acoustics systems – results from measurements

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ABSTRACT

Active Acoustic systems or Acoustic Enhancement systems have been around for quite some time now. The need for variable and optimized acoustics has increased and so has the willingness to install a multitude of loudspeakers and microphones. In the history of active acoustics systems there have been many different approaches to shape the acoustics. The two basic approaches, inline and non-inline, differ mainly by how much acoustic feedback from loudspeakers to microphones is allowed in the system and by the number of microphones. In any case, a dispassionate discussion on the quality of different approaches is only possible if standardized room acoustic evaluation methods are applied, namely measurement of ISO 3382 parameters and subjective tests or interviews with musicians, conductors, critics and concert managers. In this study two installations of active acoustics systems, one in a concert hall (700 seats) and one in a congress center (2300 seats), have been investigated by taking room acoustic measurements with different system parameters set. It can be shown that for example C80 can be altered significantly while maintaining the same reverberation time. Hence, the active acoustics systems in the installations under study were able to shape the subjective impression of clarity vs. reverberance.

Keywords: Active, Acoustics, Enhancement

1. INTRODUCTION

Demands for venues in terms of versatility increased rapidly in the last decades. This does apply to infrastructure and acoustics. For acoustics these demands are defined by the kind of event hosted, and the architecture and size of the venue itself. New concert halls for just one kind of event are extremely rare. Therefore, variable acoustics become more and more important. To provide a satisfying speech intelligibility for verbal shows, strong early reflections are mandatory to achieve high levels of clarity. At classical concerts a more enveloping sound, in form of stronger late reverberation, is intended. Such varying challenges can only be accomplished by expensive structural measures (coupled rooms and variable absorbers) or active acoustics. When it comes to variable acoustic solutions, active acoustic systems become more and more popular. Active acoustics is a term comprising several techniques to influence the sound field. This includes generation and distribution of early reflections, generation and shape of late reverberation and the generation and projection of 3D audio scenes. This kind of approach to room acoustic enhancement exists since the 1950's, starting out at the Royal Festival Hall in London (1). Since then different variants of such systems are commercially available.

During the 1990s research was undertaken to understand the basics of how to control feedback. After this era, it seems that further research was left to companies developing active acoustic systems. However, an acoustician needs to have a certainty of planning when using different acoustic measures. To evaluate to what extend an active acoustic system can influence a sound field without an unsatisfyingly sounding outcome, experiments have been carried out by the authors. In this paper the results of these case studies are discussed, and conclusions have been drawn.

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2. THEORY

2.1 Active acoustics

Active acoustic systems consist of microphones, preamplifiers, A/D- and D/A-converters, a signal processing unit, amplifiers and loudspeakers. Figure 1 shows a schematic drawing of such systems.

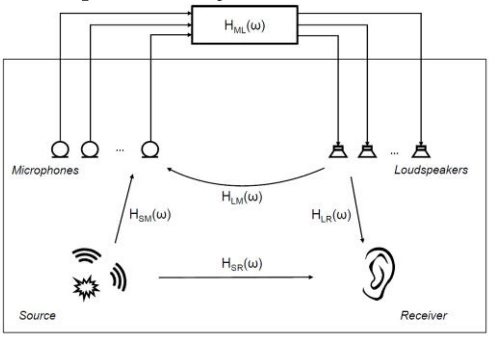

Figure 1 – A basic model of an electro-acoustic system with the relevant transfer functions (2)

The most critical issue is audible feedback. If the transfer function $H_{LM}(\omega)$ exceeds $H_{SM}(\omega)$ the system becomes unstable and feedback audible. The ratio of these transfer functions is called loop gain. This is one condition to prevent instability, which implies: If the signal from the source exceeds the signal from the loudspeaker at the microphone position, the system cannot be unstable.

$$g = \frac{|H_{LM}(\omega)|^2}{|H_{SM}(\omega)|^2} \tag{1}$$

There are two basic approaches to active acoustics: In-Line and Regenerative (Non-In-Line) systems. The main difference between those approaches is how feedback is handled, thus how $H_{LM}(\omega)$ is handled.

In-Line systems use 4 to 8 directive microphones placed near (usually inside the critical distance) the source picking up direct sound and many loudspeakers distributed in the auditorium. The reverberation is either generated by algorithmic methods or convolution with impulse responses (measured or artificial). Due to directive characteristics and spatial separation of microphones and loudspeakers, the loop gain is very low, and feedback is avoided.

The regenerative approach uses signal loops between microphones and loudspeakers to generate reverberation. Microphones are placed outside the critical distance of sound sources and loudspeakers. The basic idea is that the picked-up signals are fed back using different delays via the loudspeakers, however, a reverberation stage can be added to the signal chain to gain more flexibility. The loop gain therefore is higher, and audible artefacts like ringing tones are more likely to occur. In order to increase stability a very high number of microphones and loudspeakers has to be used.

A combination of both approaches is typically called a hybrid system (e.g. Amadeus Active Acoustics, (3)).

2.2 Regenerative systems

The behavior of energy caused by a source in a room can be described by an energy balance (4):

$$V \cdot \frac{\partial E}{\partial V} = P_0 \partial t - P_{absorbed} \partial t \tag{2}$$

where V is the volume [m³], P_0 the power of the source [W], $P_{absorbed}$ the power loss [W] and the term ($\partial E/\partial V$) is called energy density [J/m³], which describes the energy per unit volume. From here on w denotes energy density.

By applying conversions according to Svensson and Möser (4,5), the steady state energy density and the reverberation time for n_L added loudspeakers result as follows.

$$w_{s,on} = P_0 \cdot \frac{4}{A'c} \cdot \frac{1}{1 - n_L S^2} = w_{s,off} \cdot \frac{1}{1 - n_L S^2} \tag{3}$$

$$T_{on} = \frac{246}{c\,log(e)} \cdot \frac{V}{A'} \cdot \frac{1}{1 - n_L S^2} = T_{off} \cdot \frac{1}{1 - n_L S^2} \tag{4}$$

The power loop gain S^2 is dependent on the channel gain μ, the total absorbing area A' and the speed of sound c (4). If the term ($n_L S^2$) comes close to one, the energy density goes towards infinity. An instability occurs because more energy is added to the room than can be absorbed.

Adding n_M microphones to each channel complies to a decorrelated matrix amplification. All microphone signals as well as all loudspeaker signals (at microphone position) are decorrelated. So, equation 3 and 4 change to (4)

$$w_{s,on} = w_{s,off} \cdot \frac{1}{1 - n_M n_L S^2} \tag{5}$$

$$T_{on} = T_{off} \cdot \frac{1}{1 - n_M n_L S^2} \tag{6}$$

If a room with a volume of 4000m³ and a reverberation time of 1s is assumed, a model can be calculated by applying equation 3 and 4. In Figure 2 the reverberation time and the power loop gain are drawn for different numbers of independent channels (number of loudspeakers).

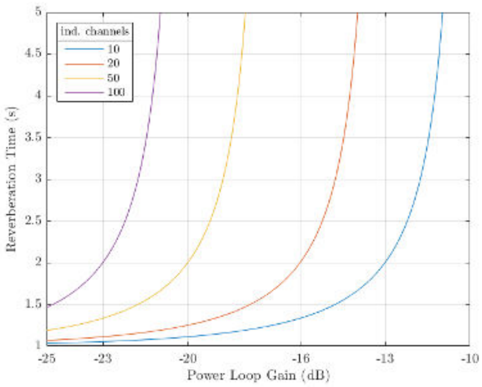

Figure 2 – Reverberation time over power loop gain for different channel numbers

It can be seen, that for a doubling of channels, a 3dB smaller power loop gain is required to yield the same reverberation time. The more channels, the smaller the power loop gain, thus, the channel gain (see equation (17)).

The stability of the system is described by

$$H_{loop} = \frac{H_{ML}(\omega)}{1 - H_{ML}(\omega) H_{LM}(\omega)} \tag{7}$$

To evaluate the stability the mean loop gain is used, which describes a frequency average, so the denominator of equation 7 is bigger than zero. It is defined by the square root of the frequency-averaged squared product of the in-loop frequency responses (4).

$$\bar{S} = \sqrt{\overline{|H_{ML} H_{LM}|^2}}\,, \tag{8}$$

where the averaged frequency response H_{ML} is dependent on the gain factor μ, and H_{LM} on the room conditions. In real systems, the difference between the frequency responses mean and peak value is about 10dB. So, the mean loop gain should be chosen at least 10dB less than one, to be stable.

Barbar and Griesinger recommend assuming a difference (mean to peak) of 12dB and use an additional headroom of 8dB. This results in a total gap of 20dB to avoid feedback and coloration (6,7). Coloration is an unnatural sounding reverberation, often caused by peaks in the frequency response of systems.

A system, which for example rises the reverberation time from one second to two seconds and consists of ten independent channels, can easily result audible artifacts. By adding independent channels, the gain factor can be chosen smaller, maintaining the same reverberation time. So, the gain of the transfer function H_{ML} decreases and the system gets stable. Further, for a system with few independent channels, the probability of instability rises slowly with increasing gain factor and for a system with a high number of channels it rises quicker. Thus, the gain factor has to be decreased more for systems with a small number of channels to be at the same probability of instability (8).

3. METHODS

In order to evaluate the performance of active acoustic systems two case studies have been carried out. Active acoustic systems using a hybrid approach were installed in two venues differing in geometry and size. The experiment was to create different presets of the system yielding similar reverberation times but significantly different results of clarity (C80) or center time (t_s). Measurements were conducted following ISO 3382 (9).

In the following paragraphs the two venues, where experiments have been carried out, are presented. Further the installed active acoustic system's impact is described.

3.1 Convention Center Suhl

The Convention Center Suhl (CCS) is a multi-purpose hall with a capacity of up to 2.300 seats and a volume of 34.000 m³. Because events like television shows, theater plays, chamber orchestral and symphonic concerts are hosted variable acoustics are desirable. To provide appropriate acoustics, an Amadeus Active Acoustics was installed. Amadeus Active Acoustics is a hybrid acoustic enhancement system. It consists of a regenerative stage and an algorithmic reverb. The installation consists of 109 loudspeakers and 14 microphones distributed over the audience area and the stage.

Measurements have been carried out in compliance to ISO 3382-1. The chosen positions for one source and six microphones are shown in Figure 3. Four system presets have been tuned for varying acoustic demands: "Theater", "Camber Music", "Symphony" and "Cathedral". To depict the different acoustics used in CCS, the reverberation time and energy decay curves are shown in Figure 4. A strong impact of the enhancement system is evident.

By comparing the impulse responses of the room, the same early, strong reflections are noticeable (see Figure 9 and 12). They arrive at the microphone position (M3) 24ms, 30ms and 38ms after the direct sound. Further these three reflections show similar levels at about 20% of direct sound energy. This implicates, that these reflections originate from the room itself. Because of the shape of CCS, the reflections are likely to come from the ceiling. Possible paths for the sound to travel in the given times (24ms, 30ms and 38ms) are drawn in Figure 5. By measuring the length of the direct path (sd (red)) and the two reflection paths (sr1 (grey), sr2 (blue)) in Figure 5, distance differences and time differences have been calculated (see Table 1). The time differences for the reflections correlate with the Reflectograms of this position.

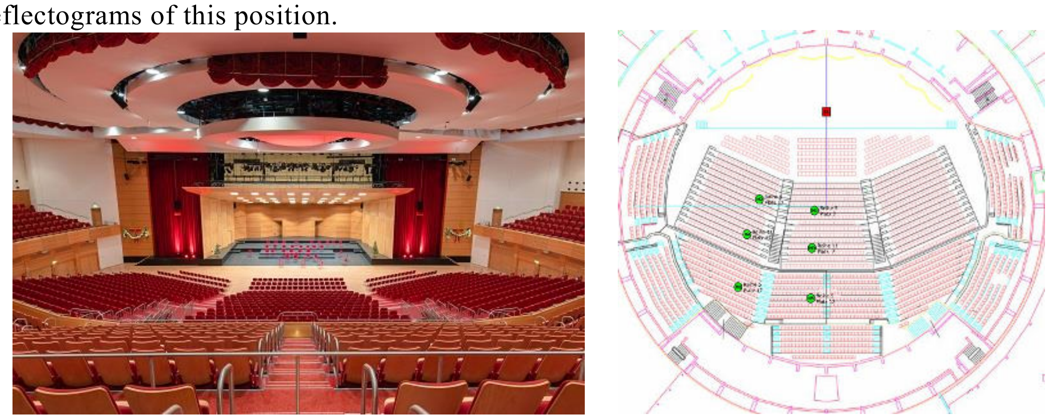

Figure 3 – Picture and layout with measurement positions of CCS

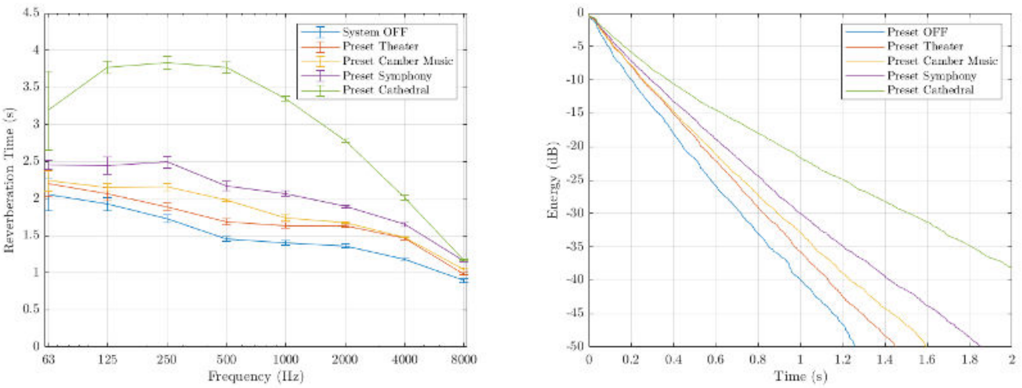

Figure 4 – Position-averaged reverberation time and EDC (broadband, pos. M1) of standard presets (CCS)

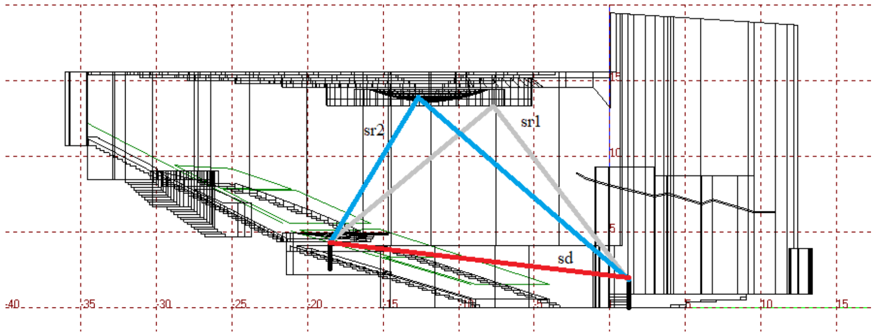

Figure 5 –Likely path of strong reflections from source (left) to receiver (right) (CCS)

Table 1 – Delay of strong, early reflections

Path	Length	Difference in length	Difference in time (c = 343m/s)
sd	20,1m	-	-
sr1	28,7m	8,6m	25ms
sr2	29,5m	9,4m	27ms

3.2 Andermatt Concert Hall

The Andermatt Concert Hall (ACH) provides space for up to 700 visitors and a volume of 5400m³. Due to events like seminars, conferences and concerts of all kinds, variable acoustics are desirable.

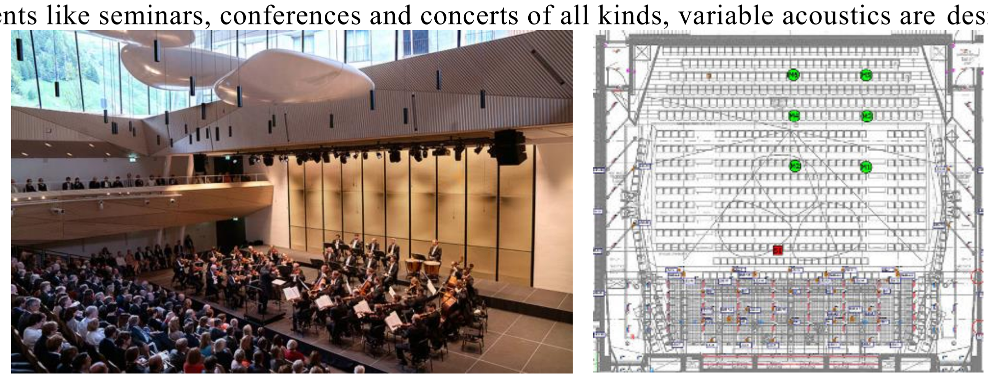

Figure 6 – Picture and layout with measurement positions of ACH

The Amadeus Active Acoustics system was also installed here. The installation consists of 64 loudspeakers and 32 microphones which have been distributed over the audience and stage area. In addition to that 8 subwoofers have been integrated to the system.

In Figure 6 the microphone positions are shown. For this venue three system tunings have been fitted: “Concert half occupied”, “Concert full occupied” and “Cathedral”. The effects of these presets are shown in Figure 7. “Concert half occupied” and “Concert full occupied” are the main settings and should be active at symphonic concerts with full and half occupied auditorium. Figure 7 (left) shows the reverberation time averaged over 6 microphone positions. The intervention of the system is clearly visible. As per Figure 7 (right) the energy decay is shown.

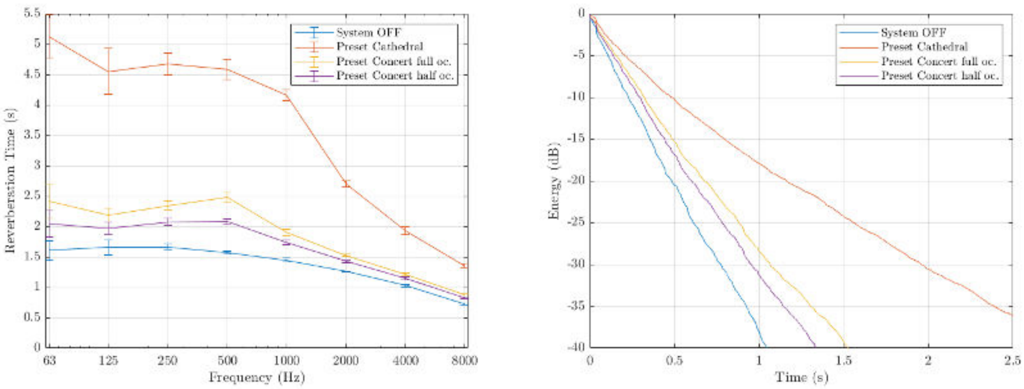

Figure 7 – Position-averaged reverberation time and energy decay (pos. M6) of standard presets (ACH)

4. RESULTS

All measurements have been processed using the ITA-Toolbox from the Institute of Technical Acoustics of the RWTH Aachen University (10,11).

In the first experiment, at Convention Center Suhl, two different approaches were compared. Approach one is to mainly use the regenerative part of Amadeus Active Acoustics which generates reverb by controlled regeneration. Approach two is to mainly use the algorithmic reverberation part of the system. The aim of this task was to tune the system to the same reverberation time for both approaches. In Figure 8 the reverberation time and early decay time of the "Regenerative Approach" (only first stage of the regenerative part) and the "Algorithmic Approach", (only the algorithmic reverberation part with feedback held minimal) averaged across six microphone positions, are compared.

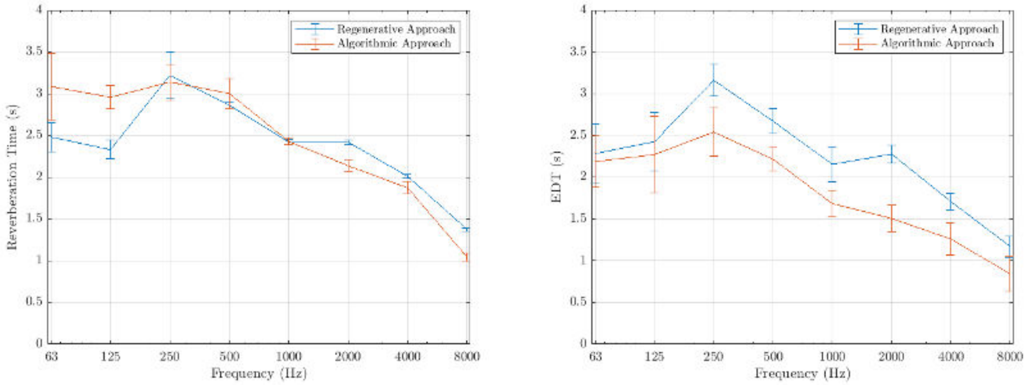

Figure 8 – Position-averaged reverberation times T30 and EDT (CCS)

In mid frequencies the variation of reverberation time T30 is kept similar in reasonable limits. The standard deviation, drawn as vertical lines, overlap for three octave bands. The early decay time on the other hand diverges for frequencies from 250Hz upwards. This result already indicates differences in the curving of the energy decay and thus the perception of reverb.

Figure 9 shows Reflectograms ($p^2(t)$) of both system settings at the same position in the room. Both are normalized to the the peak of the response. A difference in energy due to varying level and distribution is clearly recognizable. In Figure 10 (left) the center time is plotted for both tested approaches of acoustic enhancement averaged over six microphone positions. The regenerative approach produces a significantly longer center time than the algorithmic one. With a difference of 45ms averaged from 500Hz to 1kHz the value is above the Just Noticeable Difference (JND) of 10ms, according to ISO 3382-1. For the clarity index a JND of 1dB according to ISO 3382-1 was exceeded in frequency bands from 250Hz upwards (Figure 9 right).

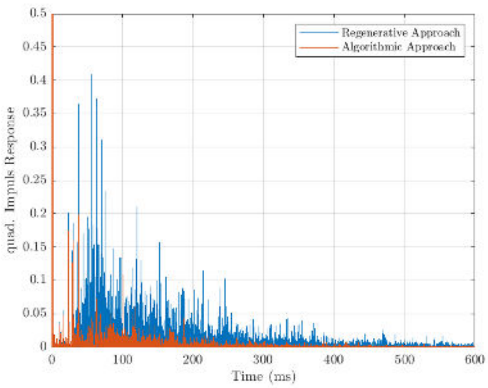

Figure 9 – Reflectograms of both approaches (direct energy scaled to 1, zoomed in to 0.5 and 600ms)

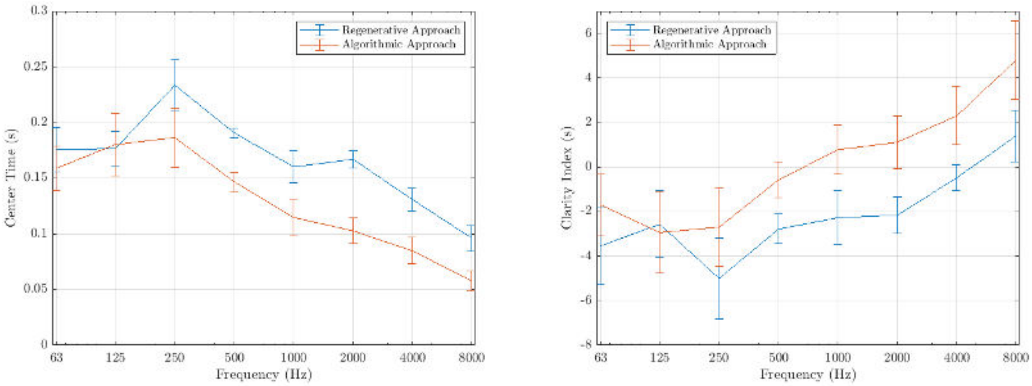

Figure 10 – Position-averaged Center Time (left) and Clarity Index (right) (CCS)

It should be mentioned that the two approaches state quite extreme settings of the system and the room acoustic sound would not be acceptable. Especially the loudness of the presets was very different.

Three more presets have been tested which yield a better sound and also a realistic sound level and reverberation time. The goal here was again to yield the same reverberation time for all three settings but tune the system in a way to change clarity perceptible. In this experiment all system parts (regenerative and algorithmic) were used.

In the following measurements one representable microphone position has been selected for reasons of simplicity. Figure 11 shows the reverberation times (T_{30} and EDT) of Preset 13, 16 and 17.

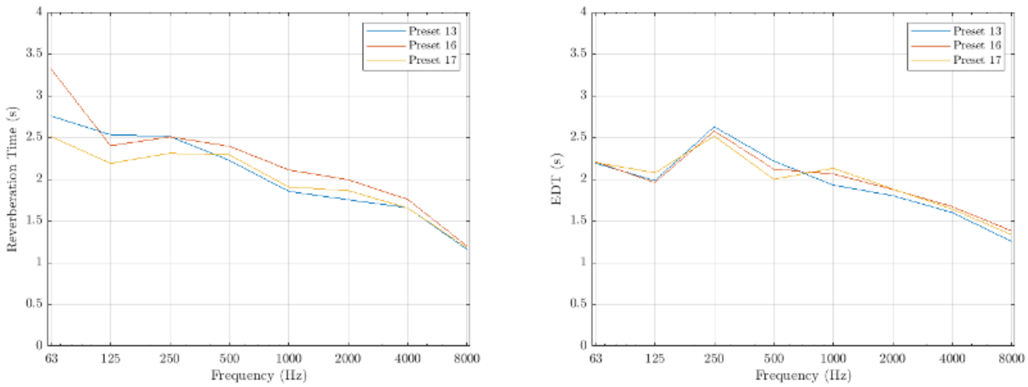

Figure 11 – Reverberation times T_{30} (left) and EDT (right) of position M6 (CCS)

The reverberation times show similar behavior and magnitude. The early decay times are with a maximum difference of 0,22s at 500Hz just noticeable (JND of 5% according to ISO 3382-1 ≙ 0,1s). In Figure 12 the reflectogram of each preset is plotted. While the level of the strongest first reflections changes slightly, the distribution from 50ms to 400ms differs a lot.

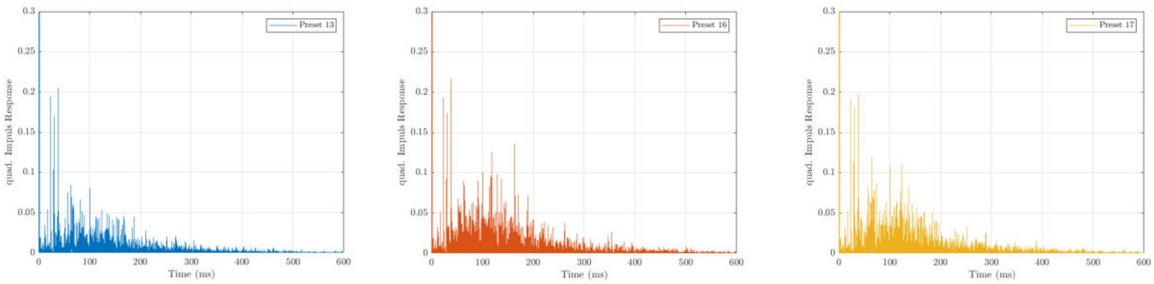

Figure 12 – Reflectograms of Presets 13, 16, and 17 with similar reverberation time (direct energy scaled to 1, zoomed in to 0.5 and 600ms) (CCS)

Adding reflections after 80 ms decreases the clarity index and results in longer center times (Figure 13). Differences in clarity levels (Clarity Index, Figure 13 (right)) of more than 2dB over most octave bands alter the perception of auditory events.

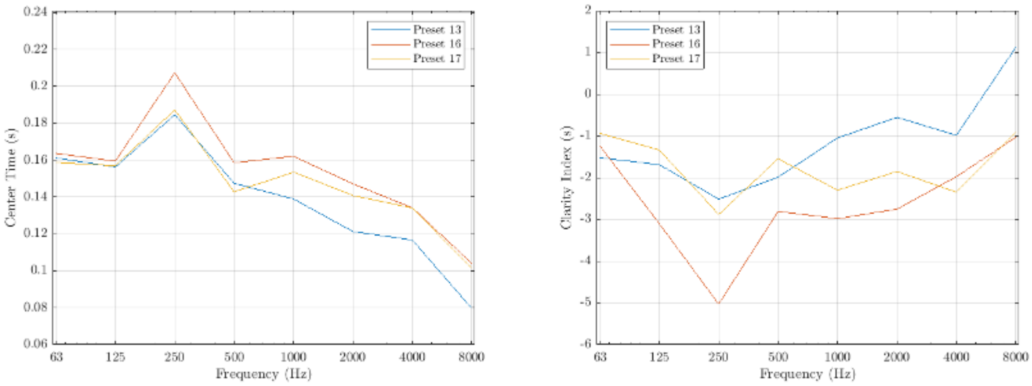

Figure 13 – Center Time (left) and Clarity Index (right) of three tunings with similar reverberation time at position M6 (CCS)

Similar tests have been carried out at Andermatt Concert Hall (ACH). A preset has been tuned to sound comparable to the setting for a full occupied concert (see section 2.3.2) while reaching higher levels of clarity ("Preset Clarity"). In Figure 14 the reverberation times for these tunings are displayed. The new tuning ("Preset Clarity") shows less than 0,2 s difference to the concert preset. The early decay times are for 250Hz upwards almost equal. The early parts of the Reflectograms in Figure 15 overlay (up to 30ms). From about 50ms upward the new preset consistently adds less energy to the room than the concert setting. Therefore, clarity increases. Despite a higher reverberation time and a similar early decay time, "Preset Clarity" reaches a higher Clarity Index over all frequency bands (Figure 14 vs Figure 16 (right)). The center time for the concert preset in Figure 16 (left) is higher for the whole frequency response. Therefore, the energy is shifted to the later part. This correlates with the fact that more early energy improves clarity.

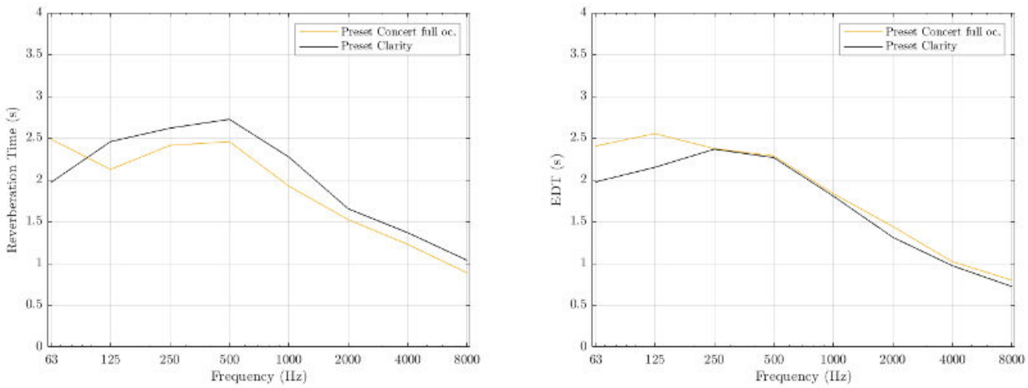

Figure 14 – Reverberation times T30 (left) and EDT (right) with different clarity (ACH)

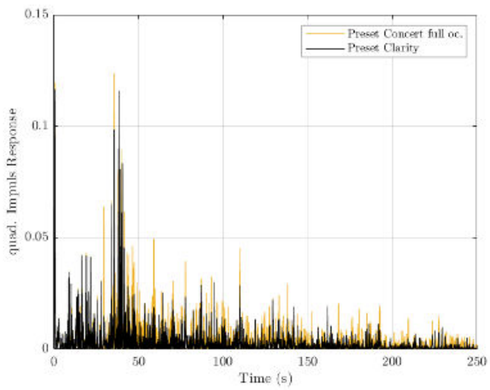

Figure 15 – Impulse response of tunings with different clarity (direct energy scaled to 1, zoomed in to 0.15 and 250ms) (ACH)

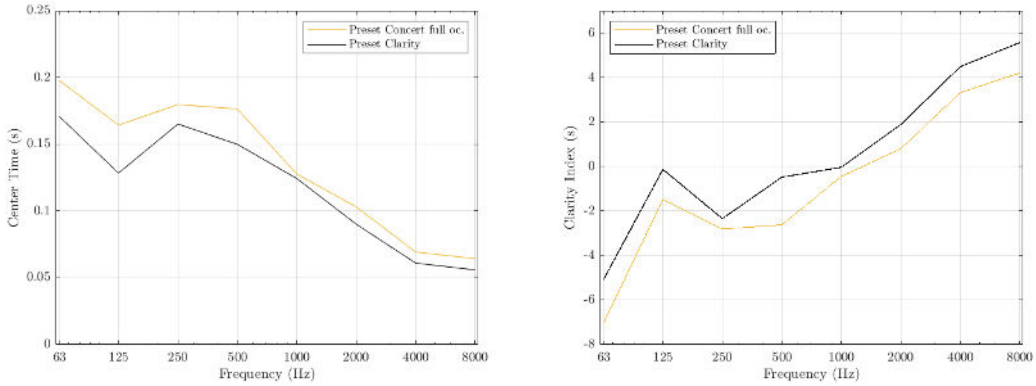

Figure 16 – Center Time (left) and Clarity Index (right) of two tunings (ACH)

5. CONCLUSIONS

This paper investigated the performance of an active acoustic system concerning the tuning of room acoustic parameters. The question was if parameters can be tuned independently within reasonable limits. Therefore, measurements of ISO 3382 parameters have been carried out in two existing installations in rooms having quite different architectural properties. The system parameters in both rooms were tuned yielding similar reverberation times in the room but clearly audible differences in clarity. This could be confirmed by the measurements showing clear differences in clarity index (C80) and center time (t_s).

However, there are limits to do so. An extreme setting using only the algorithmic part of the system can yield a very weak perception of the reverberation making such a preset not feasible. Further, one should never be able to locate a loudspeaker as a part of an active acoustic system. To ensure a homogenic listening experience, enough microphones and loudspeakers have to be distributed around the audience and stage area. More loudspeakers and microphones also enhance the stability and so ensures a listening experience free of coloration and audible feedback. In general, all artefacts of acoustic enhancement systems diminish their purpose. An active acoustic system should not be noticed during an event.

If these limits are considered in the tuning process an active acoustic system represents a tool for the acoustician where not only reverberation times can be altered but also the fine parts of the reverberation can be influenced independently.

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