ACTIVE ACOUSTICS DESIGN USING ROOM SIMULATION

Philip Coleman L-Acoustics, 67 Southwood Lane, London, N6 5EG
Yann Chauvard L-Acoustics, 67 Southwood Lane, London, N6 5EG
Simon Brown L-Acoustics, 67 Southwood Lane, London, N6 5EG
Frederic Roskam L-Acoustics, 67 Southwood Lane, London, N6 5EG

1 INTRODUCTION

Active acoustic systems are an effective tool for optimizing the acoustics of space depending on its required function. A typical use-case is that of a multipurpose performing arts centre, where it is desirable to achieve a high level of acoustic clarity for drama and amplified music (requiring the room to be adequately treated with absorbing material) whilst being able to accommodate performances such as opera and unamplified music which benefit from greater reverberance. With the aid of microphones, electronic processing and loudspeakers, an active acoustics system allows the venue to instantaneously alter the room's acoustic response.

We define the *passive room* as the physical room in which a system is to be installed, and the *active room* as the electroacoustically enhanced space. In active acoustics, the system and space are intertwined such that the passive room influences the sound system design, reverberator design, and eventually the naturalness of the active room. Due to this inherent dependency, active acoustics should be considered early in the conception of a new venue or renovation.

Classically, there are some simple equations that can express the approximate performance that one expects from an active acoustics system as well as best-practice design rules. For a pure regenerative system (i.e., single connection from microphone to loudspeaker), the ratio of reverberation time (RT) for the active room T_{Act} can be approximated for a certain passive room with RT T_{Pas} as [1]

$$\frac{T_{Act}}{T_{Pas}} = 1 + \mu^2 N, \tag{1}$$

where N is the number of channels and μ is the loop gain, which for approximation is typically set in the range of –17 to –20 dB. With the addition of reverberation in the electronic path, greater reverberation times can be achieved. In this case the limiting factor tends to be the linearity of the active room decay. Poletti [2] found that reasonable linearity can be achieved when the RT of electronic reverberator, T_X, matches T_{Pas}. In this case [2]

$$\frac{T_{Act}}{T_{Pas}} = \frac{\Gamma}{1 - \sqrt{(\Gamma - 1)}}, \tag{2}$$

which depends on the power gain provided by a certain number of channels

$$\Gamma = \frac{1}{1 - \mu^2 N}. \tag{3}$$

Increasing T_X beyond this value will lead to more pronounced double-sloping which may be perceived as unnatural [3].

In addition, an underpinning assumption for systems with a regenerative element is that the microphones are in the acoustic far field of the loudspeakers, such that a unitary reverberator

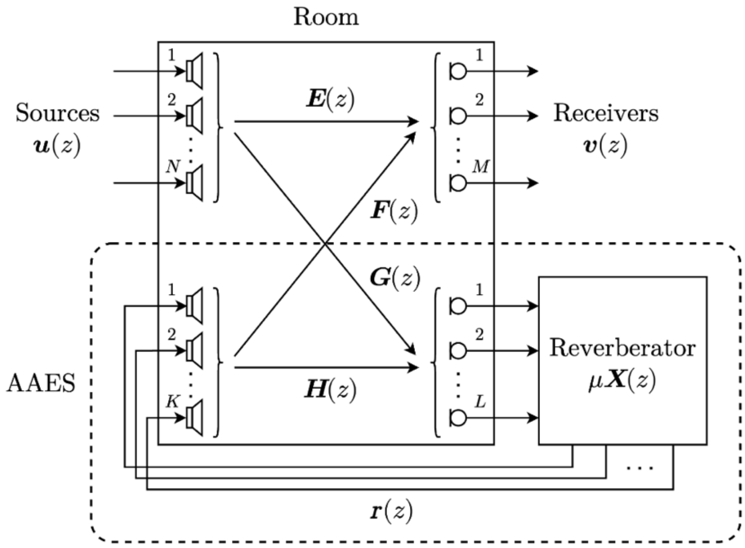


Figure 1 System diagram of an active acoustics system, with the active part marked as AAES. The model comprises the room (physical or simulated), the reverberator the overall loop gain.

structure does not alter the statistics of the feedback loop. It was found in [4] that a direct-to-reverberant ratio DRR of –6 dB is sufficient to achieve the same statistical properties as if there was no direct sound present. This corresponds to a distance of around double the reverberation radius for an omnidirectional microphone, which can be decreased by employing more directional microphones.

The approaches above can guide an acoustic consultant on the overall appropriateness of active acoustics for a certain space and the range of natural reverberation enhancement that may be achieved with a certain number of channels. Nevertheless, at the moment of detailed electroacoustic design, this approach cannot account for practical aspects such as loudspeaker and microphone directivities, constraints on rigging points, or acoustic artefacts specific to a specific passive room.

Workflows to simulate the response of a sound system in a room are well defined for sound reinforcement design, especially in challenging spaces. Such workflows can be extended with knowledge of the active acoustics processing to simulate the active room response based on an acoustic simulation of the system in the passive room. Such an approach allows the consultant to:

- Compute acoustic metrics at various locations within a certain room;
- Listen to the active simulation as an aural sketch of what active acoustics can achieve;
- Fully understand the system design, including the actual positions of the microphones and the directivity of the specific loudspeakers intended for the installation;
- Refine the passive design to support the electroacoustic installation and achieve the best result.

In this paper we will investigate the feasibility of an active acoustics prediction and simulation approach. First, we experimentally validate a classical closed form solution for active acoustics prediction; then we present a simple case study to predict the active room response from simulated room impulse responses.

2 THEORY

An active acoustics system comprises several acoustic paths and electronic signal paths, as illustrated in Figure 1. The direct paths from the performer(s) $\boldsymbol{u}(z)$ to the audience $\boldsymbol{v}(z)$ exist whether or not the active system is switched on, and is always the least reverberant condition. The system's microphones also capture the performance through the room $\boldsymbol{G}(z)$, and the loudspeakers emit sound both to the audience $\boldsymbol{F}(z)$ and back to the system microphones $\boldsymbol{H}(z)$, creating a feedback loop. The characteristics of the active room, and the feedback system, are governed by the combination of the acoustic feedback path and the electronic processing applied on the microphone signals, as well as the overall gain of this processing μ. Typically, the *open loop matrix* $\mu\boldsymbol{X}(z)\boldsymbol{H}(z)$ can be measured and analyzed to understand the gain before instability of the feedback system and its resonant frequencies. Poletti [5] showed that a frequency-domain closed-form solution exists to combine the various acoustic and electronic paths to predict the closed-loop response from the individual open loop measurements,

$$\boldsymbol{v}(z) = \boldsymbol{E}(z)\boldsymbol{u}(z) + \boldsymbol{F}(z)[\boldsymbol{I} - \mu\boldsymbol{X}(z)\boldsymbol{H}(z)]^{-1}\mu\boldsymbol{X}(z)\boldsymbol{G}(z)\boldsymbol{u}(z). \tag{4}$$

The first term of the summation describes the passive system, while the second term describes the contribution of the active system. In principle, this equation is a powerful tool which allows us to predict the influence of room acoustics and reverberator design on the closed-loop active response, without needing to be physically present at the installation.

3 VALIDATION MEASUREMENTS

To understand the utility of the prediction approach, it was important first to validate the prediction equation. We implemented an active acoustics system in L-Acoustics' Immersive Lab, enabling us to measure the transfer paths in both open-loop and closed-loop configurations. The space is approximately 20 m x 8 m, and has a pitched roof with maximum height approximately 7 m from the floor. The space has retractable curtains which allow the RT to vary from 0.6 s with the curtains fully deployed to 0.9 s with the curtains fully retracted. The system has 16 microphones and 27 loudspeakers. An omnidirectional sound source was placed in the room to represent a source on stage, while two omnidirectional microphones were used to sample the audience area; one close to the omnidirectional source and one in the far field. The curtains were fully retracted for the experimental results presented in this paper.

We first followed our usual procedure to calibrate the active acoustics system. The electronic processing uses a 346 crosspoint gain-delay matrix, with other reverberation processing disabled. The procedure calculates equalization for each microphone channel to ensure stability and colouration-free regeneration. Then, we measured the room impulse response (RIR) from the stage source to the audience receivers using the swept sine method to capture the real-world performance of the closed-loop system.

For the prediction, we measured the open loop acoustic transfer paths from all loudspeakers to all microphones with the microphone equalization active. We separately measured the impulse response of the gain delay matrix. Finally, we combined the measured RIR matrices in the frequency domain to compute the predicted open-loop response according to Equation 4.

Figure 2 shows an example of the energy decay curves in octave bands. Generally, there is very good agreement between the curves. Some measurement noise is evident in the lower octave bands, and in some of the mid-range bands we can also observe that the noise floor differs between the prediction and the measurement, while the decay itself is very similar.

Two further illustrations of the validation performance are presented in Figure 3. Figure 3 (a) shows the time domain impulse responses at each microphone. We observe that the structure of the impulse responses match quite well. The visibility of the red predicted curve behind the black validation measurement implies a slight over-prediction of the amplitude. Figure 3 (b) shows the percentage difference in reverberation time (T20). The results are presented as a boxplot summarizing the variation across nine combinations of repeat measurements (three each for the prediction RIRs and validation measurements). It is worth noting that there is indeed a distribution across the nine variations, and as such one should be cautious to make conclusions based on a single measurement. Moreover, while using the identical procedure for the two microphones, we do observe a difference based on the measurement location. Figure 3(b) confirms that there is generally a modest over-prediction of the active system’s energy. The difference in RT is around 5% above the 1 kHz band, which is around the just-noticeable-difference. At lower frequencies the errors are larger in some

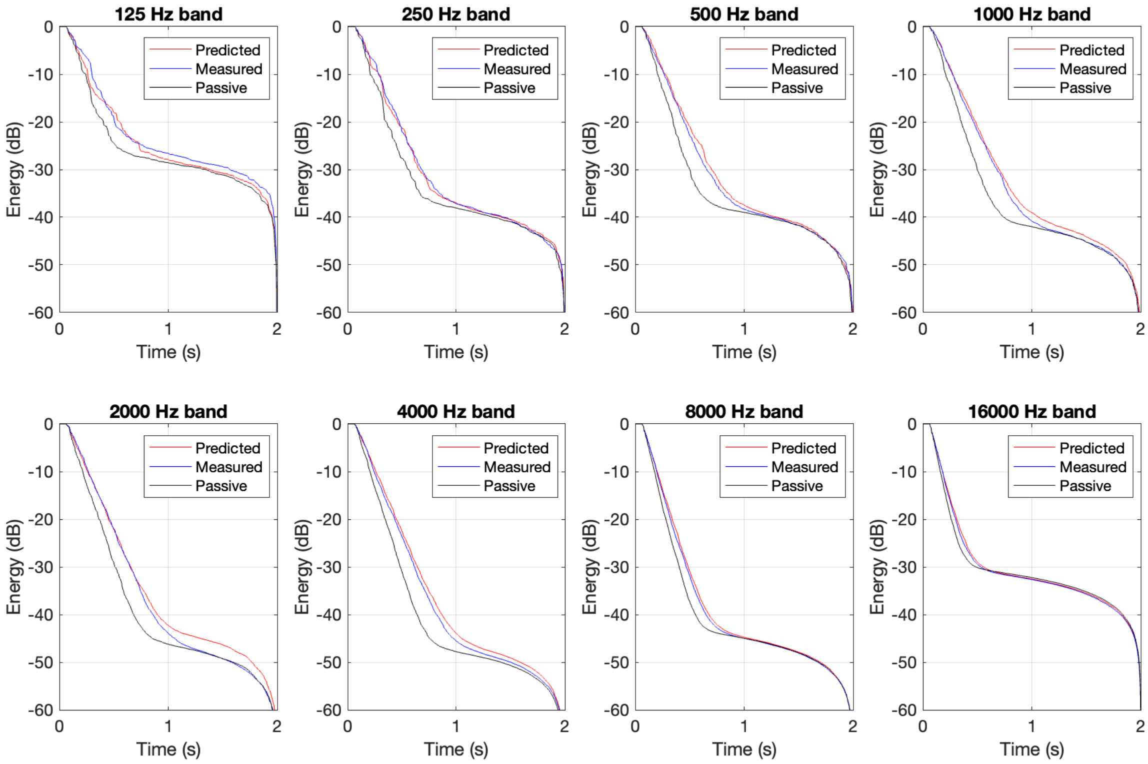
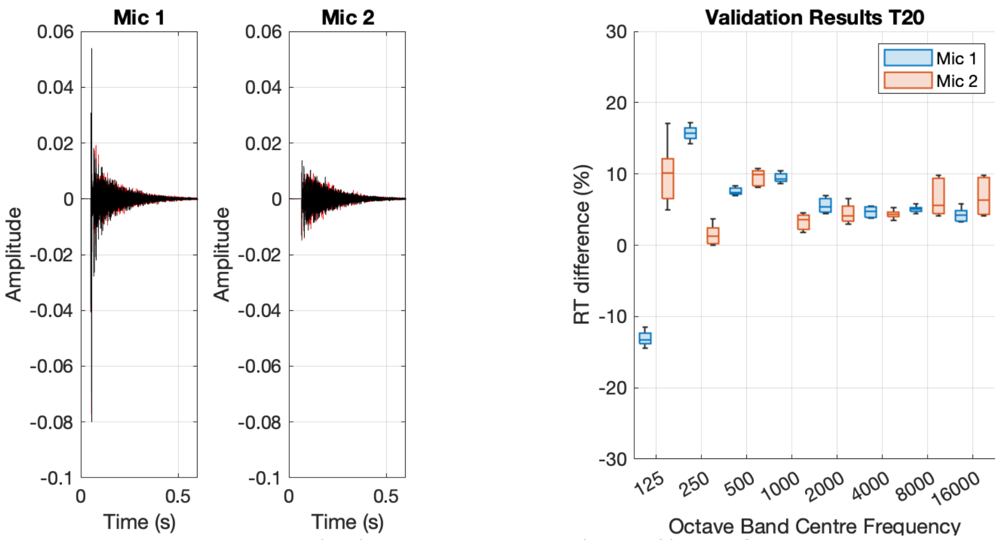


Figure 2 Predicted (red) and measured (blue) octave band energy decay curves, compared to the passive room decay (black).



a) Time domain IR, predicted (red) and measured (black) for the two microphone positions.

b) RT % error for each microphone, with distributions across repeat measurement combinations.

Figure 3 Validation performance showing a) time domain and b) RT error.

cases, although we have checked the energy decay curves and suspect that RT differences might be due to noise in the measurements.

Overall, the validation measurements demonstrate the value of predicting the closed-loop measurement from its open-loop components. In practical terms, it might be prudent to slightly reduce the loop gain in prediction to avoid over-predicting the RT.

4 SIMULATION-BASED DESIGN EXAMPLE

Having established that the proposed open-loop to closed-loop prediction works in practice, we can exploit the prediction workflow to replace measured transfer paths with simulated ones. This gives full flexibility to investigate the influence of system design (e.g., the number of transducers and their type, position, and orientation) and room design (e.g., geometry and absorption). Moreover, it presents opportunities for acoustic consultants and sound system designers to undertake detailed design work side-by-side well ahead of the system commissioning.

In our experiments, we have modelled the Immersive Lab in the Treble[1] web app. Treble is a cloud-based acoustic simulation tool comprising a hybrid geometrical acoustics solver and wave-based solver. In our simulation, we used the hybrid solver with a cutoff frequency of 360 Hz. The directivity of the L-Acoustics X8 sources was modelled in the simulation, while all receivers were modelled as 2nd order Ambisonic IRs. Where relevant, microphone directivity was simulated by post-processing the Ambisonic IR with an idealized cardioid. No attempt was made to match the in-situ tuning of the loudspeaker system, nor to perfectly replicate the stage source or audience receiver positions.

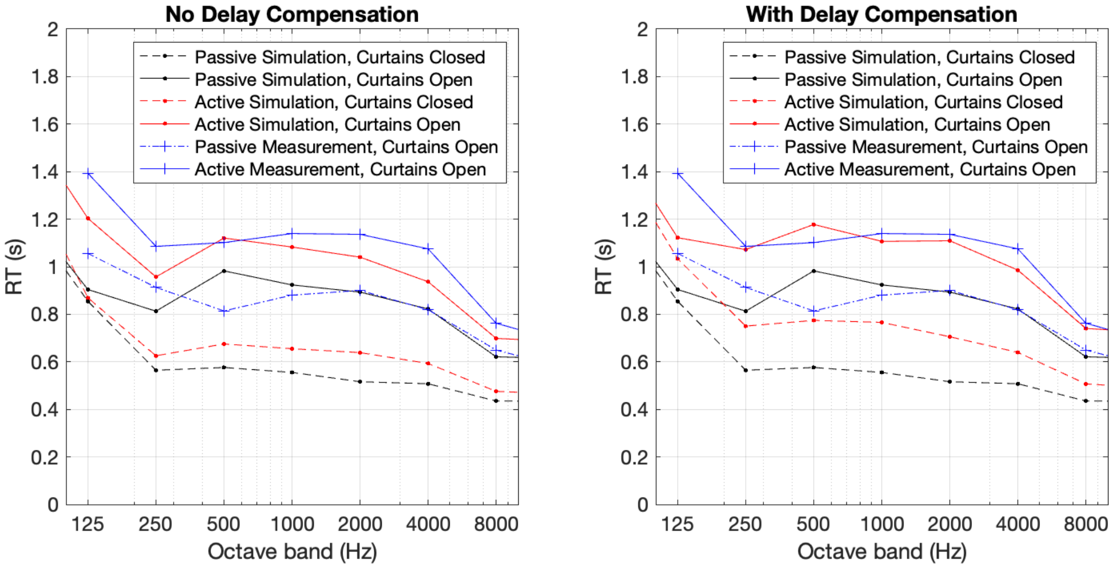
Before the curtains were installed, we used the Treble simulations to predict the acoustic damping that we would expect in the room with the curtains closed. Using the workflow proposed in this paper, we can extend the analysis to predict the active system's behaviour with the curtains closed. Therefore, two versions of the acoustic model were produced, modelling the wall absorption with the curtains drawn and curtains retracted, respectively. As above, the simulated RIRs were fed into the prediction software with the identical gain-delay matrix used in Section 3. This time, as no reference tuning was available, we auto-computed equalization filters using our calibration tools. The loop gain was again set to –4 dB from instability.

The initial results are shown in Figure 4(a). From the figure, we can first compare the solid black line to the dashed black line to observe the change in the simulated passive RT when the curtains are closed. By design, to facilitate playback of amplified music in the space, the RT is reduced to under 0.6 s above the 250 Hz octave band. The red lines show the simulated effect of the active acoustics system. Comparing the Treble model of the lab with the curtains retracted (black line) with a real measurement of the passive room in the same conditions (blue line), we observe generally a good match. However, we noticed that the simulated active room did not reach the same RT as the real-world system that we implemented.

One difference of the Treble model compared to the true active acoustics system concerns the processing latency of the microphone preamplifiers, bus processing, digital network transport latency, and latency introduced in the amplifiers. Instead, the Treble model contains only the acoustic latency from source to receiver. We accordingly added some pre-silence to the simulated RIRs to model this processing latency. The RT is shown in Figure 4(b), where only the red lines differ from Figure 4(a). With the additional delay, we see that the system is able to provide more regeneration for the same safety margin, and the simulated active response matches fairly well with the measured response.

As expected, for the same electronic signal processing and differing room absorption, we observe a reduced capacity to extend the RT when the passive room is damped. This follows from adopting a simple gain-delay matrix as a reverberator, which relies on regeneration of the passive room acoustics as its primary mechanism to extend the RT. With the simulation approach, we could analyse the effects of introducing additional reverberation processing to try to bring the dashed red line towards the solid black line, i.e., using the active system to restore the original RT. Alternatively, we could model an increased number of channels to achieve the same goal.

[1] https://www.treble.tech



a) Without delay compensation   b) With delay compensation

Figure 4 Active RT predictions based on Treble simulations for room configurations with curtains open (solid lines) and closed (dashed lines), together with the corresponding measurements (blue lines).

5 CONCLUSIONS

In this paper, we have explored the use of acoustic simulations to predict the closed-loop active acoustics response at positions in the audience area. Compared to design-rule-based approaches, this allows us to obtain a detailed acoustical prediction which can be analysed with acoustic metrics and even listened to by auralization. We first validated the classical prediction equation, obtaining a reasonable agreement between the predicted and measured closed-loop RIRs. We did find that the results depended on the microphone position and combination of repeat measurements used, and that a slight over-prediction of RT was typical. This is likely due to differences in measurement noise and the slight time-variation present in the real-world case.

Having established the basic utility of the prediction equation, we compared the active acoustics performance in a room with two simulated passive acoustic conditions. We found that the simulations predicted the expected change in performance with the introduction of more absorption, although it was important to introduce the system processing latency into the model to obtain results more consistent with the real-world measurements.

Overall, we believe that using acoustic simulations for active acoustic design will be valuable in two main ways. Firstly, simulation gives a good means for comparative studies including passive acoustic design changes, system design options, and electronic processing approaches. Second, simulation allows more detailed spatial and temporal analysis of the likely active system performance, and the opportunity to listen to the result. Of course, no room simulation is perfectly accurate, and so such use-cases should only ever be taken as an approximation of how the system performs in situ. However, we still believe that the approach brings value to help acoustic consultants and system designers explain the opportunities and indicative performance of active systems.

6 REFERENCES

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