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Evaluation of a real-time auralization system for a 42-loudspeakers array

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ABSTRACT

Auralization is the process of simulating the auditory perception of a specific environment or sound source. When processed in real-time, it enables participants to be immersed in a virtual acoustic environment. However, when using a loudspeaker array for real-time auralization, there is a risk of unintended acoustic feedback which can distort the desired sound output. To address this issue, various methods have been proposed in the literature leading to various efficiency. In this paper, one selected method, a calibration-based feedback cancellation system was implemented and objectively evaluated both for a monophonic setting and for a 42-loudspeaker array dedicated to spatial sound diffusion. A feedback reduction of more than 25 dB was observed for each individual loudspeakers, and 27dB for the 42-loudspeakers system. Spatial Room Impulse Response were measured inside three rooms and measured a second time inside the auralization system. Reverberation Time and Clarity of both measurements were similar.

1 Introduction

Auralization is the process of simulating the auditory perception of a specific environment or sound source [1]. When processed in real-time, it enables participants to be immersed in a virtual acoustic environment [2], [3]. However, when using a loudspeaker array for real-time auralization, there is a risk of unintended acoustic feedback which can distort the desired sound output. To address this feedback cancelling issue, various methods have been proposed in the literature leading to various efficiency [4]. These methods could be grouped into four categories: phase modulation, gain reduction, spatial filtering, or room modeling.

The intended application of our works involves evaluating musician interpretation according to room acoustics within a research framework. This kind of study could be realized directly inside room to be tested [5], [6], but it requires to the participant to move from place to place and makes impossible to compare distant sites. To evaluate this influence, the auralization method has been preferred. Investigating acoustic environment on musician interpretation is not new: as an example previous studies [2], [7] aimed to realize the same kind of experiments. However, they used microphone in proximity to the musician to avoid acoustic feedback. This configuration requires a specific calibration for each instrument and could be considered as intrusive by the participants. For this reason, we choose to implement an anti-feedback system with fixed microphone outside the performance area. Our goals require precise control of diffused signals and immediate treatment of the feedback. That is why a calibration-based feedback cancellation system was implemented, which could be considered as part of

room modeling methods. The placement and properties of the microphone could also have an influence on the performance: the spatial filtering is one of the methods among one presented in the literature. The spatial filtering could be induced by the directivity of the microphone, and/or the distance to the loudspeakers. Our hypothesis is that the best situation for the microphone is to maximize the energy from the source to auralize, and to minimize the energy from the loudspeakers. However, using a loudspeaker array and letting free the central zone, it is impossible to minimize the distance to the microphone for the whole loudspeakers, and using a directivity pointing only.

Evaluation of anti-feedback performances were also addressed in the literature. The main evaluation criteria could be grouped into three categories [4], [8], [9], [10]: the added stable gain (ASG), the audio quality and the reactivity. The ASG is the maximal gain before the system being unstable. The most application for anti-feedback system concern the sound reinforcement, especially for public address systems. In this work, the intended application was different because the system aimed to reproduce successive echoes. Echoes should be distributed in a similar way to real case. Then, evaluation of the system could differ from usual system dedicated to sound reinforcement.

In this paper, an objective evaluation of an anti-feedback system based on a calibration is presented. The evaluation was conducted both for a monophonic and a multichannel setting. The paper is organized as follow. First, the experimental setup is presented, including the implementation of the anti-feedback system and the evaluation process. Then, results for

both a monophonic and a multichannel setup are presented.

2 Experimental setup

Setup was composed of a 42-loudspeaker spherical array of 2m radius placed inside a semi anechoic chamber as used in [11]. Loudspeakers were Genelec 8020 and auralization microphone was a cardioid Rode NT5 placed on the spherical structure, as illustrated in the Figure 1. The computer was an Apple MacPro Intel Xeon 8-core, and the audio interface was a dante Focusrite RedNet PCIe. Buffer size was set to 64 sample at the sample frequency of 48 kHz.



Figure 1: Part of the 42-loudspeakers spherical array, and auralization microphone on the center.

The canceler system implemented in this study was based on the implementation proposed in a previous study [12]. For a monophonic setup, the general synoptic is presented on the Figure 2 and implemented in Max/MSP. This implementation required a canceler design, processed with slight differences from [12], as follow. The transfer function between the loudspeaker and the microphone was measured using Matlab. A 10-s logarithm sine sweep from 20 Hz to 22 kHz was used for this measurement. The impulse response obtained with this measurement was windowed and first samples corresponding to the buffer size were removed to obtain a total length of 0.3s.

The evaluation was conducted using both a control loudspeaker placed 40cm above the center of the system and a control microphone placed at the center of the system as illustrated in the Figure 3.

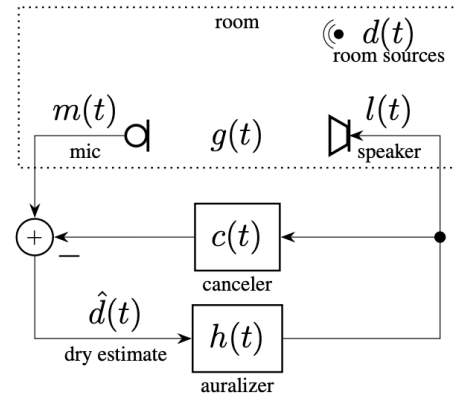


Figure 2: General synoptic for a monophonic setup, from [12].



Figure 3: Evaluation setup composed of control loudspeaker Genelec 8020 and a control microphone mh acoustics em 32 (for the multichannel evaluation).

The evaluation was conducted on two parts:

A) Delayed impulse

First, the auralization filter consisted of a single 10-second delayed impulse and the control signal transmitted through the control loudspeaker was a white noise lasting 9 seconds. The analysis was centered on the timeframe from 20 to 29 seconds. Ideally for a perfect system, the control microphone would capture no sound during this period. The evaluation aimed to quantify the residual sound level within this timeframe. A Performance Gain G is computed as follow:

$$G = \frac{\sqrt{\frac{1}{N} \sum_{t=20}^{29} x_{without}^2(t)}}{\sqrt{\frac{1}{N} \sum_{t=20}^{29} x_{with}^2(t)}}$$

where N is the number of samples, $x_{without}$ the measured signal without anti-feedback, x_{with} the signal using anti-feedback.

For this measurement, the control microphone was a Rode NT5 with omnidirectional membrane NT45-O. The loudspeakers of the spherical array were first assessed separately and then all together. When evaluate the 42 loudspeakers all together, the auralization filter consisted into a 5th order ambisonic impulse response. All the 36 channels were zeros, excepted the first (omnidirectional component) which was a 10s delayed impulse.

B) Measured SRIR

The second step of the evaluation was made using three measured Spatial Room Impulse Responses (SRIR) as auralization filters, only for the multichannel setting. The rooms used for this evaluation were varied in size and reverberation times as reported in Table 1.

Two of them were medium-sized chapels localized in Peyrolles and Bras. The third one was the Great Chapel of the Palais des Papes in Avignon. Measurement protocol of the first two is presented in [13], and the same protocol was used inside the great Chapel of Avignon. These SRIR were measured using the mh acoustics Em32, a spherical microphone array allowing to capture sounds at 4th order ambisonics. A Genelec 8020 was placed 40cm above the microphone as illustrated with Figure 4. The direct path was removed by windowing the beginning of the impulse response.

These SRIR were then measured a second time inside the auralization system, using the same measurement setup than inside the real rooms. Both sets of measurements named “*insitu*” for initial measurements, and “*inlab*” for second measurement were used to computed two room acoustic descriptors across 24 third-octave bands, centered from 50 Hz to 20 kHz. The descriptors were the Reverberation Time (RT60) and the Clarity C80.

Table 1: Name, Volume and Reverberation Time of the three tested rooms

Name	Volume (m ³)	RT _{1kHz} (s)
Peyrolles	550	0.9
Bras	280	3.6
Avignon	12000	8.5



Figure 4: Measurement protocol inside the chapels.

3 Result

A) Delayed impulse

First, a single loudspeaker placed at 1m distance of the auralization microphone was considered. The amplitude of the impulse, the auralization filter, was adjusted in such a way the gain of the delayed noise measured with the control microphone was close to the gain of the direct noise. The temporal gain of measured signals with and without anti-feedback is represented in the Figure 5.

Signals were very similar during the first 20s, with a slight difference around 10s and 20s: the noise floor is slightly higher without anti-feedback. Between 20s and 30s, amplitude of the signal without anti-feedback is around -10dB, and around -40dB with anti-feedback.

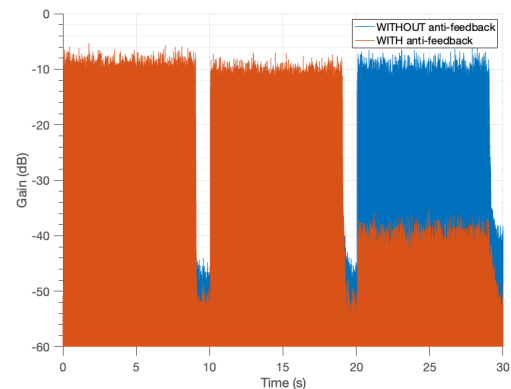


Figure 5: Temporal gain with and without anti-feedback.

The Performance Gain was then computed for this specific measurement and four other loudspeakers of the spherical array. Performance Gain for these measurements is represented on the Figure 6

according to the distance to the auralization microphone. The Performance Gain was always superior to 25 dB, reached a maximum of 29.5 dB for at 1m and decreased for shorter and longer distances. Measurement using the whole 42-loudspeakers led to a performance gain of 27.1 dB.

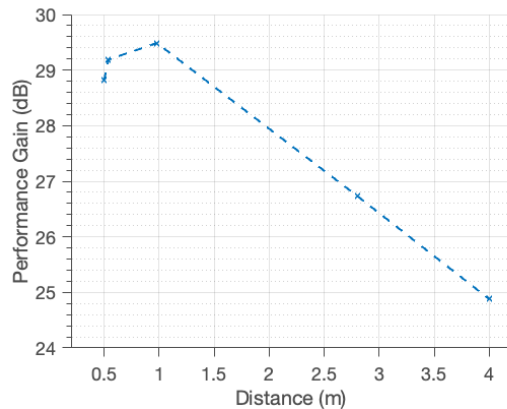


Figure 6: Performance Gain of the anti-feedback system for five loudspeakers placed at distance of the auralization microphone from 0.5m to 4m

B) Measured SRIR

Acoustics descriptors RT60 and C80 computed from measurements *insitu* and *inlab* are represented in the Figure 7.

For Peyrolles, the less reverberant room, RT60 were equivalent at ± 0.2 s for the two conditions between 150 Hz and 10 kHz. At low frequencies, *inlab* RT60 were higher than *insitu* RT60. The highest difference stood for 100 Hz, where RT60 was 2.1s *inlab* and was 1.3s *insitu*. C80 were equivalent at ± 6 dB between 100 Hz and 10 kHz.

For Bras, the *inlab* RT60 was always superior to the *insitu* RT60. The difference was mainly inferior to 0.5s excepted at 630 Hz and below 160 Hz where the difference was more than 1s. C80 were equivalent at ± 5 dB between 100 Hz and 10 kHz.

For Avignon, RT60 were equivalent at ± 0.5 s between 100 Hz and 700 Hz. The *insitu* RT60 was superior to 1s between 700 Hz and 4 kHz. Below 100 Hz, the *inlab* RT60 was inferior to the *insitu* RT60: at 60 Hz, *inlab* RT60 was 10s whereas *insitu* RT60 was 14s.

C80 were equivalent at ± 4 dB between 100 Hz and 10 kHz.

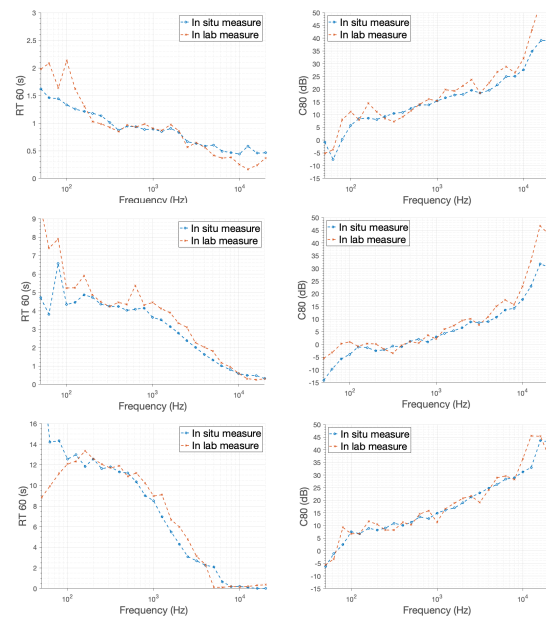


Figure 7: Acoustic descriptors of the three rooms. Left: RT60; Right: C80. From top to bottom: Peyrolles, Bras, Avignon

4 Discussions

Results regarding the delayed impulse were very promising, with a gain of at least 25dB. As can be seen in Figure 5, when using anti-feedback system, residual noise was close to the background noise level. However, the system was not perfect as there is always some residual noise present. Measurements at different distances showed peak performance at 1m, and a decrease for higher distances. This contradicts our initial hypothesis, which assumed that increasing the distance would simplify the problem. However, the measurement distance does not encapsulate all experimental conditions. The measurements were conducted within a 42-loudspeakers spherical array, and the angle between each loudspeaker and the microphone was different for each tested loudspeaker. The directivity of the loudspeaker and the microphone could influence these results. To definitively conclude, additional measurements should be conducted along the same axis between the microphone and speaker. The combination of the 42 loudspeakers gave similar results than individual loudspeakers.

Acoustic descriptors measurements revealed promising trends, even though the results were not identical from initial measurements. For the least reverberant room, the reverberation time values are nearly identical and may fall within the measurement

variability of a real room. The differences were slightly greater for more reverberant rooms, but tendency were similar. Auralization was still effective for highly reverberant spaces without the system diverging.

5 Conclusions

In this paper, an anti-feedback system was implemented for a 42-loudspeaker array placed inside a semi anechoic room. Performance of this system were evaluated for separate loudspeakers, and for the whole system. A Performance Gain of more than 25 dB was observed for all the individual loudspeaker and 27 dB for the whole system. Contrary to initial hypothesis, we noticed a decrease of performance with the increase of distance beyond 1m. The comparison of acoustic descriptors computed from Spatial Room Impulse Response measured *insitu* and *inlab* revealed encouraging result. For the room with little reverberation (RT60=0.9s at 1kHz), RT60 were equivalent at +/- 0.2s in middle frequencies. For the rooms with high reverberation times, differences in RT60 were inferior to 0.5s for most of frequencies but could be superior to 1s at specific frequencies.

As a perspective of these works, a multiple microphones anti-feedback system could be implemented. It could make possible to consider the directivity of the sound source to auralize. Moreover, this also opens up the possibility to take into account the anisotropy of the acoustic of the room.

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