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## Recording in a Virtual Acoustic Environment

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### ABSTRACT

A method is presented for high-quality recording of voice and acoustic instruments in loudspeaker-generated virtual acoustics. Auralization systems typically employ close micing to avoid feedback, while classical recording methods prefer high-quality room microphones to capture the instruments integrated with the space. Popular music production records dry tracks, and applies reverberation after primary edits are complete. Here, a hybrid approach is taken, using close mics to produce real-time, loudspeaker-projected virtual acoustics, and room microphones to capture a balanced, natural sound. The known loudspeaker signals are then used to cancel the virtual acoustics from the room microphone tracks, providing a set of relatively dry tracks for use in editing and post-production. Example recordings of signing in a virtual Hagia Sophia are described.

### 1 Introduction

Advances in signal processing and acoustics measurement have made the synthesis of real-time virtual acoustics possible, allowing rehearsal, performance, and recording in the acoustics of inaccessible or no longer extant spaces. For recording, even for existing, accessible spaces, studio-based systems have benefits over on-site recording environments, including reduced location noise and simpler logistics, such as access to equipment and power.

Auralization systems process sound sources according to impulse responses of the desired acoustic space [1, 2]. They often use close mics or contact mics for acoustic instruments and voice to avoid feedback. Doing so, while adequate for generating virtual acoustics, does not capture sound with sufficient quality for music recording, mainly due to the microphone proximity to

the sources. In this work, we consider the problem of high-quality recording in a virtual acoustic environment. We focus on voice and acoustic instruments, and describe a method that borrows from classical and popular music recording methods, and allows sound source positioning and room acoustics manipulation in post production or interactively, for instance in a virtual reality setting.

Real-time processing is needed to capture musician interactions with the virtual space; musicians will adjust their tempo and phrasing, alter pitch glides, and seek out building resonances in response to the acoustics of the space [3–6]. For instruments not affected by acoustic feedback, the instrument signal may be processed and played over loudspeakers, providing real-time auralization. The Virtual Haydn project [7] took this approach to record keyboard performances in

nine synthesized spaces. For acoustic instruments and voice, virtual acoustics systems include (a) close miced singers and either loudspeakers or headphones [8, 9], or (b) many microphones and loudspeakers installed in the space, with processing tuned to the desired acoustics [10, 11].

Presenting virtual acoustics over headphones allows high-quality room mics to be placed away from the musicians, while still capturing dry signals. The problem is that headphones impair the performers' ability to hear and interact with one other. For auralization systems presenting virtual acoustics over loudspeakers, one approach to recording is to place microphones about the hall or studio, as if the hall were generating the acoustics heard [12–15]. A drawback to this approach is that the recording locations are fixed, and the captured reverberation is not easily adjusted or made interactive. Another approach is to use the close mic tracks driving the auralization. However, this approach has the drawback that close mics capture a poor balance of the radiated sound from the instrument or voice and pick up unwanted source sounds.

Here, signals from close mics and contact mics are processed to produce loudspeaker-projected virtual acoustics, and high-quality microphones are placed about the performers and the space. The known loudspeaker signals are then used to cancel the virtual acoustics from the room microphone recordings, providing a set of high-quality, relatively dry tracks to use in editing and post production. In this way, the musicians are performing in and fully interacting with the virtual acoustic space, while dry tracks are recorded with high-quality microphones.

The dry room mic tracks are used to do primary edits, and a combination of the edited dry room mic and close mic tracks are used to synthesize virtual acoustics and to position the sources within the space while mixing or in post production. This facilitates editing between takes and affords the producer more options to modify the auralization after tracking.

The cancellation method described here is similar to that of the adaptive noise cancellation approach developed by Widrow [16], in which a primary signal is the sum of a desired signal and unwanted noise. In the approach, a reference signal, which is correlated with the unwanted noise, is used to estimate and subtract the unwanted noise from the primary signal. Related

literature also includes echo cancellation and dereverberation [17–19]. These approaches are often aimed at improving speech intelligibility, and produce artifacts that are undesirable in a recording context.

To cancel the loudspeaker-produced virtual acoustics from the room microphones, the live room is configured so that there is an unobstructed (or at least unchanging) first arrival between each of the loudspeakers and room microphones. Microphone polar patterns can also be selected to favor performer positions over loudspeaker positions. A number of impulse response measurements are made between each of the loudspeakers and room microphones. The loudspeaker signals are then processed according to the impulse response measurements, and the processed signals subtracted from the room microphone signals so as to cancel the virtual acoustics from the room microphone tracks.

The idea behind collecting a number of impulse responses between each loudspeaker-microphone pair is that certain time-frequency regions of the impulse response can vary over time, for instance with different positioning of the musicians. For example, we have seen that the onset of the impulse response is expected to be more stable than the tail. When canceling the loudspeaker signal from the microphone signals, the cancellation is eased in the portions of an impulse response showing greater time variation so as to minimize the loudspeaker energy present in the canceled room mic signals.

In the following, we describe details of the recording system architecture, § 2, and cancellation processing and performance analysis, § 3. In addition, example recordings in a virtual Hagia Sophia acoustic are discussed, § 4.

## 2 Recording in Virtual Acoustics

We begin by proposing a methodology for recording in a simulated acoustic environment. Our goals include capturing high-quality recordings, providing a comfortable and convincing auralization for the musicians, and producing flexibility throughout the editing and mixing processes. Previous work has shown that musical performances are effected by room acoustics. Tuning, timing, and other performance attributes are mediated through acoustic spaces and must be considered throughout the recording process. Because of this fact, musicians must perform live, interacting with the auralization of the virtual space in order to record them.

## 2.1 Recording Approach

For the musicians to perform in a virtual space, we mic each musician and use real-time convolution reverberation to produce the auralization. Musicians tend to prefer to not wear headphones while performing so they can better interact with one another. Because of this, we provide the auralization over loudspeakers. We use close-micing techniques to feed the auralization system so we can separately control the relative level and equalization of each musician, and avoid feedback between the microphones and the loudspeakers.

While this is sufficient for a real-time performance system, it leaves much to be desired for high quality recording. Contact microphones and close-micing techniques capture an unnatural perspective for most instruments and voice. In general, we do not listen to musical instruments from close ranges and placing a microphone close to an instrument will record an unfamiliar sound. To record an acoustic ensemble, we suspect most recording engineers would prefer to rely on well-positioned, high-quality room microphones as the basis for the mix. To that, close (accent) mics may be added to enhance details that are not well captured by the far-field microphones.

One could conceivably place room microphones in the loudspeaker-driven virtual soundfield, however, the relative dry/reverberant balance will be set accordingly. While this is preferable compared to relying solely on close microphones, it does not provide many options in post production. In particular, the ideal wet/dry levels for the performers may be different than is desirable for the final mix.

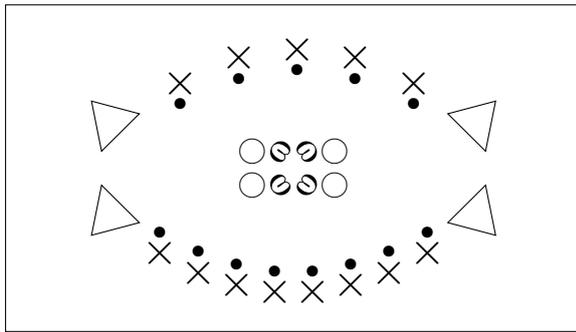
In order to provide flexibility for mixing and post production, we cancel the virtual acoustic reverberant signal from the room microphones, as described below in § 3.1. Through this method, our goal is to produce relatively dry room microphone signals even if the virtual acoustic space is highly reverberant. This facilitates both the recording and editing processes. Splice edits are easier to make with low amounts of reverberation, and the assumption is that an appropriate amount of reverberation can be added in once the editing has been completed. The reverberant tail of the loudspeaker-driven auralization may have a poor signal-to-noise ratio and may be corrupted by noise. By canceling the virtual acoustics and digitally reintroducing the reverberation in post production, we can alleviate these problems.

Our method combines pop-style editing with classical “tonmeister-style” recording setups. It affords a high amount of flexibility for editing and post processing while allowing the recording engineer to optimally place microphones in the physical space. Last, it also allows the auralization to be provided over loudspeakers to enhance the musicians’ comfort and interaction with each other and the virtual acoustic space.

## 2.2 Recording Configuration

One must carefully consider the placement of the musicians, speakers, and microphones when recording in a virtual acoustic space. First, the musicians are placed in the room. Close microphones should be positioned to capture each musician/instrument. Naturally, acoustic isolation is desirable but challenging to achieve. We recommend utilizing close mics with narrow polar patterns, e.g., hyper-cardioid mics. Then, the loudspeakers providing the auralization are positioned around the musicians. They should be located such that the musicians can hear one another and a well balanced “room” sound. Then room microphones should be well positioned to record the ensemble as a whole. We recommend a “tonmeister” approach to positioning the microphones based on listening to the natural room sound as well as the full loudspeaker-driven auralization. Care must be taken to position the loudspeakers and room microphones in such a way that the direct path and any significant early reflections are not impeded by the musicians or noticeably altered by their movement.

An example setup from a 2016 recording session with the a cappella group *Capella Romana* is shown in Fig. 1. The musicians were arranged in two arcs based on vocal role, facing one another. The musicians were close miced using hyper-cardioid lavalier mics made by Countryman Associates, model number B6D. These microphone signals were mixed and processed according to statistically independent room impulse responses, using real-time convolution to produce the live auralization as described in [9]. These signals were projected through Adam AX Series loudspeakers positioned above and behind the musicians and angled slightly downwards. The room microphones were positioned in the center of and above the musicians and speakers. We positioned two Neumann ORTF pairs and two DPA spaced omni pairs, each pointed at one of the two arcs of musicians. In addition to capturing a



**Fig. 1:** Example Recording Configuration. Above, musicians  $\times$  are close mic'ed  $\bullet$ , and arranged about omnidirectional  $\circ$  and cardioid  $\ominus$  microphones, and flanked by loudspeakers  $\triangle$ .

balance of the musicians, having several microphone pairs provides options when mixing after the recording session. This positioning of the room mics and chanters generated recorded signals that were noticeably drier than what the performers experienced.

In addition to recording all the microphone signals, we store the “wet” auralization signals that are projected from the loudspeakers for use in removing the auralization from the room mic recordings. We also use swept sinusoids to measure the impulse responses between all of the room microphones and loudspeakers for every configuration of musicians. Ideally, a number of impulse responses are measured for each configuration, so that the variation in the impulse responses due to performer movement, air circulation and the like is understood.

### 3 Cancellation Processing

It remains to describe the method for removing or “canceling” the unwanted loudspeaker auralization signals from the room mic signals. The idea is to estimate the auralization present in each room mic signal by convolving each loudspeaker auralization signal with its corresponding measured impulse response to the room mic in question. The “wet” signal estimate is then simply subtracted from the room mic recording.

#### 3.1 Cancellation Method

Consider a system with one source, one loudspeaker, and one room mic. Referring to Fig. 2, denote by  $s(t)$

the source close mic signal, and by  $h(t)$  the auralization impulse response, where  $t$  represents time. The loudspeaker signal  $l(t)$  is then the convolution of the source signal and auralization impulse response,

$$l(t) = h(t) * s(t). \quad (1)$$

Denoting by  $g(t)$  the impulse response between the speaker and room mic, the auralization signal picked up at the room mic is  $g(t) * l(t)$ . The room mic signal  $r(t)$  is then the mix of the desired “dry” source room mic signal  $d(t)$  and the auralization signal processed by the room,

$$r(t) = d(t) + g(t) * l(t). \quad (2)$$

The desired “dry” signal  $d(t)$  is estimated as the difference between the room mic signal  $r(t)$  and the convolution between a canceling filter  $c(t)$  and the known loudspeaker signal,

$$\hat{d}(t) = r(t) - c(t) * l(t). \quad (3)$$

where  $\hat{d}(t)$  is the “dry” signal estimate.

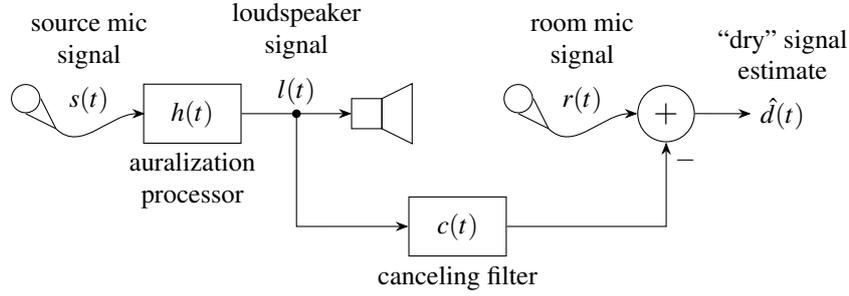
The question is how to choose the cancellation filter  $c(t)$ . It turns out that simply using the impulse response measured between the loudspeaker and room microphone can be overly aggressive. This can be seen by noting that there will be certain time-frequency regions in which the measured impulse response will be inaccurate, for instance in the reverberant tail due to performer movement and air circulation, and in low frequencies due to ambient noise in the room. In regions where the impulse response is not well known, the cancellation should be reduced so as to not introduce additional reverberation.

Here, we choose the cancellation filter impulse response  $c(t)$  to minimize the expected energy in the difference between the actual and estimated room microphone loudspeaker signals. For simplicity of presentation, for the moment assume that the loudspeaker-microphone impulse response is a unit pulse,

$$g(t) = g \delta(t), \quad (4)$$

and that the impulse response measurement  $\tilde{g}(t)$  is equal to the sum of the actual impulse response and zero-mean noise with variance  $\sigma_g^2$ . Consider a canceling filter  $c(t)$  which is a “windowed” version of the measured impulse response  $\tilde{g}(t)$ ,

$$c(t) = w \tilde{g} \delta(t). \quad (5)$$



**Fig. 2:** Simulated Acoustics Recording and Cancellation System. Close mic signals  $s(t)$  drive an auralization rendered over loudspeakers  $l(t)$ . High-quality room mics capture the combination of auralization and musician signals,  $r(t)$ , and are processed to remove the auralization contribution.

The expected energy in the difference between the auralization and cancellation signals at time  $t$  is

$$\mathbb{E}[(gl(t) - w\tilde{g}l(t))^2] = l^2(t) [w^2\sigma_g^2 + g^2(1 - w^2)]. \quad (6)$$

Minimizing the residual energy over the the “window”  $w$ , we find

$$c^*(t) = w^* \tilde{g} \delta(t), \quad w^* = \frac{g^2}{g^2 + \sigma_g^2}, \quad (7)$$

When the loudspeaker-microphone impulse response magnitude is large compared with the impulse response measurement uncertainty, the window  $w$  will be near 1, and the cancellation filter will approximate the measured impulse response. By contrast, when the impulse response is poorly known, the window  $w$  will be small—roughly the measured impulse response signal-to-noise ratio—and the cancellation filter will be attenuated compared to the measured impulse response. In this way, the optimal cancellation filter impulse response is seen to be the measured loudspeaker-microphone impulse response, scaled by a compressed signal-to-noise ratio (CSNR).

Typically, the loudspeaker-microphone impulse response  $g(t)$  will last hundreds of milliseconds, and the window will be a function of time  $t$  and frequency  $f$ , multiplying the measured impulse response:

$$c^*(t, f) = w^*(t, f) \tilde{g}(t, f), \quad (8)$$

$$w^*(t, f) = \frac{g^2(t, f)}{g^2(t, f) + \sigma_g^2(t, f)}. \quad (9)$$

We suggest using the measured impulse  $\tilde{g}(t, f)$  as a stand-in for the actual impulse  $g(t, f)$  in computing

the window  $w(t, f)$ . We also suggest smoothing  $g^2(t, f)$  over time and frequency in computing  $w(t, f)$  so that the window is a smoothly changing function of time and frequency.

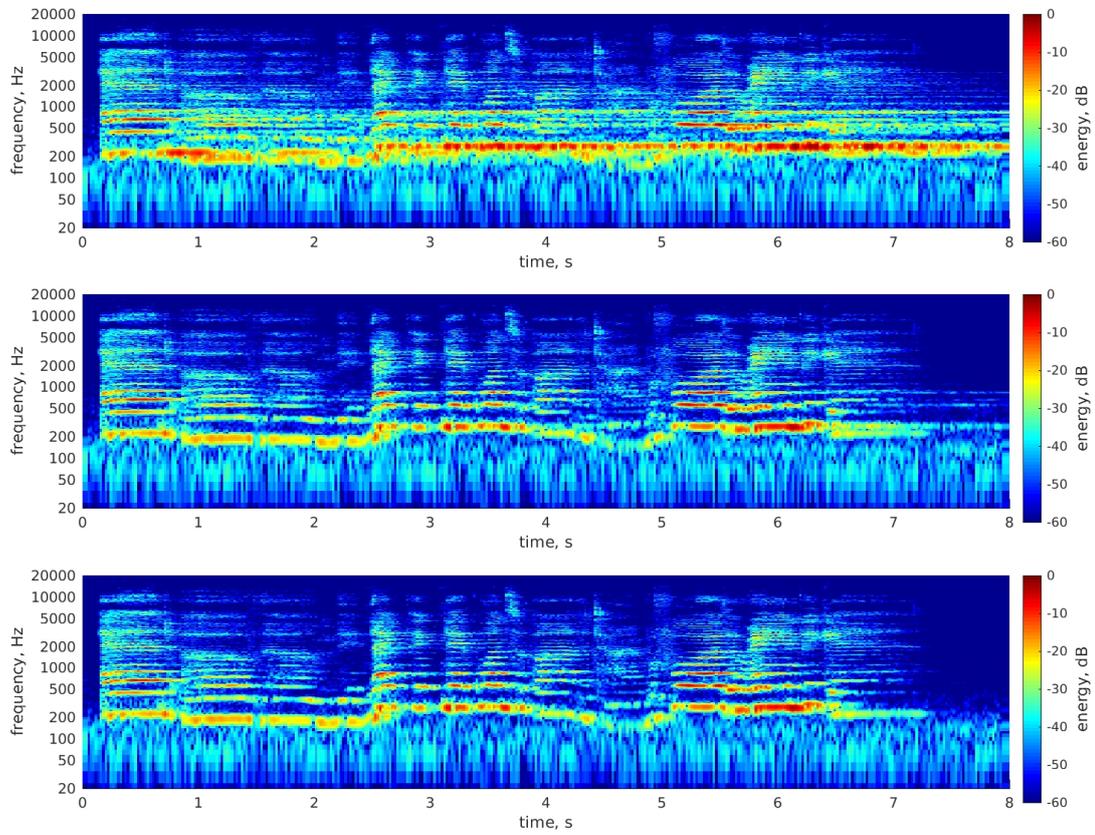
### 3.2 Practical Considerations

In the presence of  $L$  loudspeakers and  $R$  room microphones, a matrix of loudspeaker-microphone impulse responses is measured, and used to subtract auralization signal estimates from the microphone signals. Stacking the microphone signals into an  $R$ -tall column  $\mathbf{r}(t)$ , and the loudspeaker signals into an  $L$ -tall column  $\mathbf{l}(t)$ , we have

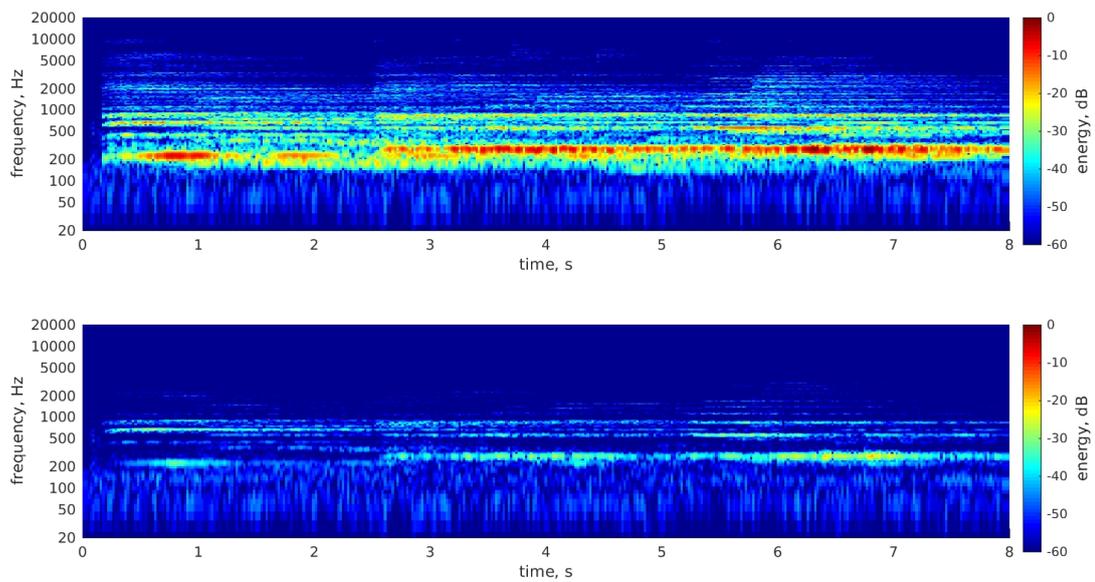
$$\hat{\mathbf{d}}(t) = \mathbf{r}(t) - \mathbf{C}(t) * \mathbf{l}(t), \quad (10)$$

where  $\mathbf{C}(t)$  is the matrix of loudspeaker-microphone canceling filters, and  $\mathbf{C}(t) * \mathbf{l}(t)$  represents the convolution of the canceling filter matrix  $\mathbf{C}(t)$  with the loudspeaker signal column  $\mathbf{l}(t)$ , essentially a matrix multiply, with the multiplication operations replaced with convolutions. As in the single-loudspeaker, single-microphone case, the canceling filter matrix is the matrix of measured impulse responses, each windowed according to its CSNR.

It will often be the case that the overall level of the measured impulse responses is unknown. In this case, the levels may be estimated via least squares as the ones providing the best fit of the loudspeaker signal convolution to the recorded room mic responses. Consider, for example, the case of a single room mic with its samples  $r(t)$ ,  $t = 0, 1, \dots, T$ , stacked to form a column  $\boldsymbol{\rho}$ . The  $L$  loudspeaker signals processed by their corresponding canceling filters are similarly stacked to form an  $T \times L$  matrix  $\boldsymbol{\Lambda}$ . The column of  $L$  unknown



**Fig. 3:** Recorded (top), Estimated Source (middle), and Source (bottom) Signals.



**Fig. 4:** Recorded Auralization (top) and Canceled Residual (bottom).

canceling filter gains  $\boldsymbol{\gamma}$  is then the one producing the best fit to the room mic signal  $\boldsymbol{\rho}$ ,

$$\hat{\boldsymbol{\gamma}} = \underset{\boldsymbol{\gamma}}{\operatorname{argmin}} \boldsymbol{\varepsilon}(\boldsymbol{\gamma})^\top \boldsymbol{\varepsilon}(\boldsymbol{\gamma}), \quad (11)$$

where  $\boldsymbol{\varepsilon}$  is the difference between the recorded room mic signal sample column and its estimated auralization component,

$$\boldsymbol{\varepsilon}(\boldsymbol{\gamma}) = \boldsymbol{r} - \boldsymbol{\Lambda}\boldsymbol{\gamma}. \quad (12)$$

The estimated gains are

$$\hat{\boldsymbol{\gamma}} = (\boldsymbol{\Lambda}^\top \boldsymbol{\Lambda})^{-1} \boldsymbol{\Lambda}^\top \boldsymbol{r}, \quad (13)$$

and the estimated “dry” room mic signal, the projection of the room mic signal orthogonal to the columns of  $\boldsymbol{\Lambda}$ ,

$$\hat{\boldsymbol{d}} = \left[ \boldsymbol{I} - \boldsymbol{\Lambda}(\boldsymbol{\Lambda}^\top \boldsymbol{\Lambda})^{-1} \boldsymbol{\Lambda}^\top \right] \boldsymbol{\rho}, \quad (14)$$

In the presence of multiple microphones, the process described above is applied separately for each microphone.

It is useful to anticipate the effectiveness of the virtual acoustics cancellation in any given microphone. Substituting the optimal windowing (7) into the expression for the canceler residual energy (6), the virtual acoustics energy in the cancelled microphone signal is expected to be scaled by a factor of

$$v = \frac{\sigma_g^2}{g^2 + \sigma_g^2}, \quad (15)$$

compared to that in the original microphone signal. Note that the reverberation-to-signal energy ratio is improved in proportion to the measurement variance for accurately measured signals,  $\sigma_g^2 \ll g^2$ . By contrast, when the impulse response is inaccurately measured, the reverberation-to-signal energy ratio is nearly unchanged,  $v \approx 1$ .

### 3.3 Cancellation Analysis

To evaluate the performance of the cancellation, we configured a system similar to that described in Fig. 1, with a single loudspeaker source (a Klein + Hummel M52 single-driver powered monitor) in place of the musicians. A dry track, a section of Suzanne Vega’s “Tom’s Diner,” was played out the source speaker, with acoustics simulating the reverberant Hagia Sophia nave rendered through four Adam XS Series speakers. Three Neuman KM184 cardioid microphones captured a mix

of the source and auralization signals. Impulse responses between the loudspeakers and microphones were measured using an exponentially-swept sinusoid technique.

The source speaker and auralization speaker responses were recorded separately, and mixed to form data for cancellation experiments involving combinations of the four auralization loudspeakers. In each case, the room source and auralization signals were mixed so that they had equal energy, regardless of the number of auralization loudspeakers used.

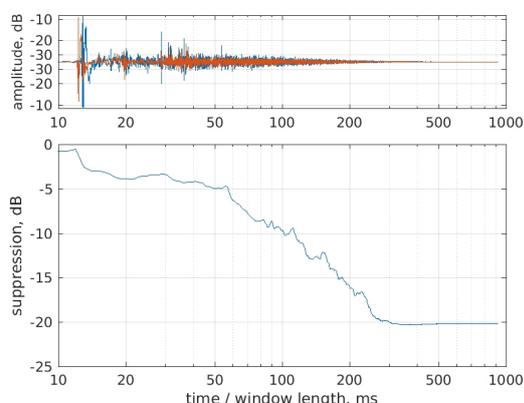
Typical results are seen in Fig. 3, which shows the spectrogram of a room mic recording (top), which is a mix between room mic recordings of the source speaker and the reverberant output of two auralization speakers.<sup>1</sup> The original (bottom) and estimated (middle) dry signal spectrograms are also shown. Significant reverberation is evident in the room mic recording, for instance as seen in the smearing over time of the component at roughly 200 Hz. Comparing the actual and estimated “dry” signals, very little of the reverberant auralization signal is present. Fig. 4 shows the spectrogram of the auralization (top) component of the recorded signal, along with the residual error (bottom) in the “dry” signal estimate. In this case, the additive auralization was suppressed by a factor of 20.2 dB. The auralization cancellation ranged from [17.5 dB, 22.5 dB], depending on which combination of loudspeaker auralization sources was used.

To explore the effect of slightly increased impulse response estimate variance with increasing time over the impulse response, we estimated the dry signal using impulse responses windowed to lengths ranging from 5 ms to almost one second, in 1 ms intervals. The auralization suppression was computed, and plotted against the measured loudspeaker-microphone impulse responses in Fig. 5. We see that about 5 dB of suppression is available using the direct path and a few early reflections, and that another roughly 15 dB of suppression is available in the late field onset, with little benefit available afterward.

## 4 Conclusion and Future Work

In this paper we described a methodology for recording in virtual acoustic environments. By providing

<sup>1</sup>Audio examples can be found at <https://ccrma.stanford.edu/~kermit/website/cancellation.html>.



**Fig. 5:** Canceling Impulse Responses (top), Residual Auralization Energy as a function of canceling impulse response window length (bottom).

the auralization over loudspeakers, musicians can fully interact with one another as well as the acoustics of the virtual space. We take a “tonmeister” approach to recording, using well-positioned, high-quality room microphones. By canceling the loudspeaker signals from the room microphones, we acquire relatively dry signals, which both provide flexibility in the editing and production process and allow an appropriate level of auralization to be added in post production or interactively.

Additional applications are possible if the canceling filtering is implemented in real time. A “canceling reverberator” can be made using an architecture similar to that of Fig. 2, but arranged in a feedback loop in which the microphone signals are reverberated and sent out the loudspeakers, and cancellation processing is used to suppress feedback. This approach has potential applications in art installations and virtual reality.

## Acknowledgements

We would like to thank Steve Barnett for valuable discussions and insights during his tenure producing an *Icons of Sound* virtual acoustics recording of *Cappella Romana* in November, 2016, Eoin Callery for significant effort configuring and running the recording experiments, and Kurt Werner for drawing several figures. We would also like to thank CCRMA and *Icons of Sound* for supporting this work.

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