Virtual Reconstruction of Spacial Reverberation
Graduate Project Proposal

Team Signal Wizards

Eric Farmer	Will Carroll
edf63@msstate.edu	woc17@msstate.edu
Leader	Member

September 30, 2018

Literature Review

To prepare for this experiment, the authors conducted a review of published literature. Six of these publications have been summarized and their applications to this project examined in the following paragraphs.

One application of synthetic reverberation is virtual reality [1]. In this article, the author demonstrates the necessity for acoustic feedback in video games to improve the user experience. The author explains the necessity for accurate 3-D audio in virtual reality, and he designs an example game to showcase the effects. Virtual reality has uses in many industries, and the ability for programmers to recreate realistic audio is important to the immersive experience.

An artificial reverberation effect can be generated by convolving the impulse response of a space with a given signal input [2]. In the article, the author presents discrete-time linear, time-invariant filters using feedback delay networks or single-input single-output state space. Additionally, the algorithm presented accounts for various parameters that effect the reverberation of a space. This article presents a description of algorithms that can perform real-time artificial reverbation effects using Matlab or VST.

A current field of research is the use of adaptive learning networks to optimize the spacial presence of a sound in a room [3]. This is important for a variety of applications e.g. live audio performances or studio recording. In [3], the author presents a system used to test the relationship between certain reverberation properties. Reverb is crucial to the improved subjective quality of a sound.

A key aspect to this experiment is determining the impulse response of a space. A paper from the Audio Engineering Society [4] discusses using an area microphone and a balloon pop to measure the impulse response of a room. It discusses the success of the measurement compared to traditional methods of room impulse response (RIR) measurement. This experiment did not use the results to recreate the reverberative response of the room for an arbitrary input signal, but the results are helpful in defining a method for measuing the impulse response in this experiment.

The presentation in [5] gives examples and real-world implementations of artificial reverb modeling techniques. These techniques were all modeled using specifically designed reverb systems and compared to the real reverberation effects of the room using qualitative listening tests.

Many common equations used to describe reverberative effects and gives a list of conditions that must be met in order to implement a useful artificial reverberation system are defined in [6]. These conditions may be useful to this project in evaluating the output of the test system.

Description of Project

The objective of this project is to create a system that artificially reconstructs the reverberative properties of an environment. Using the impulse response of the designated space, this system allows the user to simulate a space's acoustics for various inputs. The output of the artificial space is compared to the output of the real-world space. This ability is useful for sound engineers to test audio performance in different environments without the difficulty of a real-world experiment. Additionally, this system could reduce configuration time for public address (PA) systems by allowing the user to identify anomalies in the room's frequency response. This application is beyond the scope of this project, but serves as an example of the potential applications this experiment.

A stretch goal for this project is to find the minimum filter size that is qualitatively indistinguishable from the full filter size in the simulated output. This is a practical consideration to address real-time scenarios. There are applications for this in virtual reality and live presentation scenarios.

Description of Data

This experiment requires two sources of data — the impulse response and the audio input. Using a microphone, the impulse response of the room can be approximated by recording a balloon pop in the testing space. [1] From this measurement, a discrete finite impulse response (FIR) approximation for the space is obtained. Each impulse response is tested against three different audio input signals — a speech sample, an animal sound, and a musical sample. These input signals are constant throughout the experiment, but are arbitrary.

Description of Experiment

As previously stated, the objective of this project is to create a system, using Matlab, which artificially reconstructs the reverberative properties of an environment. Each environment has an infinite impulse response (IIR) that defines the reverberation that occurs in the space. Since IIR filters cannot be implemented in digital signal processors, the objective of this experiment is to determine a FIR filter that approximates the impulse response of the IIR filter. An FIR approximation of each space is obtained by inputting a impulse in the space and recording the output using a microphone. Because of the sampling rate and duration of sampling, this produces an FIR filter with a length of at least 50,000. While a large filter may not be practical for real-time implementation, there are applications in post-processing effects and room analysis that do not require real-time speed. However, a stretch goal is to minimze the filter size without an appreciative loss in quality, as determined by the qualitative analysis, so that the filters may be used in applications such as live performances or virtual reality simulations. The experiment is conducted by measuring the room impulse response, inputting an audio sample into the room while recording the output, and comparing to a synthesized version of the output from Matlab.

To provide a variety of reverberation scenarios, the RIR is captured from the following locations:

- Eric Farmer's Living room
- Simrall Hall Anechoic Chamber
- Simrall Hall Stairwell
- MSU SSRC Foyer
- MSU Old Intramural Fields.

The first step to measuring the RIR is to determine the noise floor of the room and the microphone. This is accomplished by recording the output of the room when there is no input. Using identical balloons filled to the same circumference, the balloon pop simulates the impulse input into each space. To prevent uneven spacial effects, the balloon and microphone are colocated in the center of the space. An operator pops the balloon using a sharp needle while standing at least one meter away while the microphone records the output of the room. The operator is not colocated with the microphone and the balloon to prevent distortions in the reverbation. The microphone records the impulse response until the room returns to a threshold set by the root mean square (RMS) of the noise floor. This process determines a large FIR filter for the space and is repeated for each location.

After measuring the RIR and the space returning to its noise floor, each audio input sample is played and the output of the room recorded on the microphone. To prevent spacial distortions, the audio input sample is played through a single speaker colocated with the microphone. Both the speaker and the microphone are positioned such that the focus of the output/input is upward. This process is repeated for each location.

After collecting the RIR for each space, Matlab is used to convolve each audio input sample with each spacial impulse response. The results are stored for later commparison.

The stretch goal is performed by using Matlab to reduce the size of the filter and calculating the output for each audio input sample at each location. The reduction is performed by removing every nth sample based on some reduction factor. For example, if the reduction factor was 2, then every 2nd sample would be removed. Thus, the filter size is scaled to $L_h = (1 - RF^{-1}) * L_h$.

Once the steps above are complete, the data is ready to be analyzed.

Description of Data Analysis

The data in this experiment is analyzed both quantitatively and qualitatively. The qualitative analysis is performed by five test subjects, which listen to the simulated and real outputs for each space and rank the similarity in reverberative qualities based on a scale from 1-10. The quantitative analysis is performed using Matlab. The simulated and real outputs are aligned and normalized, and the mean square error (MSE) between the two outputs is calculated. A MSE is also calculated in the frequency domain between the two samples by taking the fast Fourier transform (FFT) of both outputs. The MSE results show the variation from the real-world and simulated outputs.

For the stretch goal, only qualitative analysis is performed. The objective is to determine the smallest filter design that has no apparent differences to the listener. Hopefully, the results of the qualitative analysis reveal the smallest filter size.

This experiment contains possible sources of error. Neither the microphone nor the speaker can have a perfectly-flat frequency response, which partially distorts the measurements. The presence of a tester in the space to pop the balloon also introduces slight variation to the RIR, even though the tester is distant from the system. Other sources of possible error may be identified as the experiment is conducted.

Project Schedule

PR6	Obtain the equipment required to conduct the experiment. Run a collection test to refine the testing procedure and test for any unpredicted scenarios.
PR7	Collect data from test site one.
PR8	Collect data from test sites two and three.
PR9	Collect data from test sites four and five.
PR10-12	Perform data analysis and write the final report.

Both team members will be responsible for assisting with data collection and analysis.

References

[1] M. Beig, *Scalable immersive audio for virtual environments.* PhD thesis, 2018.

[2] W. L. Koontz, “State space filters for artificial reverberation effects,”

[3] D. Johnson and H. Lee, “Perceptually optimised virtual room acoustics,”

[4] J. S. Abel, N. J. Bryan, P. P. Huang, M. Kolar, and B. V. Pentcheva, “Estimating room impulse responses from recorded balloon pops,” in *Audio Engineering Society Convention 129*, Audio Engineering Society, 2010.

[5] V. Välimäki, J. D. Parker, L. Savioja, J. O. Smith, and J. S. Abel, “More than 50 years of artificial reverberation,” Feb 2016. Available: https://mycourses.aalto.fi/pluginfile.php/46618/course/section/19552/Artificial

[6] M. R. Schroeder and B. F. Logan, “" colorless" artificial reverberation,” *IRE Transactions on Audio*, vol. 9, no. 6, pp. 209–214, 1961.