

MEASURING AND MODELLING THE REVERBERATION OF A BARE ROCK TUNNEL

Julian Parker, Sami Oksanen, Archontis Politis, and Vesa Välimäki

Aalto University, School of Electrical Engineering
Department of Signal Processing and Acoustics
P.O. Box 13000, FI-00076 AALTO, Espoo, Finland

julian.parker@aalto.fi, sami.oksanen@aalto.fi, vesa.valimaki@aalto.fi

Abstract

Acoustic impulse responses of an excavated tunnel were measured. Analysis of the impulse responses shows that they are very diffuse from the start. A reverberator suitable for reproducing this type of response is proposed. The input signal is first comb-filtered and then convolved with a sparse noise sequence of the same length as the filter's delay line. An IIR loop filter inside the comb filter determines the decay rate of the response and is derived from the Yule-Walker approximation of the measured frequency-dependent reverberation time. The particular sparse noise sequence proposed in this work combines three velvet noise sequences, two of which have time-varying weights. To simulate the directional soundfield in a tunnel, the use of multiple such reverberators, each associated with a virtual source distributed evenly around the listener, is suggested. The proposed tunnel acoustics simulation can be employed in gaming, in film sound, or in working machine simulators.

1 INTRODUCTION

Recorded impulse responses of spaces have been directly used as a means to add natural reverberation of a specific character to a dry recording, or as a way to listen how a specific space would affect a source signal, a process that in the context of architectural and virtual acoustics is termed auralisation [1]. A straightforward way to achieve that is to convolve a source signal with a recorded room impulse response (RIR), which herein is termed as convolution reverberation [1]. An alternative approach, which provides more flexibility and computational efficiency, is to estimate the most relevant parameters from a recorded RIR and then map these parameters to common reverberation algorithms, such as combinations of delay-lines and feedback delay networks [2, 1]. Common parameters are mixing times between the early and late reverberant part, arrival times and amplitudes of discrete early reflections and reverberation times of the late diffuse part [3].

In Sec. 2, we describe the measurement of the B-Format impulse response of an excavated rock tunnel, and present some analysis of its main characteristics. In Sec. 3 we present a method of approximating this reverberation for arbitrary input through a novel multi-channel sound reproduction system. In Sec. 5, we conclude.



Figure 1: *Overall view of the main practice drilling tunnel. The sound source was located at the far end of the tunnel and receiver locations varied along the tunnel. (Photo Otto Hedström)*

2 IMPULSE RESPONSE MEASUREMENTS IN A TUNNEL

The measured space is a teaching and test tunnel located at the Aalto University campus, see Fig. 1. The characteristics of the tunnel match closely the ones at excavation and underground construction sites. The tunnel floor consists of a layer of gravel on top of the bedrock. Parts of the tunnel walls and ceiling are reinforced with sprayed plaster coating, but the majority of the wall area is bare and extremely uneven rock.

The acoustic properties of the space were determined using the standard log-sweep method. A sound source is located at a certain position within the tunnel, and a set of logarithmic sine sweeps are played. The acoustic response to this sweep is measured using a microphone positioned at a pre-determined location, at a great enough distance to be well within the reverberant soundfield. Sound signal playback and recording were carried out with custom made software in Matlab. Log-sweep duration was 10 s and frequency range was 20 Hz to 20 kHz. The excitation signal was repeated five times in each speaker and receiver position to help avoid sources of random error.

The excitation signals were produced using a stand-mounted active loudspeaker (Genelec 1032A). To approximate an omnidirectional acoustic excitation with the directional speaker, the measurements were conducted in seven different speaker orientations and averaged. The set of speaker orientations consisted of six positions in the horizontal plane 60 degrees apart from each other and one where loudspeaker was pointing upwards. The response measurements were also carried out using a passive omnidirectional loudspeaker. The omnidirectional loudspeaker was limited in efficiency and was not

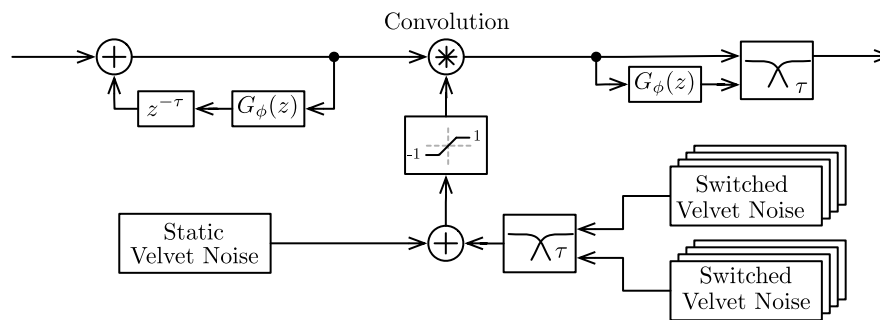


Figure 2: *Block diagram showing structure of a single channel of the novel reverberator.*

capable of producing the high SPLs required at greater distances, so the omnidirectional measurements were used to verify the results produced by the directional speaker.

Acoustic responses were measured using a pressure-field microphone (B&K 4192) and a B-format Soundfield microphone (ST 350) simultaneously. The microphones were placed approximately 1.5 m above the floor and close to the axial line of the tunnel. The distance between microphones was approximately 30 cm. The distance of the receiver point from the sound source was varied from 5 m to 20 m.

3 NOVEL DIRECTIONAL DIFFUSE REVERBERATION MODEL

As can be seen in the analysis of the measured responses of the tunnel given above in Sec. 2, the defining characteristic of reverberation in an irregular rock tunnel is that no strong early reflections are present, only a diffuse tail. The density of echoes is also extremely high from the start of the response. The other interesting aspect of the measured reverberation is the frequency-dependent variation of T_{60} with angle. A successful model should be able to reproduce this fast onset of diffuse sound, as well as being computationally efficient enough to be used in multi-channel or spatial audio context to reproduce the directional nature of the sound field.

3.1 Single-channel velvet noise reverberation

The basic reverberation structure is similar to those presented by Lee et al. [4], with some modifications. The core idea of the approach is that the diffuse tail of an acoustic response can be approximated by convolution with exponentially decaying white noise. However, direct convolution with exponentially decaying white noise has some drawbacks – namely that it is inefficient and lacks the possibility of varying reverberation time with frequency. Instead, we can use a comb filter to produce a series of exponentially decaying repeats of the input signal, and then convolve with a shorter sequence of noise to fill the gaps. The result is very close to the direct convolution, but computationally much less expensive. Frequency dependent reverberation time can be obtained by the addition of a damping filter into the feedback loop of the comb filter.

The efficiency of such a structure can be improved further by utilizing a form of sparse noise (noise that contains many zeros) instead of Gaussian white noise. Karjalainen and Järveläinen [5] propose one such type of noise, which they call ‘velvet noise’.

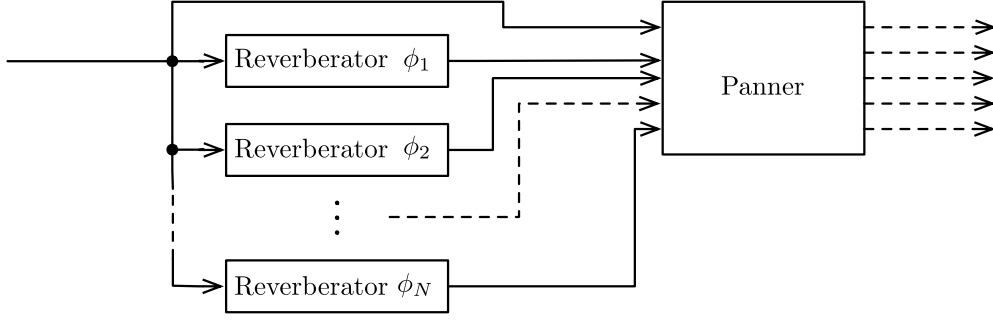


Figure 3: *Block diagram showing structure of complete multi-channel reverberator.*

This particular sparse noise possesses the desirable quality of a constant, non-lumpy distribution in time. This quality manifests itself as an audible smoothness. This density of this form of noise may be greatly reduced (down to as low as several thousand impulses per second) whilst still being audibly equivalent to Gaussian noise [5, 6].

The other major challenge of this form of reverberation is modifying the sparse noise sequence so that continuity is maintained between the sections delineated by the echoes of the comb-filter. In the case of frequency-independent reverberation time, this is simply a case of providing the noise sequence with an exponentially decaying envelope dependent on the length and damping coefficient of the comb filter. In the case of frequency-dependent reverberation time, Lee et al. [4] suggest crossfading between an unfiltered noise-sequence and the same noise sequence filtered with the damping filter.

Figure 2 shows the proposed structure of the reverberator. It consists of a comb filter of length τ samples, containing a damping filter $G_\phi(z)$. This damping filter is an IIR filter derived from Yule-Walker approximation of the frequency-dependent T_{60} of the measured tunnel response for a particular direction. The output of this comb filter is convolved with a sparse noise sequence of length τ . This sparse noise sequence is a new hybrid of the non-overlapping sequence and overlapping sequence methods described by Lee et al. [4]. Three noise sequences are generated. Firstly, a static noise sequence of low density (500 impulses per second). Secondly, two denser noise sequences (2000 impulses per second) are generated, and crossfading is performed between them over each comb-filter period τ . At the end of a comb-filter period, the first noise-sequence (where the crossfading started) is replaced by the second noise sequence, which is then in turn replaced by a newly generated noise sequence. This process repeats for every comb-filter period τ . The crossfaded noise sequence is added to the static noise sequence, passed through a hard clipper to remove the occasional occurrences of co-incident (and hence double amplitude) impulses, and then convolved with the output of the comb-filter. This combination of static and varying sparse noise was chosen as it successfully suppressed artifacts whilst still being sparse enough to keep the convolution efficient. Finally, the convolved signal is crossfaded over each period τ with a version of itself filtered by the damping filter $G_\phi(z)$, in order to approximate the correct envelope. Experimentation showed that best results were obtained when the comb-filter period is $\tau \approx 30$ ms.

3.2 Multi-channel extension of the reverberator

To extend the reverberation structure to a directional model, we make the assumption that due to the highly diffuse nature of the reflections, the sound arriving from different directions is essentially uncorrelated. We can then approximate the directional response by the use of a number of separate reverberators, each of which is treated as a virtual source within the space, and spatialized according to some established system for distribution to a loudspeaker system (see Fig. Figure 3). In this case, we use VBAP [7] to perform this spatialization. Each reverberator ϕ_n has its damping filter G_{ϕ_n} derived from the directional T_{60} time measured in the same direction as the corresponding virtual source. For the purpose of this work we employed 8 reverberators and hence 8 virtual sources, distributed evenly at increments of $\frac{\pi}{4}$ in a plane around the listener. The dry signal is also treated as a virtual source, and placed at the front of the space, to be consistent with the position of the source during the measurements. It is possible that a smaller number of virtual sources and reverberators could be used without much loss of directional information due to the limited angular resolution of the B-Format microphone.

4 RESULTS

Figure 5 shows a spectrogram and T_{60} estimate of the response produced using the average of directional speaker responses, taken at a distance of 13m between source and receiver. The response is created from the B-format signal by creating a virtual microphone pointing towards the source. Inspection of the spectrogram shows that the response consists of only diffuse sound, with no clear discrete echoes visible. Figure 5 shows spectrogram of the impulse response of the first reverberator (and hence first virtual source) in the model described above. The first reverberator is placed at zero angle (i.e. straight ahead). This response can be compared to the measured response given in Figure 5, which is the response to which the frequency dependent reverberation of this particular reverberator has been fitted. The diffuse nature of the reverberation seems to have been captured correctly, and the T_{60} is approximated well.

The model was tested with a variety of input sounds, on an 8.1 surround audio system in a listening room conforming to the ITU-R BS.1116 standard. The resulting sound was consistent with what we experienced in the tunnel during the measurement process, and compared favorably to the sound produced by a first-order ambisonic decomposition of the B-format impulse response.

5 CONCLUSIONS

In this work we have presented measurements of the acoustic response of an excavated rock tunnel, and proposed a reverberation structure which can replicate these results. The reverberator could be applied in any situation in which the acoustic environment of a rock tunnel needs to be replicated. This could include game sound, film sound, and sound for simulators used for the training of machine operators. The reverberator is computationally efficient, and can additionally be applied in other applications where directional reverberation consisting of only diffuse sound is desired – for example as the late-reverberation portion of a more general reverberator. A more detailed description of the method is presented in [8].

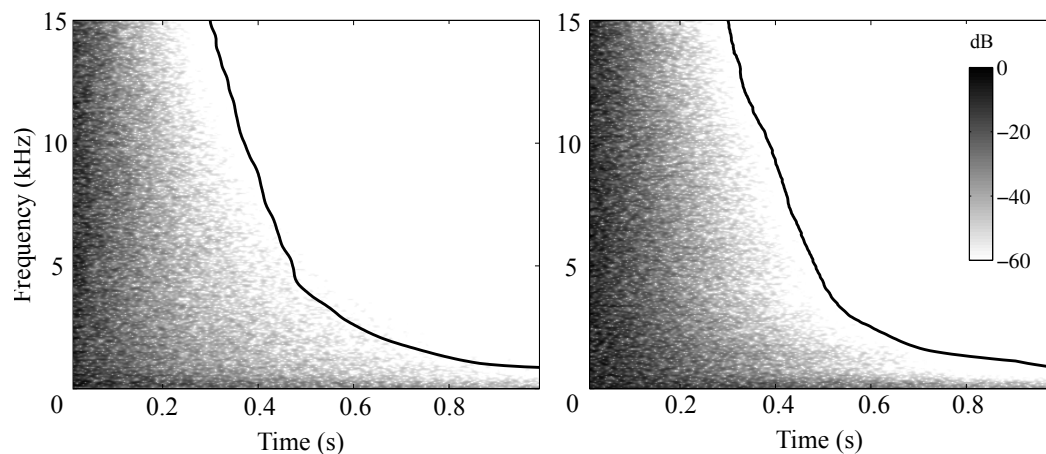


Figure 4: Spectrorams and T_{60} estimates (solid line): measured impulse response (left), and modeled impulse response (right).

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