B-FORMAT ACOUSTIC IMPULSE RESPONSE MEASUREMENT AND ANALYSIS IN THE FOREST AT KOLI NATIONAL PARK, FINLAND

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ABSTRACT

Acoustic impulse responses are used for convolution based auralisation and reverberation techniques for a range of applications, such as music production, sound design and virtual reality systems. These impulse responses can be measured in real world environments to provide realistic and natural sounding reverberation effects. Analysis of this data can also provide useful information about the acoustic characteristics of a particular space. Currently, impulse responses recorded in outdoor conditions are not widely available for surround sound auralisation and research purposes. This work presents results from a recent acoustic survey of measurements at three locations in the snow covered forest of Koli National Park in Finland during early spring. Acoustic impulse responses were measured using a B-format Soundfield microphone and a single loudspeaker. The results are analysed in terms of reverberation and spatial characteristics. The work is part of a larger study to collect and investigate acoustic impulse responses from a variety of outdoor locations under different climatic conditions.

1. INTRODUCTION

The reverberation characteristics, or *acoustic fingerprint* of a space can be digitally captured by the measurement of acoustic impulse responses. Such impulse responses can be used to artificially add reverberation to a sound using convolution. The result is a rendering of the sound as if being played in the space where the impulse response was originally recorded. This realistic and natural sounding reverberation technique is used for a range of applications including music production, sound design, computer gaming and virtual reality systems. Furthermore, by measuring impulse responses in real spaces using surround sound recording techniques, it is possible to capture spatial information relating to the reverberation of the space. This information can be used to provide surround sound auralisation, in order to artificially immerse a listener in the measured space.

Currently, impulse responses recorded in a range of outdoor environments are not widely available for surround sound auralisation purposes or for research. As a result, this work presents the first steps of part of a larger study to collect and investigate acoustic impulse responses from a variety of outdoor locations. We present an analysis of B-format impulse responses that were captured at three locations in the forest of Koli National Park in Finland when the forest floor was thick with snow. The results are investigated in terms of the reverberation characteristics of the locations, including a spatial analysis of the sound propagation. In addition, the resulting acoustic impulse responses have been made available online and can be freely downloaded either for creative or research purposes.

In previous related work, a number of acoustic surveys in forest environments have been reported with the aim of investigating sound attenuation rates [1, 2, 3]. Investigations into reverberation decay curves of pine forests were introduced in [4]. In [5] acoustical characteristics of a forest environment were compared directly with those of a concert hall. The latter paper describes impulse responses that were measured from binaural recordings of pseudo-random noise played through a dodecahedron loudspeaker. Reverberation time and spatial information in the form of inter-aural time delay and cross-correlation were then calculated for the forest and compared with measurements in a concert hall. The study was extended further to investigate a bamboo forest in [6].

For this work we made use of logarithmic swept-sine waves in our measurements, as described in [7]. The method has a number of advantages over other approaches, in particular that non-linear harmonic distortion produced by driving the loudspeaker at a high level can be directly removed from the final result. In order to capture spatial information, impulse responses were captured in B-format using a Soundfield microphone [8]. Reverberation parameters were measured for each site according to [9] and elements of the *Spatial Impulse Response Rendering* (SIRR) analysis technique are used to investigate the directional properties of the measured sound propagation [10].

2. SITE CHARACTERISATION

Measurements were made at three sites at different parts of the forest in Koli National Park, referred to as Sites 1, 2 and 3 for convenience. Figure 1 shows the locations of the three measurement sites, along with their estimated geographic coordinates. The forest at the survey location is dominated by both deciduous birch and evergreen spruce, with sizes ranging from very small to an estimated 21 m in height with trunk diameters of up to about 60 cm. The landscape in the area is hilly in general, although no steep slopes or cliffs could be found in the immediate vicinity of the measurement apparatus. The measurements were made at the end of March, and the ground was covered with between 50 cm and 70 cm of snow, hiding any low lying vegetation or foliage that might otherwise be found.

Background noise levels at all sites were very low. Traffic noise was barely audible and disturbance from wind was negligible. Owing to the time of year, sounds generated by wildlife were also very scarce, even during daylight hours. However, during the measurement at Site 2 there was the presence of birdsong which can be heard as a very low level artifact in the resulting impulse responses. Temperatures at the time of measurements were recorded as -8.5°C at Site 1, 0°C at Site 2 and -1.5°C at Site 3.

The choice of the sites was limited by the requirement to find

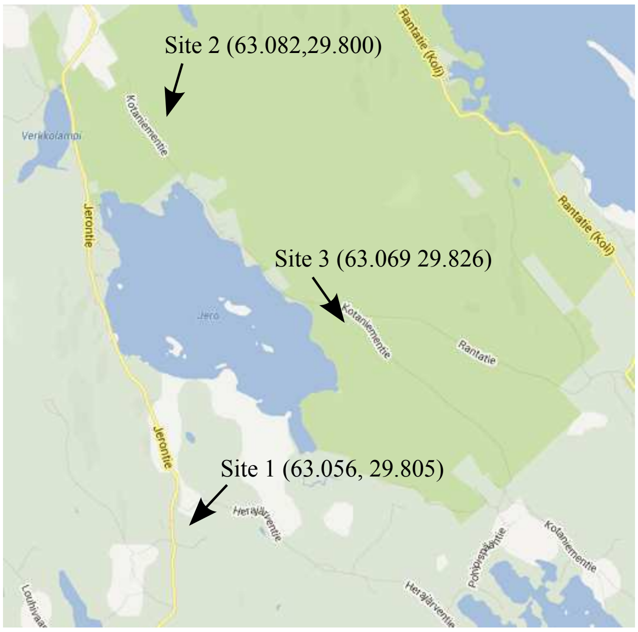

Figure 1: *Local map showing locations and coordinates of the three measurement sites. ©2013 Google.*

locations within the forest that were easily accessible by foot, given the need to carry playback and recording equipment in the deep snow. The three sites were chosen in different areas of the park with the aim of sampling a variety of characteristics. The first site was notable because trees on one side of the measurement apparatus were mostly birch trees without leaves while those on the other side were largely made up of evergreen spruce. At the other two sites the forest was mostly made up of both tree types mixed together and the areas were slightly more densely populated with trees than the first site. In the area around Site 3, the trees were older and larger in general than the other two sites.

3. ACOUSTIC MEASUREMENTS

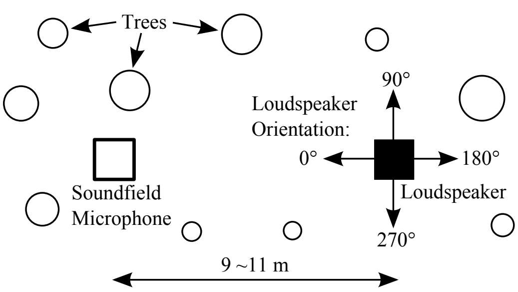

Figure 2: *Birdseye view of measurement apparatus in the forest environment, showing the four loudspeaker orientations.*

A single Genelec 8130A digital loudspeaker was used as the sound source for each measurement. The loudspeaker was powered by a silent portable battery pack. Signals were recorded in B-format using a portable Soundfield ST450 kit. A Tascam DR-680 portable recorder was used to record the four channels of audio from the microphone and a second DR-680 was used to drive the loudspeaker with a pre-recorded sine sweep. The two recorders were synced digitally using the digital through connection of the loudspeaker.

The loudspeaker and microphone were each placed at a height of 1.5 m from the ground and were separated by a distance of 9 m for Site 1, 11 m for Site 2 and 7.2 m for Site 3. A 30 second logarithmic sine sweep, sweeping from 22 Hz to 22 kHz, was played back through the loudspeaker and simultaneously recorded by the microphone at a sampling rate of 96 kHz. For each site the process was repeated 4 times, with the loudspeaker pointing in four different directions each time. The loudspeaker was first pointed directly at the microphone and then it was rotated on the vertical axis at 90^o increments, as illustrated in Figure 2.

Impulse responses were then obtained from the recorded sweeps using the deconvolution process outlined in [7]. For each site, we generate two B-format impulse responses which we refer to as IR-1 and IR-2. IR-1 was generated from the recording made with the speaker pointing directly at the microphone, and IR-2 was produced by summing the resulting impulse responses measured for each angle of the loudspeaker. The effect of the summation is to emulate a loudspeaker array of four loudspeakers at the same point in space all pointing in different directions. In this way, in the horizontal plane the sound source behaves as a closer approximation to an omnidirectional point source. This is an attempt to provide more information in the results by boosting the reverberated sound in the measurement. In addition, IR-2 complies better with the guidelines in the ISO 3382 standard for room acoustic measurements, which state that the sound source should be "as omni-directional as possible" [9]. It should be noted at this point that the ISO 3382 standard was designed to measure reverberation in indoor environments, however we consider its guidelines and techniques to analyse reverberation to be a good starting point for this research.

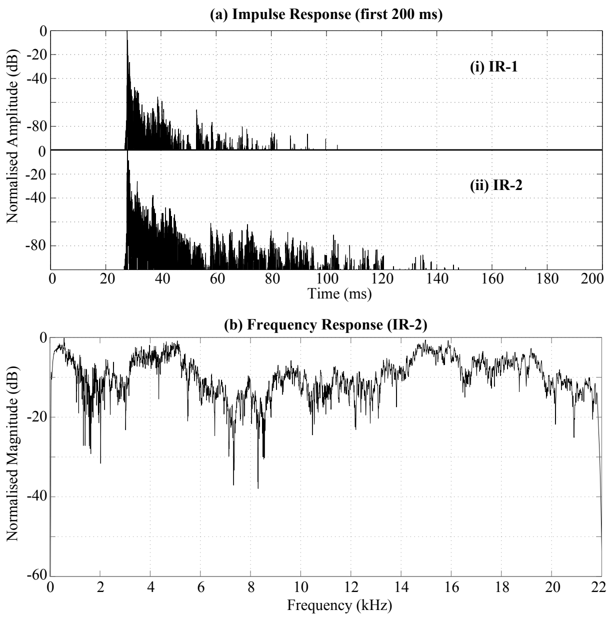

Figure 3: *(a) Measured amplitude envelopes of impulse responses (i)IR-1 and (ii)IR-2 and (b) frequency response of IR-2 at Site 1.*

Figure 3(a) displays the first 200 ms of the amplitude envelopes of the W-channels of (i) IR-1 and (ii) IR-2 measured at Site 1. It is evident that more reverberant energy is measured in IR-2 than in IR-1, as would be expected as a result of simulating a multi-directional loudspeaker array. The frequency response of IR-2 measured at Site 2 is displayed in Figure 3(b). All resulting impulse responses are available to download from the OpenAIR library website [11], as well as photographs taken at the sites at the time of measurement.

Although not within the scope of this paper, it is interesting to note that the multi-directional loudspeaker array is equivalent to a circular array with uniform angular distribution. Such an array can be used to approximate a variety of specific source directivity patterns by applying cylindrical harmonics [12]. However the resolution of the directivity patterns is limited by the order of the available harmonics, which in turn is limited by the number of elements in the array.

4. ANALYSIS

Reverberation Time (RT60) and Clarity (C50) measures were obtained and investigated for each site, as defined in [9]. RT60 is a measure of the time taken for the sound level of the impulse response to decay by 60 dB below the level of the direct sound. Specifically, the parameter T30 was used which extrapolates the RT60 value from a linear regression of the decay curve between -5 dB and -35 dB below the level of the direct sound. C50 is a measure of how clearly speech can be heard in a space. Generally, the higher the value of C50 the better the intelligibility of speech heard in the space. It is defined as the logarithmic ratio of early sound energy, arriving in the first 50 ms, to late sound energy that arrives after 50 ms:

$$\mathrm{C50} = 10\log\left(\frac{\int_0^{50} p^2(t)\mathrm{d}t}{\int_{50}^{\infty} p^2(t)\mathrm{d}t}\right) \tag{1}$$

where t is time in milliseconds and p is the instantaneous sound pressure level of the impulse response. C50 is measured in the decibel (dB) scale.

The AcMus Room Acoustic Parameters library for Matlab [13] was used in order to extract the T30 and C50 reverberation parameters from the impulse responses using the method described in [14]. Parameters were measured in octave bands from 63 Hz to 8 kHz. Resulting RT60 times for all three sites are displayed in Figure 4(a) and C50 measurements can be seen in Figure 4(b). For comparison IR-1 parameters are shown with a dotted line, and IR-2 parameters with a solid line. The two sets of measurements agree closely for lower frequency bands up to 250 Hz, owing to the inherent omni-directional nature of the loudspeaker at lower frequencies. At higher frequencies the 2 sets of values diverge, as the loudspeaker becomes more directional. For the remainder of the paper, we will concentrate on the IR-2 results.

Interestingly, the IR-2 RT60 times measured at each site show similar frequency dependent characteristics, with most of the reverberant energy apparently concentrated around the 1 kHz octave band. This would indicate that sounds at these frequencies are reflected most effectively by surrounding trees under these conditions. This may be explained by the fact that the width of the larger tree trunks, in the region of around 10 cm to 60 cm, are comparable in size to the wavelength of the propagating sound at frequencies around the 1 kHz octave band. C50 values show a high ratio between early and late energy, implying that speech intelligibility in these conditions is very good. Again, a frequency dependent pattern is shared by all three measurement sites, with C50 values reaching a highest value in the 125 Hz octave band and a lowest value in the 1 kHz octave band. Overall the reverberation parameters indicate a relatively dry reverberant space, although around the 1 kHz octave band the measured reverberation time is quite significant. For example, reverberation times found in the 1 kHz octave band of between 1.3 s and 1.7 s are comparable with that of a small church or concert hall [15].

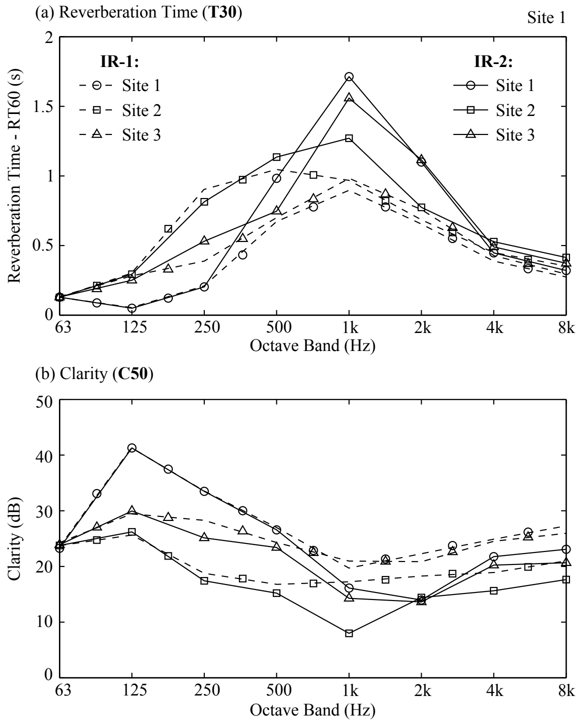

Figure 4: *Graphs showing (a) Reverberation Time (T30) and (b) Clarity (C50) calculated for the impulse responses measured at the three sites.*

To investigate the spatial characteristics of the reverberant sound propagation as measured by the B-format impulse responses, the instantaneous intensity vector $\mathbf{I}$ was measured for a range of frequencies at discrete points in time. Measurement of the intensity vector is part of the analysis method used in SIRR [10]. In order to perform the analysis, the B-Format signal was first divided into overlapping time frames of 512 samples in length, with an overlap of 256 samples. Each time frame was windowed using a Hanning window and then a short-time Fourier transform (STFT) was performed on each channel of the B-format response. The resulting frequency domain signals of the four B-format channels are labelled W, X, Y and Z respectively. The intensity vector at each time frame was then estimated in the frequency domain using the following equation [10]:

$$\mathbf{I}(\omega) = \frac{\sqrt{2}}{Z_0}\Re\{W^*(\omega)\mathbf{U}(\omega)\} \tag{2}$$

where $\mathbf{U}(\omega)$ is the vector $[X(\omega), Y(\omega), Z(\omega)]$, Z_0 is the characteristic acoustic impedance of the surrounding air and * denotes the complex conjugate.

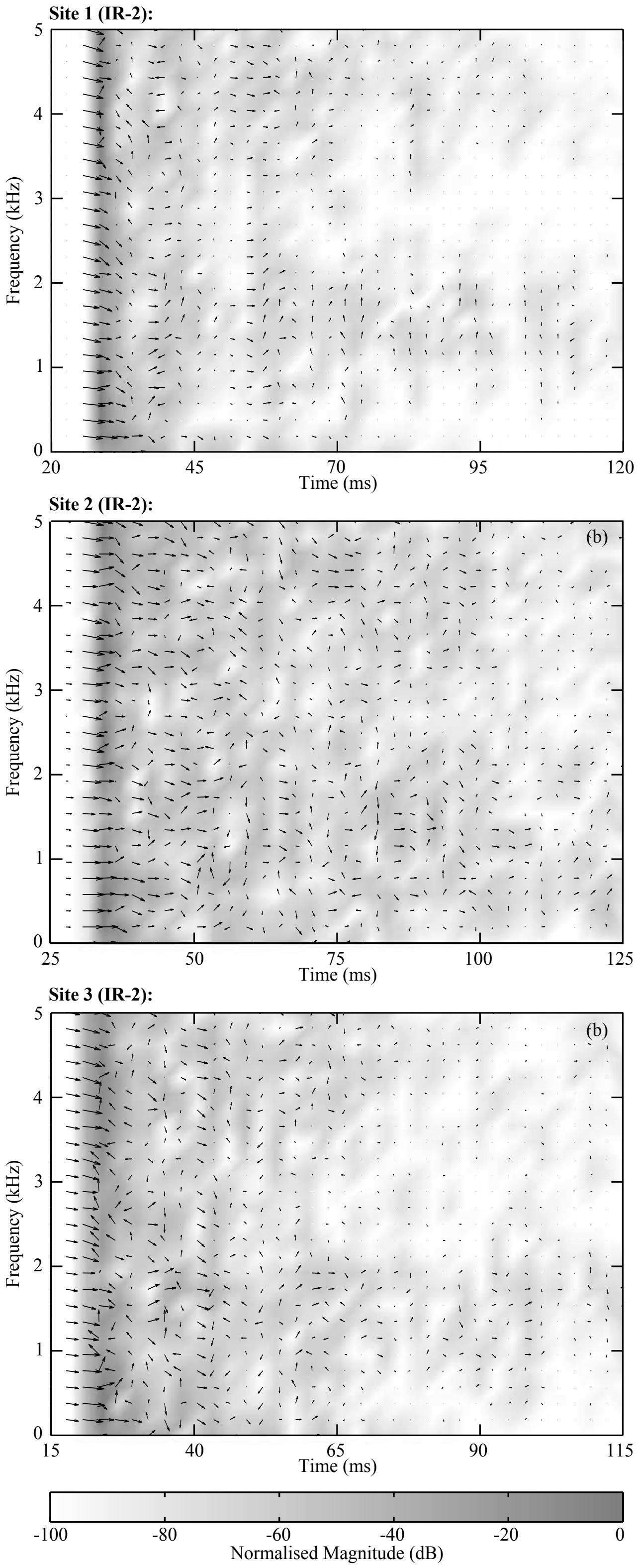

Figure 5: *Plots showing time-frequency distributions of instantaneous intensity vectors for each site superimposed on the spectrograms of the omni-directional responses.*

Note that for this purpose, only the horizontal plane was considered in the subsequent analysis and information from the Z channel of the B-format response was discarded. The intensity vectors measured using the IR-2 data for frequencies up to 5 kHz are plotted in the form of a quiver plot in Figure 5. The plots are shown for the first 100 ms of the impulse responses after the arrival of the direct sound, with the period of silence before the direct sound removed. The direction of the arrows correspond to the direction of sound in a particular time-frequency window, and the magnitude of the arrows are mapped to the logarithmic magnitude of the intensity vectors in decibels.

The plots reveal a strongly directional direct sound in each measurement, with the sound in the reverberant tail appearing to be very scattered as it reaches the microphone. However occasional clusters of vectors can be found pointing in the same direction lined up along the frequency axis, which would indicate the presence of distinct reflections, rather than purely scattered sound. Such reflections observed at higher frequencies (3 kHz plus) would most likely indicate reflections from nearby tree trunks. Interestingly, some of these reflections appear to be at lower frequencies, perhaps indicating reflections from nearby gradients in the terrain.

5. CONCLUSIONS

In summary, this work describes the process of measuring impulse responses in the snow covered forest at Koli National Park. The result is a series of B-format impulse responses measured at three different sites. These impulse responses have been made available online at [11]. In order to analyse the impulse responses, we have calculated values for both reverberation time (T30) and clarity (C50). In addition to these traditional methods we have looked at the time-frequency distributions of instantaneous intensity vectors for each site, in an attempt to investigate the spatial characteristics of the reverberant sound. The results of this analysis are displayed using quiver plots which give an interesting insight into the reverberant behaviour of sound in forest environments. The analysis shows us that in this particular type of forest, the amount of reverberant energy is concentrated around the 1 kHz octave band, with reverberation times of up to 1.7 s.

In future work, we will return to Koli National Park in the summer to repeat the measurements at the same locations in order to study how the acoustics of this environment changes with the different conditions. Without the presence of snow, we expect the forest floor to be more acoustically absorbent, resulting in some reduction in the reverberation time of the space. The reverberation time may be further reduced by the increase in temperature, resulting in an increase in the speed of sound propagation at the measurement sites. In addition, the arrival of foliage on the surrounding deciduous trees may slightly alter the acoustic properties of those trees at certain frequencies, perhaps acting to increase their absorptive properties and reducing the reverberation time even further.

We would also like to look at other ways by which we can both capture and analyse spatial information relating to the reverberation in these environments. In future measurements, we will experiment with longer sine sweeps to try to improve the signal-to-noise ratio in the results in order to better explore the nature of echoes and reverberation in the later parts of the reverberant tail that are unique to outdoor environments.

6. ACKNOWLEDGMENTS

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