SONIFYING MULTICHANNEL ULTRASOUND DATA FOR PERIPHONIC LOUDSPEAKER ARRAY

Timothy Schmele, Juan Alzate Romero, Thomas A. Troge

Hochschule für Musik Karlsruhe,
Institut für Musikwissenschaft
und Musikinformatik - IMWI,
76131 Karlsruhe, Germany
t.schme@gmail.com

Nicole Ruiter, Michael Zapf

Karlsruhe Institute of Technology,
Institut für Prozessdatenverarbeitung
und Elektronik (IPE),
76131 Karlsruhe, Germany
michael.zapf@kit.edu

ABSTRACT

The following paper describes several transformations of ultrasonic data into musical material for hemispheric loudspeaker setup. This data comes from a medical prototype for the purpose of early breast cancer detection via 3D multimodal imaging. The data is acquired in a semi-ellipsoid mesh of thousands of ultrasonic emitters and receivers surrounding the measurement object with water as medium. A hemispheric loudspeaker array can reflect this aperture design and offers the possibility of projecting this data in a direct fashion for auditory display. The ultrasounds in the megahertz range are inaudible and need to be transformed into the audible range. We here describe our investigations and methods to transfer medical ultrasound data in an artistic context and report on our insights using different sonification strategies to gain audible, musical material.

1. INTRODUCTION

This work describes different musical approaches to sonifying large datasets of ultrasonic measurements in real-time. We use the arrangement of the ultrasonic transducers in a semi-ellipsoid mesh to map the sonification of the measured signals in a 1-to-1 spatial correspondence, exploiting spatial patterns and appearances that emerge from said data. Because this work is of artistic nature, generating musical material is the main focus on each of these methods. Hence, the discussion centers around sonification for artistic purposes and spatial composition practices.

Ultrasounds, by definition, are sounds composed frequencies above the hearing range limit, which includes any sound above 20 kHz [1]. In the case of 3D USCT (3D Ultrasound Computer Tomography) around the center frequency of 2.5MHz with a 3dB bandwidth of approximately 1MHz [2]. Because of their small wavelengths and special properties, ultrasound based methods have a wide range of applications, including industrial cleaning, welding, geometry acquisition or even acoustic levitation. Particularly in the field of medicine, they are quite popular, as devices using ultrasounds for imaging and intervention applications are a viable, safe and cheap alternative to methods subjecting the patients to radiation, unpleasant procedures and more bulky and expensive devices.

The musical nature of an ultrasonic measurement is a dualistic one, as the signal itself was an actual sound wave travelling through a physical medium, which contains abstract information that represents real life objects. In this project, we want to look behind the data and reflect on its ontology, the constitutive elements of the pure signal and their sonic qualities and, in turn, what the signal itself represents, i.e. the interpretation and deduction of knowledge.

Sonifying any kind of data for musical purposes is a search for patterns contained in said data. Alexander et. al. [3] found that our auditory sense is able to better recognize patterns and thus irregularities, as compared to our sense of vision, promoting sonification as a valuable method for data analysis. Because the data is extracted and shaped with algorithms and directly turned into audible sounds, one may speak of this approach as a subset of algorithmic composition [4][5]. It may be said that a composer of algorithmic music leaves detailed compositional decisions to the algorithm or data, but the composer may equally argue that an algorithm or the data is not (yet) self-conscious and hence is not able to make any decisions at all, nor map itself to musical material. The composer instead seeks to intervene in the musical creation on higher, more abstract levels, with the algorithm adding a deterministic layer that transforms his compositional decisions into a (partly unexpected) sounding outcome. That said, in sonification for musical purposes, we do not seek to use algorithms as tools for musical creation, but we seek ways of making those algorithms audible themselves, alongside the data they use as input.

The outcome of these investigations may serve both aesthetic and scientific purposes. The techniques described here can lead to insights into the nature of the acquired data, thus leading to deeper scientific understanding of itself and its representation. Examples of artistic sonifications that are motivated by artistic merit but try to stay as close and true to the data for possible analytic usability can be found in [6][7][8][9][10][11][12][13]. For their database of musical works in sonification, Schoon and Dombois [14] used three criteria for inclusion: the transformation of the inaudible into audible signals, the acquisition of knowledge through listening and the development of listening techniques for scientific investigations. Using this list as a guideline, we will discuss our artistic approach as well as possible scientific applications in this paper.

2. SONIFICATION OF SPATIAL DATA

Spatial data, after the classification of Nasir and Roberts [15], is data that has a spatial component, usually the location, attached to it. The advantage of having a spatial dataset is the intuitive mapping of the spatial layout to the auditive panorama. In a meta-study on scientific publications related to sonification Dubus et. al. [16] identified 30 intermediate-level conceptual dimensions in the auditory domain grouped into 6 high-level categories. Most of the studies that wanted to convey location information used spatialisation in some form. The majority of these used the stereo space to position audio material between two loudspeakers, while full 3D audio, like Ambisonics [17] or Vector Base Amplitude Panning (VBAP) [18], are still relatively unexplored. In particular, the study in [16] shows that vertical location information is rather sonified with pitch or loudness than with spatialisation. Nasir and Roberts [15] also comment on the fact that spatial information has not yet been fully exploited, with sonification dimensions such as the Doppler and other time effects to convey distance being rarely used.

An example of artistic multi-channel sonification can be found in the publication by composer Andrea Polli who presented his composition Atmospherics/Weather Works in [6]. The method involved sonifying meteorological data across the whole eastern part of the USA through a 15 speaker setup, scaling a large phenomenon beyond the human grasp into a concert hall. He reports that the scaling and unpredictable rhythms and melodies generated through the sonification allowed the audience to gain insight into the complexities of nature. Similarly, Childs and Pulkki [19] further argue how 3D sound spatialization plays an important factor when sonifying spatially distributed data. In their study, they also sonify meteorological data. They argue that the human auditory sense is able to perceive simultaneous streams of information more so than the visual one, especially when distributing this information in sound across the listening space. Conversely, Katz et. al. [20] found no statistical significance in their study concerned with interactive data exploration, comparing spatialised to non-spatialised sonification methods. Still, they argue that with increased complexity of the tasks to be done, spatialisation would become a more important factor.

In medical applications, sonification can play a large role in the monitoring, analysis and diagnosis of various signals. One of the most widely known sonification techniques that has become a audible icon for the whole field of medicine itself is the heart rate monitor, with its characteristic "bleep" on every heartbeat alongside the emotional weight of the sound of a flat constant sine tone indicating the death of a human being – a widely recognized audible symbol. Apart from its cultural significance, it remains an effective and widely used sonification tool in medicine today.

In recent studies, sonification is being investigated for all fields of medical treatment. In respect to tremor diagnosis Pirr et. al. [21] have shown that even with preliminary sonification methods, a diagnosis would be very well possible and doctors could distinguish between different tremor types. Another study on the sonification of electromyographic (EMG) data [22] argues that not only does the transformation of data signals to audio free the eyes of the physiotherapist, or doctor in general, but the patient is also able to hear the signal and, in the case of EMG sonification, is able to match his or her to the target sound of a health person. Because of the cocktail party effect [1], we are very well able to steer, focus or broaden our auditory attention, having multiple streams of auditory information around us. Spatial hearing plays a large role in this ability, since the effect is destroyed when the same complex auditory scene is played back through a single loudspeaker in mono.

Apart from the few examples, most research in medical sonification seems to be centered around brain activity sonification. This is most likely due to the complexity of the brain activity and the little we still yet know of the entire mechanism. Sonification of brain data seems to be so attractive, because, for one, as explained above, that our hearing can decipher many streams of information simultaneously. In a study reviewing real-time electroencephalogram (EEG) sonification methods [23], the authors state that "[...] sound can readily represent the complexity and fast temporal dynamics of brain signals." The EEG signal is a time series signal that can be relatively effortlessly converted into audio, giving researchers and artists direct inspiration to investigate the possibilities of sonification, which they have been doing since the 1930s [23]. Studies that research into the temporal patterns of EEG signals and propose sonification for EEG signal analysis can be found in [24][25][26][27]. More specialized studies, e.g. targeting the diagnosis of epilepsy are [28][29] or the diagnosis of alzheimers in [30]. Apart from EEG signal analysis, sonification can be used to audify the direct activity of single neurons [12] or the blood flow detecting functional Magnetic Resonance Imaging (fMRI) analysis [8][13], which is said to be directly correlated to neural activity. The fMRI signal has a dense spatial volumetric resolution resulting in several thousand signals for every point in the brain. The artistic study presented in [8] makes use of this spatially distributed data with a periphonic loudspeaker system to create spatial correspondence of the sonified brain activity around the audience.

Sonifications with a spatial correspondence can also be done with EEG data, as each EEG probe has a fixed position on the scalp measuring the electrical activity in that surrounding area. Studies using multichannel or binaural sound distributions can be found in [31][32]. For Baier et. al. [31] multichannel audible space is an important sonification parameter to represent the spatial dynamics of brain activity. Their argument is based solely on the general analysis of the EEG by separating streams of information through spatial distribution. Exact and correct location of spatial dynamics is not an important issue or consideration in their study. This short review given on spatial and medical data sonification is by no means exhaustive. Nevertheless, we could not find a study outlining experiments with the sonification of medical ultrasound data, or any sort of ultrasound measurements in general, let alone spatial ultrasound data. The study is related in the sense that it uses a large dataset of spatial data in the medical field, but lies outside of what has been considered by sonification researchers so far. Hence, we believe that these investigations are particularly novel, in that they discuss the audification of previously disregarded datastreams.

3. 3D USCT II AND ULTRASONIC DATA ACQUISITION

The prototype 3D USCT II (3D Ultrasound Computer Tomography) was developed at the Institute for Data Processing and Electronics at the KIT [2]. It is developed as a better imaging method for non-invasive and early breast cancer detection. The 3D USCT II has an aperture in form of a semi-ellipsoid which holds the ultrasonic transducers uniformly distributed following a 3D point spread function pattern (figure 1). This aperture has 628 emitters and 1413 receivers. It generates an approximate spherical wave front from a single emitter at a center frequency of approx. 2.4

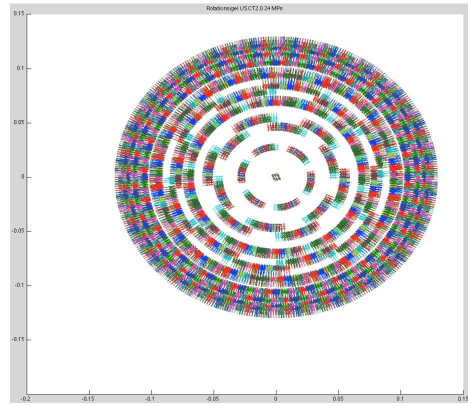
MHz with a 3dB bandwidth of approx. 1MHz. Through a mechanical rotation system, up to 23 further virtual positions of sources and receivers can be created to achieve a more accurate spatial resolution [2].

As described by Ruiter et. al.[33]:

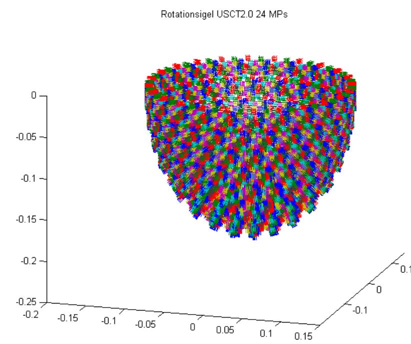
> The data acquisition is carried out with an FPGA based system which can store up to 40 GB of A-scans. The digitization is performed by 480 parallel channels (12 Bit @ 20 MHz), enabling data acquisition of one aperture position in approx. ten seconds.
>
> After digitization, the parallel data streams are processed by the FPGAs of the data acquisition hardware. The data streams are band pass filtered (1.67 to 3.33 MHz @ (-60 dB)) and the data rate is reduced by factor 6, performing a band pass undersampling. The reduced data is then stored in the internal 40 GB memory buffer. Using this approach it is possible to store up to 23 aperture positions in one data acquisition process.

For internal use, we define a TAS (Transducer A-Scan) element with 9 receivers and 4 emitters, evenly distributed at the moment, but for future versions the receivers and emitters will be also scattered over the TAS element surface to avoid harmonic resonances.

The USCT II was designed as a noninvasive breast cancer imaging method, as opposed to computer tomography (CT) imaging. It should provide medical doctors with 3D images for early diagnosis of breast cancer. Advantages of the device compared to other methods such as CT or MRI (magnetic resonance imaging) are affordability and reliability, also providing computational tissue pre-analysis. The first in vivo results can be found in [2] and research to improve the device continues at the KIT.



Top view



Side view

Figure 1: Rendering of all TAS elements in the USCT with all rotations considered.

4. ITS PERIPHONIC SOUND SYSTEM

The 3D sound experiments were carried out on the ITS (Interaktive Tonsphre) periphonic sound system of the Institute of Musicology and Music Informatics (IMWI) at the University of Music in Karlsruhe (HfM), Germany. The speaker system consists of 63 speakers arranged on a theoretic sphere plus a subwoofer channel, using every channel of a Multichannel Audio Digital Interface (MADI)[1]. The room in which this system was built is relatively small with around 56 cubic meters. Its main purpose is the study of spatial parameters and periphonic channel agnostic audio [34] in electronic music composition, as well as the perception of such musical parameters on both physiological and psychological levels.

Because the room was not specifically designed for such a system, the following trade-offs had to be made. Most importantly, it was impossible to install speakers beneath the listener, forcing us to remove the bottom third of the sphere. Additionally, the design had to take into account other objects and experiments, e.g. an experimental Wavefield Synthesis system, making it difficult to place speakers exactly where desired. To calculate the speaker positions, we used the 3LD Matlab library [35]. Instead of choosing a platonic solid or geodesic sphere and mutilating it to fit our purposes, we took the approach to modify the algorithms for the minimal energy configuration calculation with the additional constraints to find the best of the suboptimal solutions. The minimal energy configuration simulates a distribution of electrons on a sphere until they reach an equilibrium after a certain number of iterations. In our version, instead of having an initial state of random positions, we start with a small area of positions on the top of the sphere and have the points arrange themselves towards the bottom. By doing so, we can create a mirrored second instance the process and join the two theoretical spheres at the exact slice where we removed the lower third. On each iteration, as the points from each side approach this threshold where the sliced spheres were joined, the points from the lower sphere prevent those from the upper one to move beyond the threshold. A verification step may restart the whole process if the condition for each point has not been met, after reaching a steady state. This way we can guarantee a minimal configuration in the considered two-third sphere with the exact amount of speakers we want.

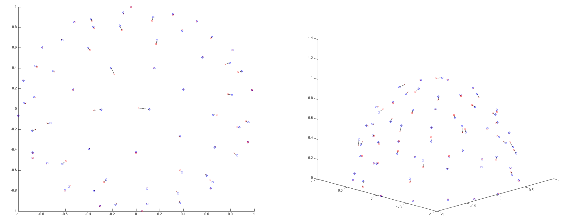
The room resonances were accounted for by measuring each speaker in its final position using a 20 Hz to 20 kHz 6s sweep and computing FIR filters using the DRC-FIR digital room correction toolkit[2]. The resulting system is by no means regular, but

[1]Details on the MADI specifications can be found on the AES10-1991 paper (revision 2003): http://www.iis.ee.ethz.ch/ felber/DataSheets/AES-EBU/aes10-2003.pdf (last accessed 26.02.2015)

[2]Details on the DRC toolkit can be found at http://drc-

the speakers were uniformly distributed. Using the IDHOA [36][3] python scripts for calculating ambisonics coefficients for irregular speaker arrays up to 5th order, resulted in very uniform energy and intensity distributions, apart from the bottom third, which has been excluded.

There are quite a few implementations for both VBAP [18] and Ambisonics [17] in several programming languages that are actually incapable of handling 63 channels. We have settled to using the ICST Ambisonic Tool for Max/MSP [37] for Ambisonics up to 5th order and the recently revised version of the VBAP implementation in SuperCollider[4]. Although, we are aiming to produce a complete implementation of a spatialisation server in SuperCollider for a distributed approach using OSC over ethernet and audio over MADI in the future.



Top view Angled view

Figure 2: Matching the USCT receiver position (red) to the speakers in the ITS periphonic speaker system (blue). Lines denote the respective error within a match.

5. SONIFICATION AND REAL-TIME INTERACTION

The delivered data was in Matlab format and had to be rendered to audio files for further processing in SuperCollider. The measurements are grouped by TAS elements. With a total of 628 emitters, 1413 emitters and 23 rotations per scanned object at 2.4 MHz sampling rate, the volume of the data is quite big for dealing with it in real-time. For this purpose, we decided to work with reduced sets of data and took different approaches to review its musical application. The data set for some of these experiments consists of 63 discrete signals taken from 63 individual receivers. The receivers were chosen based on the coordinates correlating with the coordinates of the loudspeakers of our ITS system, when also taking all possible rotations into account. This is not necessarily a trivial task, since the difference of scaling between two systems is at a level of magnitudes. For the best comparison, we normalized the distances of each TAS element and speaker with respect to their center and placed the two systems into one another so that the mean distance to all speakers and TAS elements is minimal. Also, we placed the 3D USCT II virtually on its head since the quasi-hemispheric designs of both the ITS speaker system and 3D USCT II semi-ellipsoid are rotated 180 degrees against each other. This is equivalent to the listener putting his or her head upside down into the 3D USCT II prototype. The matching result is shown in figure 2 showing the relative error of each pair of loudspeaker (circles) with matched receivers (crosses) with connecting lines. The higher error towards on the top of the hemisphere results from the lower coverage of ultrasonic receivers, as can be seen in figure 1.

The receiver selection above is efficient but rigid. As an alternative, we also matched 64 receivers equally spaced across the whole semi-ellipsoid. With the recorded receiver positions as meta-data, we could then position the extracted audio stream (as described below) via VBAP across the whole listening area, maintaining the relative distance to each other. Sonically and spatially, the results were very similar. With all the methods described below, one could perceive the same immersive sensation and spatial elongation along all dimensions in the audio signal. The advantage of this method lies both in flexibility, as this setup is channel-agnostic and can theoretically be played over any periphonic loudspeaker system and the modes of interactivity are increased, being able to rotate the whole construction in both azimuth and elevation, as well as experiment with rearrangements of the signals in space. Downsides to this approach is a slightly lesser precision in the placement of the sound sources. For example, the degradation in spatial resolution over several iterations of spatializing a 3d audio scene over VBAP can be found in [38]. Also, running 64 VBAP spatializations can can be quite heavy on a single computer. This can be avoided when outsourcing the spatialization algorithms to a separate machine. Nevertheless, due to its high speaker resolution, we chose to do the following experiments internally with the matched 63 weighing the options outlined above.

5.1. Direct Mapping and Process Sonification

Our first approach was the direct playback of these 63 chosen signals from the receivers to the corresponding loudspeakers. The responses for each emitter were concatenated one after another, creating a waveform of 1407000 samples for each receiver. These could then be played at different sampling rates, changing the pitch and duration of the sound. To provide examples we choose only two signals (left and right) to make the data and sonification process hearable and enjoyable on loudspeakers and headphones[5]. Other examples mixdown the 63 channels into a stereo field in order to hear the sonification process (i.e. controlling the amplitude of 63 harmonics) but the spatial dimension of the original data is discarded. Nevertheless, the experience in the ITS with 63 channels not only gives an immersive impression of the ultrasound scanning process, but also brings forth an enclosed space as all 63 are very similar in timbre, but mainly differ by spatially distributed time delays.

Since the original sampling frequency was about 3.33Mhz and we rendered the samples to an audio file at different audible sampling rates. For example, playing these audio files back at 10kHz the original chirps were still perceivable and the timing and rhythmic structure of the reflections were also more interesting to us than the 48kHz version. Though, of course, the signal loses in high frequency content is more drastic, as compared to 48kHz. Playing the samples back at higher sampling rates creates high pitched chirps with a fast rhythmic structure. Interestingly, in this first test, using the original, raw signals without any processing

fir.sourceforge.net/doc/drc.html (last accessed 26.02.2015)

[3]The open source IDHOA scripts can be found at https://github.com/BarcelonaMedia-Audio/idhoa (last accessed 26.02.2015)

[4]Details on the open source music programming language SuperCollider can be found at http://supercollider.sourceforge.net/ (last accessed 26.02.2015)

[5]These examples can be heard on https://soundcloud.com/3dultrasoundsonification

other than adjusting the playback speed to a hearable range, we can hear strong differences between the reflections of every emitter. The rhythms, slightly shifting and deviating over time create very interesting raw musical material. Also the rotating process is still very well hearable. Concatenating the signals in order of each emitter, of course, yields the original course of the experiment. With all received signals per emitter playing simultaneously in 3D one can perceive the path the signal took through the measured object. Theoretically, the actual object could, hence, be perceivable. The main goal in this project, though, is the emergence of musical, i.e. sonically interesting material.

5.2. Frequency Downshifting

Somewhat surprisingly, merely shifting the frequencies into the audible range did not fulfil our expectations. The idea was inspired by the approach taken in [9], where buoy data was upshifted from a 1Hz sampling rate. Instead, we wanted to see what results we would get if we shifted the signal downward from a 3.33MHz sampling rate. For this, we used ring modulation to shift the signal into the audible range (see figure 3). Afterwards, the signal had to be low-pass filtered and resampled by a factor of nearly 70 to be playable through a regular audio system. The results were disappointing, with the relatively short signal of 0.1884s consisting of nearly only noise. The problem lies in the short chip in the megahertz range, where a similarly short sample in the audible range does not yield useful information. Elongating the shifted signal in time and reinterpreting the original sampling rate only yielded similar results.

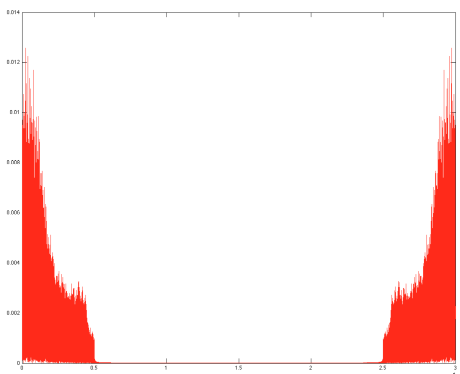


Figure 3: Shifting the inaudible frequencies into the audible range using ring modulation.

5.3. Parameter Mapping

A further approach was to use the signals as control parameters for different purposes. For a basic test we mapped the amplitude of the signal to the amplitude of sine waves. At first we played 1000Hz sine waves from every loudspeaker and controlled the amplitude of these with the scan signal. The overall rhythmic structure remains the same as in the original signal, but now we can compose a harmonic spectrum in space and map this rhythmic and spatial structure of the data to the harmonics of a single sound or several different sources. With frequency modulation and other parameters the results were similar (rhythm and space preserved) but it might require more fine-tuning and processing to make the sounds more interesting for a possible composition. A second version of this parameter mapping technique corrected the raw signals by reversing their bandlimiting and applying the match filter to obtain an approximation of the measured impulse response, which we rather used as a convolution filter (see section 5.5) instead of a parameter mapping.

5.4. Onset Detection

In this approach, we used a real-time onset detection algorithm in SuperCollider (Onsets.kr [39]) in order to change parameters during the playback and explore the rhythmic structures and quality of the reflections in the signal. We could use these onsets for different purposes (i.e. as impulse for a filter or as a trigger). The onsets were purified by detecting peaks in an envelope covering the original signal (see figure 4). We used the onsets as a trigger for sinusoidal envelopes of 100ms to sine waves of the same frequency and later to a harmonic spectrum. The main difference between the onset detection and the amplitude following method we used, is that the triggers give us a more detailed and discrete timing of the reflections. By changing the threshold we can filter out irrelevant changes in the signal or make the rhythmic structure more dense while the amplitude following was more imprecise and flat. The interesting effect from this method, is that changing the playback speed of the original signal keeps the timing of the onsets, while the amplitude following slows the overall attack and decay times of the controlled signal. After the onset detection we have the freedom to use these as trigger signals, and map them to different frequencies, either a multichannel harmonic spectrum, randomly oscillating signals or even scales and notes mapped to the loudspeakers. This last try revealed some musical patterns which were easier to identify than the harmonics or the random notes. There were some repetitions that could be heard because of the musicality, rhythm, and also the spatial distribution.

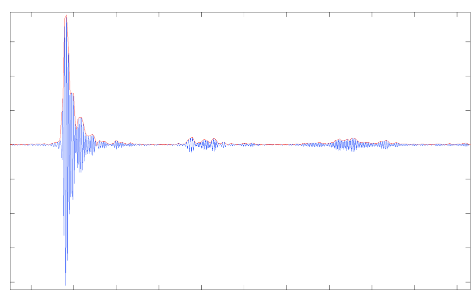


Figure 4: A match filtered signal and it envelope curve.

5.5. Convolution Filter

Instead of directly playing back the data as signal, one can use the estimated room impulses from the match filtering process and interpret it as a small space measured at high frequencies. If used without downsampling, the measurements would result in convolved signals, at least as an approach to an impulse-response method. Once downsampled, these impulse responses would react less as room responses and more like filters, drawing an input source out long and slightly shifting frequencies in different locations around the listener, giving a warped perception of the sound around him, changing constantly as the impulse responses are interpolated though each rotation. Using these FIR filter kernels gave us a spatial set up for each emitter that we could change live by crossfading between kernels, which affected the spectrum of the source but worked as a spatial setup, depending on the scanned object and the chosen emitter. An artifact which may be distracting and not musically desired is the comb filtering as a result from the time stretched FIR filters. In future work on this project, we would like to find a method of merely using found peaks in the impulse response for the FIR filter. The idea would be, that stretching those peaks in time by simply shifting their position in the signal, one can avoid comb filter effects since the filtered signal will not result in two close copies of itself, which causes the comb filter.

5.6. ISIS Downsampling

Up to this point, the data was merely downsampled, which forces a trade-off between the length of the signal and the amount of low-pass filtering as a result of lengthening the signal in time, or shifted in frequency, which did not yield satisfying results aesthetically. Also, a signal resulting downsampled to cover the whole audible frequency range would result in merely 62.5 ms of time. At least in an artistic context, compositions are a bit longer in length, so any composer would need to come up with ideas to either concatenate or further lengthen the signal in time. Merely lengthening the signal though will result in a constant downward frequency shift, which is not a desired effect at all times.

One solution is the use of an upsampling technique known as ”Intra-Samplar Interpolating Sinusoids” (ISIS) devised and developed by the composer Clarence Barlow [40]. ISIS stems from the perspective that white noise can viewed as a signal rapidly changing its frequency at every sample. As such, ISIS regards every sample in a signal as pitch information, which is used to devise a sinusoid that may fit in-between the samples. This means that a sinusoidal signal is fitted into the original signal at twice the sampling rate, where between each pair of successive samples in the original sound file, a wave emerges from the preceding sample, touches the maximum and the minimum amplitude and finishes in the successive sample. For our first tests, we used the current available implementation in SuperCollider. Timestretching the signal using this method will yield new sinusoidal components, creating an orchestra of similar melodies when played through a spherical loudspeaker system. The effect remains subtle throughout most of the audio. Mainly the transients create a burst of subtle melodies. For future work, the authors would like to implement a version of Barlow’s method in Matlab to apply it at the original 3.33MHz sampling rate and also tinker with the written algorithm to tailor the results to their liking.

6. CONCLUSIONS

The data from the ultrasonic scans taken with the 3D USCT II have proven to have rich spatial information that we can use for compositions and installations. Given the similar setups of both the parabolic 3D USTC II and the hemispheric ITS system, we could correlate the positions of the scanned signals to the spatial arrangement of the virtual audio sources, which could be used for educational purposes, possibly giving patients and doctors insights about the device and the ultrasound scanning process. Future studies using these sonification methods should focus on the didactic purpose of this specific spatial sonification.

We intend to give this data and scripts to students at the IMWI and KIT for further experiments and musical transformation. Future work would include further processing techniques, such as extracting the timings of reflections and use them for composing instrumental or electronic music, or different concatenation types of the original scans. Also the usage of different objects for the scans or real data from a patient would be another way to explore musical and sonification methods that might even yield relevant auditory display information that could be used in the medicine field.

7. REFERENCES

[1] D. M. Howard and J. Angus, *Acoustics and psychoacoustics*. Taylor & Francis, 2009.

[2] N. V. Ruiter, M. Zapf, T. Hopp, H. Kretzek, and H. Gemmeke, “3d ultrasound computer tomography at kit,” *J. Acoust. Soc. Am.*, vol. 135, no. 4, pp. 22–55, 2014.

[3] R. L. Alexander, S. O’Modhrain, D. A. Roberts, J. A. Gilbert, and T. H. Zurbuchen, “The bird’s ear view of space physics: Audification as a tool for the spectral analysis of time series data,” *Journal of Geophysical Research: Space Physics*, vol. 119, no. 7, pp. 5259–5271, 2014.

[4] I. Xenakis, “Formalized music: Thought and mathematics in composition. stuyvesant,” 1992.

[5] K. Essl, *The Cambridge Companion to Electronic Music*. Cambridge University Press, 2007, ch. Algorithmic composition.

[6] A. Polli, “Atmospherics/weather works: A spatialized meteorological data sonification project,” *Leonardo*, vol. 38, no. 1, pp. 31–36, 2005.

[7] C. Tittel, “Sound art as sonification, and the artistic treatment of features in our surroundings,” *Organised Sound*, vol. 14, no. 1, pp. 57–64, 2009.

[8] T. Schmele and I. Gómez, “Three dimensional sonificaton of fmri brain data in the musical composition neurospaces,” in *The 9th Conference on Interdisciplinary Musicology – CIM14, Berlin, Germany*, 2014.

[9] B. L. Sturm, “Surf music: Sonification of ocean buoy spectral data,” in *Proceedings of the 2002 International Conference on Auditory Display*, Kyoto, Japan, 2002.

[10] R. J. Winton, T. M. Gable, J. Schuett, and B. N. Walker, “A sonification of kepler space telescope star data,” in *Proceedings of the 18th International Conference on Auditory Display*, Atlanta, GA, 2012.

[11] L. Kessous, C. Jacquemin, and J. Filatriau, “Real-time sonification of physiological data in an artistic performance context,” in *Proceedings of the 14th International Conference on Auditory Display (ICAD 2008). Paris, France. CD-ROM*. Citeseer, 2008.

[12] M. Chang, G. Wang, and J. Berger, “Sonification and visualization of neural data,” in *Proceedings of the international conference on auditory display*, 2010, pp. 9–15.

[13] J. Thompson, J. Kuchera-Morin, M. Novak, D. Overholt, L. Putnam, G. Wakefield, and W. Smith, “The allobrain: An interactive, stereographic, 3d audio, immersive virtual world,” *International Journal of Human-Computer Studies*, vol. 67, no. 11, pp. 934–946, 2009.

[14] A. Schoon and F. Dombois, "Sonification in music," in *Proceedings of the 15th In ternational Conference on Auditory Display, Copenhagen, 18 to 22 May*, 2009.

[15] T. Nasir, J. C. Roberts, *et al.*, "Sonification of spatial data," in *The 13th International Conference on Auditory Display (ICAD 2007)*. ICAD, 2007, pp. 112–119.

[16] G. Dubus and R. Bresin, "A systematic review of mapping strategies for the sonification of physical quantities," *PloS one*, vol. 8, no. 12, p. e82491, 2013.

[17] J. Daniel, "Acoustic field representation, application to the transmission and the reproduction of complex sound environments in a multimedia context," *Engl. translation), PhD Thesis*, 2000.

[18] V. Pulkki and T. Lokki, "Creating auditory displays with multiple loudspeakers using vbap: A case study with diva project," in *Proceedings of the 5th International Conference on Auditory Display*, Glasgow, UK, 1998.

[19] E. Childs and V. Pulkki, "Using multi-channel spatialization in sonification: a case study with meteorological data," in *Proceedings of the 9th International Conference on Auditory Display*, Boston, MA, 2003.

[20] B. F. Katz, E. Rio, L. Picinali, and O. Warusfel, "The effect of spatialization in a data sonification exploration task." in *Proceedings of the 14th International Conference on Auditory Display*, Paris, France, 2008.

[21] D. Pirrò, A. Wankhammer, P. Schwingenschuh, R. Höldrich, and A. Sontacchi, "Acoustic interface for tremor analysis," in *Proceedings of the 18th International Conference on Auditory Display*, Atlanta, GA, 2012.

[22] S. Pauletto and A. Hunt, "The sonifiction of emg data," in *Proceedings of the International Conference Auditory Display (ICAD'06)*, 2006.

[23] A. Väljamäe, T. Steffert, S. Holland, X. Marimon, R. Benitez, S. Mealla, A. Oliveira, and S. Jordà, "A review of real-time eeg sonification research," in *Proceedings of the 19th International Conference on Auditory Display*, Lodz, Poland, 2013.

[24] A. De Campo, R. Hoeldrich, G. Eckel, and A. Wallisch, "New sonification tools for eeg data screening and monitoring," in *Proc. of the ICAD, 13th International Conference on Auditory Display, Montreal, Canada*, 2007.

[25] G. Baier and T. Hermann, "The sonification of rhythms in human electroencephalogram." in *ICAD*, 2004.

[26] Z. Halim, R. Baig, and S. Bashir, "Temporal patterns analysis in eeg data using sonification," in *Information and Emerging Technologies, 2007. ICIET 2007. International Conference on*. IEEE, 2007, pp. 1–6.

[27] G. Mayer-Kress, "Dynamical resonances and synchronization of auditory stimuli and evoked reponses in multi-channel eeg," in *Proceedings of the 2nd International Conference on Auditory Display (ICAD1994)*, Santa Fe, New Mexico, November 7-9 1994.

[28] C. A. Giller, A. M. Murro, Y. Park, S. Strickland, and J. R. Smith, "Eeg sonification for epilepsy surgery: A clinical work-in progress," in *Proceedings of the 18th International Conference on Auditory Display*, Atlanta, GA, 2012.

[29] G. Baier, T. Hermann, S. Sahle, and U. Stephani, "Sonified epileptic rhythms," in *Proceedings of 12th International*, 2006.

[30] A. Rogińska, K. Friedman, and H. Mohanraj, "Exploring sonification for augmenting brain scan data," *Proceedings of the 19th ICAD, Poland*, 2013.

[31] G. Baier, T. Hermann, and U. Stephani, "Multi-channel sonification of human eeg," in *Proceedings of the 13th International Conference on Auditory Display*, 2007.

[32] F. B. Vialatte and A. Cichocki, "Sparse bump sonification: a new tool for multichannel eeg diagnosis of mental disorders; application to the detection of the early stage of alzheimer's disease," in *Neural Information Processing*. Springer, 2006, pp. 92–101.

[33] N. Ruiter, M. Zapf, R. Dapp, T. Hopp, and H. Gemmeke, "First in vivo results with 3d ultrasound computer tomography," in *Ultrasonics Symposium (IUS), 2012 IEEE International*. IEEE, 2012, pp. 1–4.

[34] G. Cengarle *et al.*, "3d audio technologies: applications to sound capture, post-production and listener perception," Ph.D. dissertation, Universitat Pompeu Fabra, 2012.

[35] F. Hollerweger, "3ld - library for loudspeaker layout design a matlab library for rendering and evaluating periphonic loudspeaker layouts. iem-report 32/06," Institute of Electronic Music and Acoustics, Graz University of Music and Dramatic Arts, Austria, Tech. Rep., 2006.

[36] D. Scaini and D. Arteaga, "Decoding of higher order ambisonics to irregular periphonic loudspeaker arrays," in *Audio Engineering Society Conference: 55th International Conference: Spatial Audio*. Audio Engineering Society, 2014.

[37] J. C. Schacher, "Seven years of icst ambisonics tools for maxmsp–a brief report," in *Proc. of the 2nd International Symposium on Ambisonics and Spherical Acoustics*, 2010.

[38] T. Schmele, D. García-Garzón, U. Sayin, D. Scaini, and D. Arteaga, "Layout remapping tool for multichannel audio productions," in *Audio Engineering Society Convention 134*. Audio Engineering Society, 2013.

[39] D. Stowell and M. Plumbley, "Adaptive whitening for improved real-time audio onset detection," in *Proceedings of the International Computer Music Conference (ICMC'07)*, vol. 18. Citeseer, 2007.

[40] C. Barlow, R. Conservatoire, and J. van Stolberglaan, "Isis, an alternative approach to sound waves," in *International Computer Music Conference Proceedings*, 2005, pp. 1–4.