P.A.I.R.S.

Portable Ambisonic Impulse Response System

An Introduction to Ambisonic Impulse Response Recording

Andrew Rahman
Table of Contents

1. Abstract
2. Introduction
3. Background
   ○ Convolution reverb and impulse responses
   ○ What is ambisonics?
   ○ What is B-format?
4. Hardware
   ○ Brahma Ambisonic Microphone
   ○ Meyer Sound AMIE
   ○ Focusrite 18i8
   ○ MacBook Air
   ○ Cabling
   ○ Stands
   ○ Transport
5. Software
   ○ Apple Impulse Response Utility
   ○ Ableton
   ○ BrahmaVolver
   ○ Convolution reverbs
   ○ HISSToolbox and MaxMSP
   ○ VLC
6. Methodology
   ○ Hardware preparation
   ○ Software preparation
   ○ The Recording process
   ○ Teardown
7. Post-processing
   ○ Extracting A-format from AIRU
   ○ File management
   ○ Usage
   ○ Next steps
8. Results
   ○ Quality of files
   ○ Issues encountered
9. Future of Ambisonics
Abstract

Ambisonics, or 360-degree sound, is surround sound audio technology that incorporates the vertical axis in addition to the horizontal. It has been around since the 1970’s but, due to the space and processing power required to utilize it, it never established itself in the consumer market. Now, thanks to the resurgence of virtual reality, ambisonics has found a home. In this paper I explore recording in ambisonics using a soundfield microphone. The purpose is to capture impulse responses, or sonic snapshots, of acoustically and historically interesting buildings. The impulse responses can then be used in convolution reverb, a software audio effect which simulates the reverberation of the building in the impulse response. When an audio file or instrument is run through a convolution reverb it sounds as though it is playing inside the captured building. With the addition of the vertical axis, sounds can virtually be placed anywhere in the room, even above and below the listener. The primary application for this technology is currently in virtual reality but has the potential to expand into education, film, and music. This paper serves as a resource for anyone with an interest in using ambisonics or convolution reverb. Because this technology has been inaccessible to consumers until recently there are few definitive sources from an introductory standpoint to turn to. I hope to provide a gateway into ambisonic technology for consumers, audiophiles, composers, film sound recordists, film makers, video game designers, and early adopters.
Introduction

The purpose of building a Portable Ambisonic Impulse Response System (P.A.I.R.S.) is to capture the reverberation of historic and acoustically interesting spaces using ambisonics, an emerging technology, for archive, future research, composition and performance, and implementation in virtual reality (VR). This system is able to record an acoustic representation, or a sonic snapshot, of each space using a technology called impulse response (IR), which outputs a collection of audio files that can be used to digitally simulate the sound of the recorded space. Using an ambisonic microphone, instead of a set of traditional microphones, allows these files to be decoded into any playback format—mono, stereo, 5.1, binaural, etc. Currently, no one in the Bay Area is publically capturing IRs using ambisonics, and the only local IR available online is of Alcatraz Prison, which was recorded using an older stereo microphone technique. With this project I set out to learn the best practices for recording spaces using ambisonics, the best software for deconvolving the IRs, and to be the first, and possibly last, person to sonically document these spaces prior to their inevitable ruination.

There are several threats to historically significant sites around the Bay Area. According to the Bulletin of the Seismological Society of America, a major

---

1 Ambisonics is three-dimensional sound. One dimensional sound is stereo: a standard two-speaker system. Two-dimensional sound is surround sound, but those sounds are still on a horizontal plane only. Ambisonics records the sounds above and below the listener as well.
2 A six speaker array consisting of five full bandwidth channels and one low-frequency effects channel.
3 Binaural literally means “having or relating to two ears.” Binaural hearing, along with frequency cues, lets humans and other animals determine the direction and origin of sounds.
5 The easiest way to think of this is to imagine a recording of a guitar. Then convolving it with one of the IRs will make the guitar sound like it is being played within that space.
earthquake—magnitude 6.7 and above on the Richter scale—occurs on the Hayward Fault, which runs through Oakland and Berkeley, approximately every 150 years. History has shown that an earthquake of this magnitude can level an entire city, and with the sesquicentennial of the last major quake only two years away, when this occurs many, if not all, historic monuments along the faultline will be lost. Additionally, with global climate change causing rising sea levels, locations that are near the ocean have a high potential to be submerged. Finally, with gentrification at a national all-time high, culturally and/or historically significant sites may be purchased and repurposed or demolished. While many sites are documented through images and (more recently) video, very few are documented by their acoustics, or sound.

We have the phonograph record to thank for the impact blues music had on the Beatles, Cream, Led Zeppelin, and the rest of the British Invasion. It forever changed the shape of music, propelled rock to the musical mainstream, and became the primary mode of music propagation during the “Golden Age of recording.” Just as the phonograph is to the British Invasion, ambisonics will be to experiential multimedia. NASA saw the potential

---

13 The 60's and 70's
of binaural audio in conjunction with VR\textsuperscript{14} in 2000, and with the purchase of Oculus Rift for $2 billion in 2014 by Facebook, Mark Zuckerberg stated, “[v]irtual reality] has the potential to be the next important, or one of the next most important computing platforms.”\textsuperscript{15} Additionally, in 2016 Ossic X—a pair of 3D surround sound headphones—were under development and raised more than $5.7 million between Kickstarter and IndieGogo, which was named “the Oculus Rift for your ears” by PC Magazine.\textsuperscript{16} What used to require heavy equipment, expensive headsets, and a room full of speakers dedicated to VR can now be experienced with a smartphone, a piece of cardboard,\textsuperscript{17} and a pair of headphones.

In addition to archival and implementation in VR, these IRs can also be used in musical compositions. Yet, few people are currently composing with this technology because of its inaccessibility. Many of the available resources on ambisonics do not present a holistic approach to working with it, especially recording and implementing IRs. Most authoritative resources serve one very specific and highly technical function rendering them inaccessible to the amateur consumer and/or recordist. I intend to fill this gap by sifting through the myriad of articles, software, and best practices in hopes of finding the most elegant and effective process for recording ambisonic IRs from a consumer perspective. VR is the future of immersive multimedia and ambisonic recordings are necessary to suspend disbelief and create the sense of truly “being there.” If more spaces

\begin{itemize}
  \item \textsuperscript{14}Durand R. Begault, “3-D Sound for Virtual Reality and Multimedia,” National Aeronautics and Space Administration, last modified August 1, 2000, http://ntrs.nasa.gov/archive/nasa/casi.ntrs.nasa.gov/20010044352.pdf
  \item \textsuperscript{17}Google Cardboard https://vr.google.com/cardboard/\end{itemize}
are documented using this technology, users will be able to virtually experience and explore them through full sensory immersion—the same way pilots, military, and astronauts train for missions.

In this paper I walk through how I created a PAIRS and what I considered when selecting hardware and software for recording ambisonic impulse responses. My hope is to help this new medium flourish by exposing more people to the technology while also providing a guide for those that are interested in using it.

Background

Convolution reverb and impulse responses

Convolution reverb is a process used to digitally simulate the reverberation of a physical space. It is most often used in sound design for film, videogames, and music production. It is built upon the mathematical operation of convolution, an overlapping or blending of two functions. In convolution reverb, each sample of the unaffected audio (dry) is multiplied with each sample of a pre-recorded audio file—impulse response—of the space being simulated. The IR is a recording of the reverberance of a physical space. It is created by recording the reverberation that is caused when an impulse—an infinitesimally short sound containing all audible frequencies—is played in an acoustic space. Traditionally, the impulse was created by creating an electric spark, popping a large balloon, or firing a starter pistol. These techniques are typically referred to as “transient

\[18\] It can also be a digital model of a space but for the purpose of this paper I will refer to capturing IRs of physical spaces.
methods.” The transient method require less equipment and the recorded impulse responses can be used in the convolution reverb software immediately, without post-processing. However, this method captures a less accurate image of the space and the recordings typically contain a lower signal-to-noise ratio (SNR). The sine sweep method is a newer technique which plays back a sine wave which sweeps through the entire audible frequency spectrum—approximately 20Hz–22kHz—over a longer period of time, typically 10 to 50 seconds, which excites the space. While this method creates cleaner and clearer simulations, it requires far more equipment and the IRs require post-processing (deconvolution) before being usable. The process of deconvolution reverses the effects of convolution and is achieved by dividing each sample of the IR with the samples of the sine sweep. After deconvolution you are left with a file identical to a recording taken using a transient method, but of higher quality.

**What is ambisonics?**

Ambisonics is a full-sphere surround sound technique which is capable of transmitting a sound along the vertical plane in addition to the conventional horizontal plane. This technique is called “Periphony.” It was developed in the United Kingdom in the mid-70’s by Peter Fellget and Michael Gerzon with support from The National Research Development Corporation. Ambisonics can be understood as a three-dimensional extension of mid/side (M/S) stereo, which adds channels for height and depth. While there are several types of Ambisonics, this project and paper focuses on First Order Ambisonics.
(FOA) which require only four channels of audio to create 3D surround sound. The 4-channel signal set is called B-format. The channels are labeled W for the sound pressure (the M in M/S), X for the front-minus-back sound pressure gradient, Y for left-minus-right (the S in M/S) and Z for up-minus-down.

Due to the high cost, processing power, space required to properly utilize, and multi-speaker listening arrays, ambisonics never made it into the consumer world. Now with consumer access to high-power computing technology and the resurgence of VR, it is gaining traction and communities are beginning to form. For example, the HISSToolbox, a free package of multichannel convolution reverb and impulse response tools for sound programming language MaxMSP, both of which I will discuss later in this paper, has seen an average of 229 downloads a month for a total of 2,750 between March 2016 and March 2017. There is even an ambisonic performance space, Envelop, which opened at the end of April 2017 in San Francisco.

What is B-format?

B-format is the file type used in FOA and is processed from the raw data recorded from a soundfield microphone. Soundfield microphones, like the Brahma Standalone Ambisonic Microphone used in this project, contain four microphone capsules. Each of these capsules record one monophonic audio file, resulting in four mono audio files per recording. The four files together are called A-format and are labeled according to their location on the microphone: Front Left Up (FLU), Front Right Down (FRD), Back Left Down

---

(BLD), and Back Right Up (BRU). The A-format files do not represent the 3D space and are not intended for use without further processing into B-format. The A-to-B-format conversion consists of a series of mathematical equations which take the four monophonic recordings and encode them into a single four-channel audio file. This file represents 3D space using four audio streams: \( W \), which contains the Sound Pressure Level (SPL) and functions as an omnidirectional recording of the space, \( X \) (front-to-back), \( Y \) (left-to-right), and \( Z \) (top-to-bottom). The equation to reach these values follows:

\[
\begin{align*}
W &= FLU+FRD+BLD+BRU \\
X &= FLU+FRD-BLD-BRU \\
Y &= FLU-FRD+BLD-BRU \\
Z &= FLU-FRD-BLD+BRU
\end{align*}
\]

This simple process only functions in a mathematically pure world, thus the conversion process must also apply a set of filters to each file using convolution to compensate for the placement of the microphone capsules in relationship to one another. This is done to remove phase discrepancies and provide a higher spatial resolution. The B-format file is speaker independent\(^{21}\) which can then be converted into any number of playback formats and/or microphone types. B-format and ambisonics are open-source technologies and there are only a few dominant file types. The previous leading file type for B-format was .amb, however this is no longer supported by many audio softwares. But, thanks to an update by Microsoft in 2001, .WAV files can now hold a nearly unlimited number of

\(^{21}\)This means that even though B-format is only 4 channels of audio it can play over any number of speakers.
channels and has become the dominant file format for ambisonics as. As such, this paper will only reference the .WAV file extension when referring to multi-channel audio files.

**Hardware**

In this project I chose to use only consumer-grade hardware, opting for the most readily available and inexpensive products that meet the necessary requirements.

**Brahma Ambisonic Microphone**

At the beginning of this project only two consumer-grade ambisonic microphones were available: The Embrace Cinema Brahma Standalone Ambisonic Microphone and the Core Sound Tetramic. Each contain four cardioid polar pattern electret microphone capsules. While the Tetramic has a longer history with its first release in 2007, the Brahma, developed and released in 2014, was chosen for this project due to its ease of use and cost. The Brahma arrives complete, ready to use, in a shield, with two windscreens, mic stand mount, 9V battery-operated phantom power supply, 6ft. Ethercon cable, breakout cable, and is housed in a hardwood box, all for $935 after shipping to the US. These additional, yet necessary, pieces of equipment must be purchased separately for the Tetramic. Core Sound charges $1000 for the Tetramic alone, and $1329 for a kit containing the mic, breakout cable, power supply and connector cable. This kit is still incomplete lacking the mount, Ethercon cable (10ft. minimum), and shield/windscreen. Purchased separately these items cost $30, $32, and $35 respectively, making the total $1426 before shipping.
Embrace Cinema also sells a portable version of the Brahma mounted in a Zoom H2N field recorder. However, this was not chosen for the project as control over the recording process appears to be diminished and would most likely result in a lower SNR.

**Meyer Sound AMIE**

To reproduce the sine wave sweep in each space a flat-response, light-weight, portable speaker, small enough to fit into a standard flight case yet large enough to produce the low end of the frequency spectrum, is required. Initial research indicated that any of the Genelec 10xx series studio monitors is the speaker most commonly used in consumer and professional grade impulse response recording.\(^{22,23,24}\) With prices ranging from $2,500–$6,000 for one speaker, the Genelecs fell outside of the budget. However, thanks to the working relationship between Meyer Sound and the University of California, Berkeley’s Center for New Music and Audio Technology (CNMAT), an equivalent speaker—the AMIE—was loaned to the project. The AMIE studio monitor is a 6.5in. flat-response speaker with a frequency response of 45Hz–22kHz and weighing only 11.3kg (25lbs) and fits perfectly, with mount, into the Pelican case used in this project.

---


Focusrite 18i8

Because soundfield microphones use four microphone capsules, a lightweight and portable audio interface with four built-in preamps and phantom power is required. I chose the Focusrite Scarlett 18i8 USB Audio Interface for this project. It weighs 1.64kg (3.62lbs) and comes very highly rated with four mic preamps with up to 60dB of gain, low self-noise, and the ability to record at 96kHz, yet costs only $300.25

MacBook Air

A lightweight, powerful computer capable of recording four 10–60 second recordings at 96kHz 24-bit simultaneously is required. I chose the early-2015 13in. MacBook Air for this project. With a 2.2GHz quad-core processor capable of 3.2GHz turbo boost, 8GB of RAM, and 500GB SSD hard drive this 1.35kg (2.96lbs) laptop can fit into most bags yet is powerful enough to handle everything this project requires. Many of the available freeware ambisonic software are cross-platform, including BrahmaVolver, the processing application for the Brahma microphone. VVMic, the processing application for the Tetramic, however is a Windows only application which factored into the decision to select the Brahma over Tetramic.

Cabling

The cable runs need to be long enough so that the speaker and microphone can be approximately 60m (200ft.) apart when everything is connected. Starting from the Brahma

I used a 1.8m (6ft.) Male-to-Female (M-F) 7-pin DMX cable to extend the connection from the microphone to its phantom power adapter (PPA). Then a 30.5m (100ft.) shielded Ethernet 6 cable from the PPA to the XLR breakout cable, which is connected directly into the four mic preamps on the Scarlett 18i8. From the Scarlett into the computer I used a 0.9m (3ft.) A-to-B USB cable with two ferrite chokes. From output 1 of the Scarlett to the AMIE I used a 6m (20ft.) M-F TRS to XLR cable. For power I used two 15m (50ft.) extension cords with two matching 6-outlet surge protectors. The Scarlett comes with a 1.5m (5ft.) 12V power supply, the AMIE a 1.8m (6ft.) Neutrik NL-4 power cable, and the MacBook Air a 1.8m (6ft.) power adapter with an additional 1.8m (6ft.) grounded extension totalling 3.6m (12ft.) of length.

**Stands**

Only one mic stand and one speaker stand are required for this project. It is important to select the most sturdy stands available as they will help prevent low frequency rumble from translating to the microphone. They also help mitigate additional noise originating from the speaker stand that may read as a peak in certain frequency bands caused by a component of the stand buzzing or rattling.

**Transport**

The Pelican iM2750 Storm case was used to transport the equipment and cabling. This case is watertight, able to safely hold everything but the Brahma and stands, and sells for $250. For safety and stability, the foam padding the AMIE came in was used to prevent
the speaker from sliding. The computer and microphone were carried in a separate bag. With everything loaded it is possible for one person to transport the PAIRS nearly anywhere.

**Software**

A variety of software were explored in this project, ranging from creating sine wave sweeps, capturing recordings, deconvolving the sweep from the recordings, and convolution reverb for playback. Because the ambisonic format is open-source there are
plenty of software available that perform a variety of different functions. However, because ambisonics is not yet part of the mainstream, many have a poor user interface, require a high level of mathematical knowledge about ambisonics and/or computer programming skills, and/or have virtually no support. The following are the software I found to be best for entry into ambisonic impulse response recording.

**Apple Impulse Response Utility**

The Apple Impulse Response Utility (AIRU) was the primary software used in this project for several reasons. First, it’s free. While the software comes with Logic Pro, it is a standalone application that does not require Logic nor a serial/login to run. It has very extensive and thorough documentation available online and in the PDF user manual, both available on Apple’s website.26,27

The AIRU was used to play back the sine sweep, record the impulse responses, deconvolve, and denoise the IRs. With an elegant and easy to understand user interface, three different visual analyzers—waveform, energy, and spectrogram—and support for a bounty of formats, it’s a one stop shop for recording impulse responses. Additionally, it integrates seamlessly into Logic Pro’s convolution reverb plugin, Space Designer. While it’s great for taking nearly any type of impulse response it has one major flaw regarding B-format.

---

According to the AIRU user manual, “Impulse Response Utility allows you to encode all supported surround impulse response formats in B-format. As impulse responses do not make use of Z-axis information (vertical), Impulse Response Utility only records and encodes information from the W, X, and Y positions, saving CPU resources and disk space.” Which, as Dr. Rama Gottfried stated, is “quite presumptuous.” While this caused an initial setback I was able to find a workaround and chose to use it for the project.

**Ableton**

Ableton was used to record backup impulse responses using sine wave sweeps from Altiverb and Waves as well as combining the four A-format recordings into two stereo pairs required for use with the BrahmaVolver software. I chose Ableton simply because it is a DAW I already own and am very familiar with. However, there is one complication; Ableton does not support any multi-channel file formats. While all DAWs are capable of recording sweeps and combining mono files into stereo, one that natively supports surround files, such as B-format and 5.1, is necessary for implementation and analysis of the ambisonic files.

**BrahmaVolver**

BrahmaVolver is an A-to-B-format conversion software that comes with the Brahma microphone. It can also convert A-format recordings into 5.1 with and without a low-frequency effects (LFE) channel. The software only accepts two stereo A-format

28 Ibid.
files—FLU + FRD and BLD + BRU—and convolves them with a filter matrix unique to each Brahma microphone. This process applies calibration for phase accuracy and is essential for any ambisonic recording.

**Convolution reverbs**

Several convolution reverb software were used to test the quality of the impulse responses, however none of them properly support B-format. Altiverb 7 is a very powerful convolution reverb created by Audio Ease. It is the industry leader in convolution reverb software with support for up to 5.1 surround sound, as well as a unique sweep recording process which helps create an even cleaner reverb. Waves IR-1, Waves Audio’s convolution reverb, was also used, however it is important to note that this plugin only supports WAV files. Waves, founded in 1992, is the industry leader in digital signal processing technologies and audio effects for use in recording, mixing, mastering, post production, broadcast, and live sound. Waves also creates the IR-360, which supports up to 5.1 surround, however this plugin was not available for testing. Lastly, Logic’s Space Designer was used because it is directly linked to the AIRU, which is conventionally used to create Space Designer presets. While the Space Designer accepts B-format impulse responses, like AIRU, it does not support the vertical axis (Z). While Altiverb and Waves IR-1 are cross-platform, Space Designer only runs in Logic Pro X and thus only works on OSX.
HISSToolbox and MaxMSP

While the Huddersfield Immersive Sound System (HISS) Toolbox for MaxMSP is a free and very powerful resource for handling sweep generation, recording, deconvolution, denoising, and more, it requires a high level of computer programming skill to create a patch that does all of this. While free, it runs within MaxMSP, a visual programming language, which costs $400. The HISSToolbox was used briefly at the beginning and end of this project in an attempt to create an all-in-one B-format impulse response tool, however it was not implemented into the project due to my lack of programming skills. If you have experience programming with MaxMSP I highly recommend exploring this resource as it allows for complete customization of IR recording and processing.

VLC

VideoLAN Client (VLC), is a free, open-source, portable and cross-platform media player and streaming media server written by the VideoLAN project. VLC media player supports many audio and video compression methods and file formats, including DVD-Video, video CD and streaming protocols. This was simply used for playback as it supports multichannel audio files.

Misc.

Other freeware software under consideration were Deconvolv-0.1, a source code developed by Klaus Michael Indlekofer, Ambisonic Decoder Toolbox for MATLAB by Aaron J. Heller, ambiX v0.2 developed by Matthias Kronlachner, an ambisonic plugin set used for
post-productions, and the SoundField SurroundZone 2 VST plugin. While each of these software have their merits the first two require programming knowledge and separate applications. While the latter half are VSTs which can run in Ableton, but because of its lack of support for multi-channel audio files, these software were rendered useless. Additionally, Logic Pro X only supports audio units, rather than VST, and uses a validation tool that only allow plugins that comply with Apple’s plugin specification, rendering ambiX useless. While SurroundZone is the most accessible, its only function appears to be moving mic orientation and A-to-B-format conversion. While this software may be very useful it was not used due to Ableton’s lack of multi-channel audio support. Finally, each software, save SurroundZone, require quite a lot of additional reading and research not useful for an entry-level consumer, and thus were not used.

**Methodology**

**Space scouting and selection**

While waiting for the equipment to arrive I began scouting spaces on the UC Berkeley campus to record. I chose to focus on the oldest buildings, large reverberant spaces, structurally unsound buildings, performance spaces, and libraries. I compiled a list of the spaces of interest and reached out to the building administrator for each to gain after-hours access. It is prudent, and sometimes necessary, to access the locations after-hours because the recordings must be taken with no one else in the space as any extraneous noise will be picked up by the microphone.
Hardware Preparation

It is important to know where each of the mic capsules are located on your soundfield microphone. This is most easily achieved by first identifying the front of the microphone. Neither the microphone, capsules, nor breakout cable are labeled on the Brahma. I was able to identify the front of the Brahma by testing each capsule individually to find its position and which color on the breakout cable is associated with it. I then labeled each of the XLR cable ends on the breakout cable and the inputs on the Scarlett 18i8 with the corresponding mic capsule and color of the breakout cable. Finally, I labeled the shock mount “front” and “back” for ease of identification and would always ensure the etching of the Brahma’s production number was facing the “front” before recording.

If you are using a new speaker for your impulse responses you will want to first break it in. This is accomplished by playing sound at an above average volume through the speaker for approximately 70–100 hours. It loosens the suspension on the speaker cone allowing it to move more freely and improve low frequency response. I accomplished this by placing the speaker in a soundproof room playing low frequency sweeps, from 20–100Hz, continuously over a weekend.

Now with the microphone labeled and the speaker broken in it’s time to start testing. For the first test run use the fewest and shortest number of cables available. The results from this test will serve as your control for comparison against the longer cable throws. Once your PAIRS is set up in the test space, playback an anechoic recording—audio that contains no reverberation—that covers most of the audible spectrum at a comfortable volume. I chose to use a recording of a bassoon playing its entire range. While this
recording is playing, move around the room to find the sweet spot to place the microphone. The idea of a “sweet spot” is a bit subjective, but I found that it’s the spot that I kept wanting to return to. Once placed, play one of the sine sweeps that will be used during recording on a loop while adjusting the microphone preamps and output gain of the soundcard to find the optimum volume level—preamps and speaker are not clipping and recordings do not exceed -6.0dB. This will also ensure that each mic capsule is recording at the same volume. Once all levels are set use the AIRU to capture the impulse response of the test space. Use the deconvolve and denoise features to create the impulse response, then audition the reverb. After all tests are completed make a note of your settings, the configuration, and your results before moving on to the longer cables. Repeat the test using the most and longest cables.

After testing is complete break down your PAIRS and find an organized way of packing the equipment safely and efficiently into the travel case. It is important to find a set configuration for the PAIRS in the travel case to minimize setup and teardown times. In certain circumstances I had very limited time in the space, but by keeping the PAIRS organized I was still able to capture the IR. At this point the hardware is ready for use.

Software Preparation

In addition to hardware preparation, software preparation is essential. Prior to recording each location set the computer’s sample rate to 96kHz. For this I used Ableton because it was also used for recording backup IRs and it is not possible to change the

\[29\] Free anechoic recordings can be found at http://www.openairlib.net/
sample rate from AIRU. This allows the sample rate to be changed once and have it apply to all the necessary software. To prepare AIRU for recording, create a new project and set the Configuration dropdown to Mono/Omni to Quadraphonic then click OK and save the project in an easily accessible location. The default parameters are a good place to start for testing PAIRS in a new space but may need to be fine-tuned depending on the length of the reverb.

It is wise and recommended to record additional sweeps for redundancy. For each space I recorded five takes using AIRU for each microphone position as well as three to five takes in Ableton using sweeps from both Altiverb and Waves. To do this, open your DAW of choice, and create a new project that contains seven (7) audio tracks. The first is used to playback the sweep files and should be named accordingly. The Waves sweeps require a free account to download and contains only one 22Hz–32kHz 15 second sweep with a 96kHz sample rate and 24-bit bit depth. Altiverb’s provides a wide variety of sweeps ranging from 1–300 seconds at any sample rate, all with 24-bit bit depth, and can be downloaded directly without an account. Once downloaded, select the files with the desired sample rate and sweep length and load them onto the sweep track in your DAW with plenty of space between each. The next four (4) audio tracks correspond to the four mic capsules. Name and set the input for each track according to its capsule position (FLU, FRD, BLD, BRU) and color code them according to the color on the breakout cable. The final two tracks are used to record stereo inputs for both the front and back using inputs 1/2 and

---

30 Keyboard shortcut: ⌘N
31 Keyboard shortcut: ⌘S
3/4 respectively. This allows for two redundancies to record simultaneously for each sweep. Turn monitoring off and arm all of the microphone tracks to record. Now turn down the playback volume on the mixer for each track (or group/bus if your DAW supports it) to -3dB or more. This is to avoid clipping when playing back the recordings. Save the project in an easily accessible location.

The recording process

Once in the space, setup and connect all the equipment, but do not power it on. It is important to turn on the laptop first, soundcard second, speaker third, and microphone last. This is because when the sound card is turned on after the speaker or when the computer first connects to the sound card, the speaker generates a loud pop, which if not careful can damage the speaker or your hearing. To avoid this I power on the laptop and sound card, open the pre-made Ableton project, set the Scarlett 18i8 as the input and output device, set the sample rate to 96kHz, make sure the microphone channels are armed and monitoring is off, and turn down the volume knob on the front of the Scarlett before turning on the AMIE. After a few seconds the “On/Status” light turns green and the speaker is ready. I then load the anechoic bassoon sounds into VLC and play them on repeat as I slowly turn up the volume on the sound card to a comfortably loud level without earplugs. I then move throughout the space listening for the sweet spot to place the microphone. Once the microphone is placed I stop the bassoon sound, take a photo, power on the microphone, and open the AIRU. If Ableton is already open, AIRU will detect the correct sample rate, otherwise it will use the computer’s, which is 44.1kHz by default. I then put in earplugs and
begin running test sweeps using the spectrogram view in AIRU, which shows the amplitude of all frequencies in the recording by color from magenta (-100dB) to red (0dB) vertically, creating a visual of the recording horizontally through time. This functions as a soundcheck to ensure the microphone is not clipping and the AIRU settings are correct for this specific room. It also provides necessary insight into any external noise that may be bleeding into the recording. Once everything checks out I take and save five good recordings then close AIRU before moving over to Ableton. In Ableton I play back one of the sine sweeps monitoring the levels of the microphones to ensure none are clipping, then take a recording of each sweep, one at a time.

**Teardown**

After recording I then begin the teardown process, starting with turning off the speaker, then microphone phantom power, and finally the sound card. Wrap all cables and return to the Pelican case in the pre-set configuration.

**Post-Processing**

All post-processing should be handled off-site as time in the locations will often be limited. As mentioned previously, when using the sine sweep method for capturing IRs the recordings require post-processing before being usable. First the sweep must be deconvolved from the recording, de-noising the IR is recommended but not required, then the recordings must be converted from A-format to B-format using the BrahmaVolver. The BrahmaVolver only accepts two stereo pairs for the front and back capsules respectively
for convolving with the filter matrix and converting to B-format. Because AIRU creates 4 mono files, these must be converted into two stereo pairs using Ableton. Then, you are left with an ambisonic IR as a four-channel .WAV file. Because there are so many additional steps involved, and plenty of room for error, the following details my post-processing workflow.

**Extracting A-format from AIRU**

Before beginning, make a copy of all of the recordings and store them on a separate hard-drive as backup. First open Ableton to set the computer’s sample rate to 96kHz, then open AIRU. Review each of the AIRU recordings to find the one with the least noise. Once selected, deconvolve\(^\text{34}\) and denoise the IRs, in that order, using the “edit” menu. Cut off any silence at the beginning and use the “Energy” view to cut off any extraneous silence at the end. Having some silence at the end creates a smooth decay, however some IRs will have several seconds of silence, based on the settings and room size, which can be cut. Select File>Save As...\(^\text{35}\) and save the .SDIR file to the desktop. Right-click on it and select “Show Package Contents.” Here files 1, 2, 3, & 4.aif correspond to the impulse response for each of the mic capsules. Preview each of these files to ensure they are impulse responses (click with a reverb tail), not sine sweeps. If the files are sine sweeps ensure the deconvolution has completed in the AIRU and that you have saved the project. Copy these four files into a new folder labeled “A-format.” Rename each of the files with the name of the location, and position if multiple, and the mic capsule—1 = FLU, 2 = FRD, 3 = BLD, and 4 = BRU. For

---

\(^{34}\) Keyboard shortcut: ⌘D

\(^{35}\) Keyboard shortcut: ⌘S
example, “Hertz Hall Near FLU.aif.” Proper file management and organization is critical for handling these files as mixing up the capsules will render the ambisonic impulse responses useless. After the A-format files have been extracted from AIRU, organized, and renamed, they must then be prepared for use with the BrahmaVolver for conversion to B-format.

**A-to-B-Format Conversion**

The BrahmaVolver only accepts two stereo .WAV or .aif files (and from my experience it prefers .WAV files) so the four mono A-format files must be combined into two stereo pairs: front and back. The front pair consists of the FLU capsule in the left channel and the FRD in the right channel, while the back pair is BLD in left and BRU in right. Load the first pair, FLU and FRD, into Ableton with a separate track for each. Pan each track 100% left or right according to the middle letter of the capsule in the filename. Ensure no warping has been applied to the A-format files, nor effects applied to the tracks or master channel. Now listen to the audio to ensure the panning is correct and that no tracks are clipping. Once properly configured, render them into one stereo audio file using the menu option File>Export Audio/Video. Make sure to select WAV under the “File Type” dropdown in the Export Audio/Video dialogue box, check that the Sample Rate is set to 96kHz and the Bit Depth is 24, and set “Dither Options” to No Dither. Click export and follow the same file naming scheme used for the mono A-format files; for example, “Hertz Hall Near FLU + FRD.wav.” When the front pair have finished rendering check the stereo audio file to ensure it is correct. Repeat this process for the back pair.

---

36 Keyboard shortcut: ⌘⇧R
After the two stereo pairs are rendered create a folder named “Stereo Pairs” and copy them into it. Now open BrahmaVolver and drag and drop the two pairs into their corresponding slot under “Input.” You will see the field change from “Audio_file_0x.wav” to the directory of where your file is located. Make sure that the correct file is loaded into each slot. Select B-format from the Select Output Mode: dropdown at the top and “1 four channels file” from the How many files? dropdown. Name the file in the first slot under Output and click the large ⊗ next to BrahmaVolver in the bottom right to begin conversion. Once complete, a pop-up reads “Conversion done with success!!” The 4-channel B-format file will be located in the BrahmaVolver folder. If the program crashes immediately, check that the correct files have been loaded into the BrahmaVolver. If it crashes again try increasing the buffer size in the Preferences window, located under File>Preferences. If the issue persists try giving the stereo pairs very short names, such as “front.wav.” Finally, if it continues to crash try recreating the stereo pairs and try again. Once the conversion is complete rename your B-format file using the same naming convention used throughout, “Hertz Hall Near B-format.wav.” Finally, create a folder named “B-format” and add your 4-channel B-format file into it.

The same process can be applied for 5.1 by changing the Select Output Mode: dropdown to “5.1” then selecting “1 six channels file” and continue as above. In 5.1 mode two additional features are added, Opening Angle (deg) and Low Freqs in Lfe. It may be useful to create a version of the 5.1 IR with the Low Freqs in Lfe box checked. As before,

37 Keyboard shortcut: ⌘P
follow the naming convention and create a folder and add the 5.1 file(s) to it. Congratulations you’ve made your first ambisonic impulse response!38

File Management

Take all of the folders you’ve made—A-Format, Stereo Pairs, B-format, and 5.1—and put them in a folder labeled with the date of the recording and the name of the location; for example, “2017.03.31 - Hertz Hall.” If you took recordings of multiple positions within that location, follow all of the post-processing steps for each position and add additional subfolders respectively; for example “Long - 10m” and “Short - 4m.”

Usage

Now that you’ve created the B-format impulse responses, it’s time to use them. There are very few convolution reverbs that support ambisonics, and those that I found are for Windows only. Another possible solution is to decode them into binaural. Yet, many of the decoders are either source code or for Windows only. The plugin-in that appears to be the best at handling this on Mac is Ambi Head by Noise Makers. The software is €189 (~$200) and allows for decoding through several Head Related Transfer Functions (HRTF) which emulate the shape of different heads and ears to create a more realistic binaural experience. While this appeared to be a viable solution it is not a standalone application and must run in a DAW, but Ableton does not support multi-channel audio files, and it did not appear in Logic’s plug-in list, despite having been successfully validated by the plug-in

38 It is important to note that the BrahmaVolver only outputs 16-bit files and there is nothing in the documentation about how it handles downsampling from 24-bit.
manager. This presented me with an impasse and I was not able to further pursue decoding the IRs to binaural. Thus to continue using the same equipment some programming knowledge is required to overcome these limitations.

**Next Steps**

It is possible with MaxMSP to create a patch (software) that can handle the entire process of creating ambisonic IRs—playing the sine sweep, recording, deconvolution, denoising, normalization, cropping, combining into stereo pairs and exporting, convolving with the filter matrix, exporting B-format files, and auto-naming—as well as B-format to mono, stereo, binaural, 5.1, and 7.1 conversion. However, as far as I can tell there is currently no freeware software solution with all of these capabilities.

To then use the B-format IRs for composition and VR will require a DAW and convolution reverb that supports multichannel audio files. It seems this may be possible with Reaper, a DAW created by Cockos, however it was not explored in this project. MaxMSP could also be a solution by reverse engineering the *Convolution Reverb Pro* Max for Live patch, which comes with the Ableton Live Suite, and modifying it to work with B-Format. However, this option would not be usable in any conventional DAW other than Ableton, and would also require Max to run in the background because Ableton does not support multichannel audio files. For now a Windows computer or virtual machine, and/or programming knowledge may be required to take advantage of these recordings. But, hopefully a convolution reverb that handles B-format for OSX is right around the corner.
Results

Locations

A total of ten locations were recorded: The main room at the Center for New Music and Audio Technologies, The Hearst Memorial Mining Building, Hertz Hall, The Kroeber Hall stairwell, Woo Hon Fai Hall, and several rooms in Doe Library; Heyns Reading Room, The North Reading Room, Main Stacks, Morrison Library, and The Newspaper and Microforms Library.

Quality of files

Due to the issues mentioned in the Next Steps section, the IRs were not able to be tested to determine their quality or functionality. However, an issue with the recordings was identified during post-processing. Each of the FLU recordings had a much lower SNR than the other capsules. Troubleshooting proved difficult as the Brahma comes with very little documentation. Upon thorough hardware testing, and analysis of the recordings using iZotope Insight, it was determined the issue lies somewhere in the microphone, which was subsequently sent to Embrace Cinema for repair. Because the IRs could not be tested it is unknown if they are usable.

Issues Encountered

My hope for this section is to help people avoid some of the problems I encountered during this project.
The first issue was the late arrival of funding for the project. The funding was scheduled to arrive at the beginning of summer but did not arrive until the end of August. Because all of the locations selected for recording are on a university campus, after the summer semester many of the buildings open earlier and close later, with some remaining open 24 hours a day. Make sure to check for seasonal hours when scouting locations if you’re not able to begin recording right away. The Brahma is very sensitive and it is important to maintain the highest possible SNR for the clearest representation of the soundfield in the IRs. So, if your chosen location is in an urban area, there may be an increase in the number of people nearby at certain times of day and/or seasons, which may cause unwanted background noise, potentially ruining the recordings.

The next issue I encountered was with the Brahma microphone. None of the outputs on the breakout box were labeled and were not outlined in the documentation. This was easily solved by connecting only one XLR cable from the breakout box into the soundcard, then while wearing headphones, speak into each capsule individually, looking for the loudest one. This indicates which capsule is linked to the connected cable. Once found, mark its position on the microphone, label the breakout cable accordingly, and repeat this process until the location of each capsule, the front and back of the microphone, and all cables have been identified and labeled.

The next issue is that the AIRU does not support the vertical axis, Z, for B-format files. According to the engineering team at Apple, setting AIRU to quadraphonic, as mentioned in the Software section, functions as a workaround. This is because the AIRU does not apply spatial information when deconvolving; all spatial information is encoded in
the Space Designer preset. Another issue with AIRU is that it often crashes if the sample rate of the computer is changed while it is open. This is why I stressed using Ableton to set the sample rate first in the Methodology section.

Next, the BrahmaVolver software crashed frequently without any clear explanation when I first started using it. I determined that the software greatly prefers .WAV files over .aif, and that both files loaded into it should have the same extension. The files must also be of identical length in samples or it may crash as well. Also, after each crash the software loses the location of the filter matrices and the help document in the preferences window. To resolve this I kept these files in their default location according the preferences window. Lastly, the software frequently crashes when convolving longer audio files. To resolve this increase the buffer size in the File>Preferences>I/O options menu.

Finally, as mentioned several times above, Ableton does not support multi-channel files. I recommend when building your PAIRS to research further into DAWs that support multichannel audio files to simplify processing and testing.

**Future of Ambisonics**

Thanks to VR, ambisonics have seen a massive resurgence in the last 5 years, and 2016 might have been the biggest to date. In February, Noise Makers released their B-format to binaural conversion software *Ambi Head.* In April, YouTube added live

---

streaming spatial audio to their 360-degree video platform.\textsuperscript{40} In May, Facebook purchased Two Big Ears, a Scottish start-up that helps businesses incorporate spatial audio into their videos and games, and released a free workstation for designing spatial audio for 360 video and cinematic VR. \textsuperscript{41,42,43} Then in October they revealed Facebook VR,\textsuperscript{44} which virtually puts users together using 360-degree video and audio. The same month, Sony Computer Entertainment Interactive (SCEI) released PlayStation VR, a VR headset for PlayStation 4.\textsuperscript{45} November was a huge month for VR. Røde Microphones, a world-leading Australian-based microphone manufacturer founded in 1967, purchased Sound Field, one of the first ambisonic microphone manufacturers.\textsuperscript{46} Around the same time,\textsuperscript{47} Sennheiser, a German microphone manufacturer founded in 1945, released their first soundfield microphone branded as the “Ambeo VR.”\textsuperscript{48} Google released DayDream, a new VR headset and software. \textsuperscript{49} Even Embrace Cinema, makers of the Brahma Ambisonic Microphone used in this project,

\begin{thebibliography}{9}
\bibitem{TwoBigEars} “We’re excited to announce that we’re joining Facebook!” \textit{TwoBigEars.com}, last modified May 23, 2016, \url{http://www.twobigears.com/index.php}
\bibitem{Vanian} Jonathan Vanian, “Facebook Just Bought This Hot Virtual Reality Startup,” \textit{Fortune.com}, last modified May 23, 2016, \url{http://www.fortune.com/2016/05/23/facebook-just-bought-this-hot-virtual-reality-startup/}
\bibitem{Zuckerberg} Mark Zuckerberg, “Here’s the crazy virtual reality demo I did live on stage at Oculus Connect today,” \textit{Facebook.com}, October 6, 2016, \url{https://www.facebook.com/zuck/videos/10103154531425531/}
\bibitem{Rode} Rode Microphones, “Freedman Electronics Purchases Soundfield,” \textit{Rode Blog}, November 21, 2016, \url{http://www.rode.com/blog/all/freedman-electronics-purchases-soundfield}
\end{thebibliography}
have added two new products since this project began: the Brahma Compact Standalone\textsuperscript{50} and Brahma Large Diaphragm\textsuperscript{51} Ambisonic Microphone. Finally, at the end of the the year Google, HTC, Oculus, Samsung, Sony, and Acer joined forces to create the Global Virtual Reality Association (GVRA) to begin standardizing the medium.\textsuperscript{52} With these big investments in hardware and software we can assume 2017 will be another huge year for spatial audio.

Just this March Vimeo released support for 360-degree videos\textsuperscript{53} with support for spatial audio coming soon.\textsuperscript{54} Occis X, mentioned in the introduction, begin shipping their first wave of headphones this May.\textsuperscript{55} Just two days ago\textsuperscript{56} Google acquired Owlchemy Labs, one of the most successful VR game studios.\textsuperscript{57} The International Data Corporation (IDC), a provider of market intelligence for information technology, telecommunications, and consumer technology, project a nearly 200\% increase in revenue annually for VR until 2020.\textsuperscript{58} With VR a household name, and quickly becoming a household product, ambisonics

\begin{itemize}
\item \textsuperscript{52} Lucas Matney, “Google, HTC, Oculus, Samsung, Sony join forces to create Global VR Association,” TechCrunch.com, last modified December 7, 2016, https://techcrunch.com/2016/12/07/google-htc-oculus-samsung-sony-join-forces-to-create-global-vr-association/
\item \textsuperscript{56} May 10\textsuperscript{th}, 2017
\item \textsuperscript{57} Relja Markovic, “Welcoming Owlchemy Labs to Google,” The Keyword (blog), May 10, 2017, https://blog.google/products/google-vr/welcoming-owlchemy-labs-google/
\item \textsuperscript{58} “Worldwide Revenues for Augmented and Virtual Reality Forecast to Reach $162 Billion in 2020, According to IDC,” IDC.com, last modified August 15, 2016, http://www.idc.com/getdoc.jsp?containerId=prUS41676216
\end{itemize}
is soon to follow, but most likely by the name “spatial audio.”. It seems as though ambisonics has finally found its home in VR and is here to stay.

With ambisonics coming to the fore, I hope this document serves as a gateway for video game designers, film makers, music producers, composers, and software engineers to explore the potential of this new medium. VR and 360-degree video will be the next major medium, but it is lacking in realistic depth of space. I believe ambisonic impulse responses are the solution to creating more realistic sonic environments. But to get to the the point of consumer integration there must be ease of access and use, and there are a lot of areas available for improvement. While ambisonics is a new sonic terrain that’s yet to be fully explored, I hope this map serves as a guide leading to new directions of sound.
Bibliography


Presentation at the Audio Engineering Society 40th international conference, Tokyo, Japan, October 8–10, 2010.


Farina, Angelo. “A-format to B-format conversion.” Angelo Farina’s Personal webpage.
Rahman 38

Last modified October 18, 2008.

http://pcfarina.eng.uni-pr.it/Public/B-format/A2B-conversion/A2B.htm.


http://content.time.com/time/specials/packages/article/0,28804,1953425_1953424_1953506,00.html


http://www.popsci.com/bespoke-360-audio-how-ossic-x-headphones-works

http://www.radio.uqam.ca/ambisonic/basic_ambisonics.html.

http://doi.org/10.1016/j.apacoust.2013.02.004.

https://techcrunch.com/2016/12/07/google-htc-oculus-samsung-sony-join-forces-to-create-global-vr-association/


Noise Makers. “Ami Head released : Ambisonics to 3D binaural by Noise Makers.”

2015.


http://content.time.com/time/specials/packages/article/0,28804,1953425_1953424_1953359,00.html


http://content.time.com/time/specials/packages/article/0,28804,1953425_1953424_1953399,00.html


Rahman. 42


https://wiki.xiph.org/Ambisonics.


24.


http://www.idc.com/getdoc.jsp?containerId=prUS41676216