Making Musical Magic Live

Inventing modern production technology for human-centric music performance

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Abstract

Fifty-two years ago, Sergeant Pepper's Lonely Hearts Club Band redefined what it meant to make a record album. The Beatles revolutionized the recording process using technology to achieve completely unprecedented sounds and arrangements. Until then, popular music recordings were simply faithful reproductions of a live performance. Over the past fifty years, recording and production techniques have advanced so far that another challenge has arisen: it is now very difficult for performing artists to give a live performance that has the same impact, complexity and nuance as a produced studio recording. Live performance production technology is now used almost exclusively to recreate studio albums exactly as they were recorded. Recently, this approach has dominated the entertainment industry. In an attempt to reach superhuman levels of perfection and complexity, many elements that make live performances emotionally meaningful for audiences have been given less priority — or lost altogether.

The mission of the work described in this dissertation is to reverse this trend by investigating methods of integrating technology and live music performance in such a way that the technology allows for flexible musical expression, sound and connection to the audience, while still enabling exciting, sophisticated and "magical" production values. This dissertation identifies six objectives for the human-centric design and integration of technology in musical performance, and a methodology to support each objective. These have been developed, refined and tested with artists and performers through a series of ten large-scale projects and approximately 300 individual performances. Through this work, I demonstrate that it is possible to combine high-value production with interactive musical performance. We are now on the cusp of redefining live musical performance production as an art form just as Sergeant Pepper's Lonely Hearts Club Band redefined studio album production as an art form fifty years ago.

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0 Introduction

Fifty years ago, *Sergeant Pepper's Lonely Hearts Club Band* redefined what it meant to make a record album. Instead of capturing a simple live performance, the Beatles revolutionized the recording process using technology to achieve sounds and arrangements that were completely unprecedented. This was a major development, since until then popular music recordings primarily consisted of verbatim captures of a live performance. With *Sergeant Pepper's Lonely Hearts Club Band,* many new methods were utilized, such as non-traditional microphone placements, tape manipulation, feedback and unorthodox use of echo chambers.[1] Hence, the concept of music "production" was born into the mainstream, and the album really developed into its own art-form. Many artists began to write for the studio, rather than the stage, as the creative possibilities were vast compared to the constraints of live performance. In fact, these studio productions were something like an 'antidote' to live performance: intentionally crafted with techniques that were impossible to achieve in concert, with no intention of ever being performed live. The Beatles never did perform *Sergeant Pepper's Lonely Hearts Club Band* in front of an audience.

Over the past fifty years, recording and production techniques have become so advanced that another challenge exists: it is now quite difficult for music artists to give live performances that have the same impact, complexity and nuance as a studio-produced recording. To approximate the feeling of produced recordings in a live context, many new techniques were developed and refined. Bands would add additional musicians and change instruments rapidly. Multi-modal elements such as lighting and projection were pioneered as well, synchronized to the musicians. Thus, the field of "live production" was born. Despite these developments, the simplest means of performing a complex album live was to play the album on very large speakers and ask musicians to play along. Over time, this most basic approach has dominated the entertainment industry. Even in classical music there is a strong focus on sounding "like the album," which may include artificial edits and corrections that make the resulting "performance" an unattainable goal. It is ironic that fifty years after *Sergeant Pepper's*, there is now a push in the opposite direction: live production technology is most often used to recreate studio albums exactly as they were recorded. In trying to reach these superhuman levels of perfection and complexity, many elements that make performances emotionally meaningful for audiences are pushed aside or lost all-together.

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[1] Emerick, Geoff. Here, There and Everywhere: My Life Recording the Music of The Beatles. Gotham Books, 2010.

Timecode: A method of synchronizing many elements of a performance with exact timing using a clock.

Backing Track: A pre-recorded set of sounds on record, tape or other medium, played along with a performance to account for musical elements that cannot be performed live by the musicians on stage.

In preparation for writing this dissertation, I have spent some time going through live music recordings and asking my colleagues, friends and family about their favorite live performances. It seems that among my friends and colleagues, many of the favorite, most iconic live performances aren't from recent years. Famous Michael Jackson and Prince Superbowl half-time shows, and other remarkable live music acts are all from the 70s and 80s, with a few in the 90s. There are very few after that. For a long time I've had a theory that this is partially due to the way live shows were produced in those years versus how they are made today. In that era, people were responsible for activating all the cues for lights, sound, and other effects, so there could be flexibility and spontaneity in their performance. In the early 90s there was a large transition to running productions using backing tracks and timecode instead of letting the band play freely. Although artists were actually playing real instruments, there was no flexibility in the sound or the timing. In some cases musicians would even mime along to recordings without actually playing. As a result, many recent performances by popular touring artists are clones from night to night, with very little room for the artist to improvise or react musically or emotionally to the audience. This lack of flexibility means that no one performance is special, so there is no reason to remember any single night as a uniquely inspiring experience .

I do not mean to suggest that there are no artists performing today with flexibility and meaningful audience interaction. However, where we do find interesting musical performances now (ex. symphony orchestras, looping artists, operas, jam bands like Snarky Puppy), the level of production is demonstrably simpler than major live touring pop acts. One reason timecode has become so popular is that it allows for massively scaled, astonishing live effects to follow music very closely. These effects are akin to visual effects in film, with audiences now taking the most incredible feats for granted (ex. Kanye West flying around a stadium on his own stage with projection and lighting following him[2]). But as an industry, the entertainment business has prioritized these effects, and they can have a numbing effect on audiences, requiring bigger and bigger budgets and spectacles to make any kind of impact. The mission of this dissertation, and of all my work, is to change this trend, and to investigate methods of integrating technology and live music performance in such a way that the technology allows for flexible musical expression, sound and meaningful connection to the audience, while allowing for exciting, sophisticated and magical production values. Through this work, I hope to demonstrate that we are on the cusp of redefining live musical performance production in the same way *Sergeant Pepper's Lonely Hearts Club Band* redefined the record album fifty years ago.

[2] Knopper, Steve. "How Kanye West Made His Saint Pablo Stage Fly." Rolling Stone, 25 June 2018, www.rollingstone.com/music/music-features/how-kanye-west-made-his-saint-pablo-stage-fly-101150/.

My work on technology and live music performance builds on previous work by the Opera of the Future group, known as ***Hyperinstruments***.[3] These are a category of musical instrument which utilize technology to intuitively enhance and extend the expressiveness of a musical performance, providing nuanced control of additional musical layers and textures. In more recent work, such as Machover's *Death and the Powers*[4] and *Spheres and Splinters*[5], the Hyperinstrument concept was extended beyond sound and music to other media, with a single instrument able to provide an artistic interface for a musician to control visuals, lighting and spatial choreography in addition to sound. The work for this dissertation proposes that we can grow the Hyperinstrument concept even further to encompass multiple instruments, performers, audiences, and systems, resulting in nuanced articulation of completely interlinked performance environments. This type of experience is capable of providing the impact and spectacle of massive pop production, while being truly connected to the music, the performance, and the people present both on and off of the stage. It leaves room for musicians and performers to improvise and react to the audience more thoughtfully. Through the work in this dissertation, Hyperinstruments are ready to grow in scope, and move out of the laboratory and into the mainstream.

Hyperproduction is the name of the software that I created for my masters thesis to provide a computational means of representing and enabling these larger technology-based performances.[6] This system is capable of easily controlling many kinds of infrastructure, and it provides a user interface for both conceptually modeling and articulating space, performers, audience, interactions, analysis, and control systems. With this dissertation, I would like to expand the definition of Hyperproduction to include more than just the software, but rather to encompass the entire class of live production that connects people and technology on a large scale. The work documented in the following chapters forms the basis for a set of core objectives and respective methodologies for developing Hyperproductions:

[3] Machover, Tod: Principal Investigator . Hyperinstruments - A Progress Report 1987 - 1991 .MIT Media Laboratory. January 1992.

[4] Torpey, P. "Digital systems for live multimodal performance in Death and the Powers." International Journal of Performance Arts and Digital Media, Volume 8, Number 1. 2012.

[5] Kinoshita, J. "Spheres and Splinters – New Work by Tod Machover." Opera of the Future, 21 Nov. 2010, operaofthefuture.com/2010/11/20/spheres-and-splinters-new-work-by-tod-machover/.

[6] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (pg. 71)

1. **Universal Connection of Production Systems and People** is an objective based on the idea that components which are interconnected following the principles of Hyperproduction may be choreographed as a single entity. In many of the productions described in Chapters 2 and 3, the entire environment behaves like an instrument and relies on the unique strengths of machines and humans combined with very specific roles in order to enable special behaviors which would not be possible otherwise.

2. **Designing structured systems for flexible and fluid performance** is an important objective based on the idea that the application of technology to "creative" and "fluid" artistic experiences requires the conception of strict organizational structures. Flexibility is born through structure. Examples in Chapter 3 and 4 will demonstrate how experiences that an audience may perceive as improvised or organic are made of building blocks that can impose strict behind-the-scenes rules but ultimately enable more expression and flexibility for the artists.

3. **The choreographing of live surround sound** became a key objective and creative medium for many projects in Chapters 3 and 4. Through these projects, I was able to re-examine many tropes of traditional sound-design. Rather than making live sound experiences replicated identically for every listener, I focus on the goal of constructing worlds that are experienced slightly differently by each audience member but foster a greater sense of realism, connection and shared experience.

4. **Defining and supporting "Liveness"** is an objective that is especially important for remote or personal experiences that are mediated by technology. We must ask ourselves why it matters for an experience to happen at a specific time, and what exactly makes a live performance different from a recording, especially when distributed over the internet or experienced individually in physical isolation. The Powers Live and Sleep No More projects in Chapter 3 discuss these creative challenges in detail.

5. **Bringing content faithfully from studio to stage** in an objective that helps to ensure work translates properly from the mind of the artist to much larger spaces. There are technical and creative strategies for this, but also many logistical and organizational challenges that are similarly impactful to the end product. These are detailed in Chapter 4v, where I discuss exactly what was required to bring Jacob Collier's YouTube and studio creations to life as touring stage productions.

6. **Designing impactful sound systems for large ensembles and spaces** is the final objective that attempts to form a path from the creative aspirations of a work to the fundamental components needed to reproduce and support the work in production. Large ensembles and spaces have specific properties that require specific equipment, hardware, software and tools. These are detailed in Chapters 2, 3 and 4 throughout the description of the many projects that have taken place.

The work in this dissertation is a comprehensive look at these objectives and the formal documentation of a set of methods to achieve them. My hope is that it opens the door to future investigation and experimentation. I am extraordinarily excited to see Hyperproduction become its own art-form, eclipsing older live production techniques, and creating experiences that are truly interactive, bringing much more connection and excitement than could be possible from a recorded album. The ingredients to achieve this are all present, and for the last 12 years at the Media Lab, I have completed projects which require fashioning new tools, reassembling diverse pre-existing tools, and shaping integrated systems of these tools in collaboration with performers and creators to produce "magical" results. What follows is a careful reflection on many of these projects, along with a record of some of the concepts that have become important throughout this work.

Throughout the rest of the document, I will notate each objective using colored discs and a short description in the side margin where it is applied in each project.

1 The Evolution of Instruments, Performance, Production and Hyperproduction

Traditional professional music performances are centered around instruments that were created hundreds of years ago and then refined over time. An artist would typically compose a piece while subjected to the constraints imposed by a collection of traditional acoustic musical instruments. In recent decades, advances in technology have now removed these traditional constraints. Early electronic music instruments allowed a variety of sounds and textures to come from a single device; no longer did the physical constructions of an

More detailed description of Hyperinstrument systems can be found in section 2.3.

instrument dictate its sound. Many of these instruments, such as the Moog synthesizer, made sounds that were disconnected from the physical gesture of playing. Unlike an acoustic instrument, a tiny gesture could produce a massive sound, or vice versa. The same gesture might even be able to produce a large number of different sounds. In contrast, Hyperinstruments are instrument-technology systems that enhance performance ability using additional sensors and a computer to intentionally connect sound and gesture.[7] By analyzing the sensor data in real time, they significantly extend the expressive and creative potential of the performer by enabling responsive and intuitive interactions beyond traditional musical techniques. No knowledge of computer operation is required to play a Hyperinstrument— they use natural interfaces that are already familiar. This form of instrument encourages the conception of composed works in terms of gesture, experience and emotion, rather than performance techniques limited by traditional instruments, orchestration or even the musical skill level of the performer.[8] Where a composer would have previously notated dynamics and articulation, he or she is now able to give an abstract depiction of desired behavior or feeling with minimal specification of implementation. In recent years, these instrument-technology systems have become widely common in the mainstream production world,[9] but due to their complexity, an emotional or behavioral specification from the artist is generally insufficient to thoroughly define the nuances of an instrument's operation. It is these nuances that give an instrument its character, emotional impact, and expressiveness. So, the instrument builder must shoulder a significant creative responsibility to communicate the vision of the artist.

Today, the idea of the instrument has expanded such that we now design entire performances, tours, and experiences using the same philosophy and process as Hyperinstruments. For these more substantial works, we have coined the term Hyperproduction, which encompasses not only traditional instruments and technology, but people and logistics as well. The nuanced character and expression of these performances depend on the interaction of complex organizations of people and technologies. While previously off-the-shelf production technologies— lighting desks, mixing consoles, midi controllers, synthesizers— could be used to create performances, this new class of Hyperproduction requires the conception of new roles, workflows, software, hardware, and

[7] Machover, Tod: Principal Investigator . Hyperinstruments - A Progress Report 1987 - 1991 .MIT Media Laboratory. January 1992.
[8] Rowe, Robert. Interactive Music Systems Machine Listening and Composing. MIT Press.
[9] Hermann, Andy. "Ableton Live Expert Laura Escudé on Being Behind the Sounds on Music's Biggest Tours." Billboard, 8 Nov. 2018, www.billboard.com/articles/news/8483852/ableton-live-expert-laura-escude-behind-the-sounds-musics-biggest-tours.

organizations of people. The combination of these elements forms a large, multi-dimensional "instrument" that is used to perform the piece imagined by the artist.

In developing new approaches and designs, each new Hyperproduction becomes an experiment. We are given the chance to try something different, and we can then look at how the audience, performers, technicians and creators react, as well as how the technologies and organizations facilitate these reactions. This dissertation describes a series of Hyperproduction experiments, each providing insight into a different approach to creating and enhancing music performances with technology. They are simultaneously a body of work, a philosophy and a set of guidelines for working, and a suggestion for the next hypothesis to test with new work. The experiments come from two overarching performance efforts. The first covers projects in Prof. Tod Machover's Opera of the Future Group at the MIT Media Lab, focusing on contemporary classical operas and orchestral pieces. The second follows the process of designing the instruments and performances for Jacob Collier, a Grammy-winning London-based recording artist, including his *One-Man-Show* and *Djesse World Tour*. Together, these two efforts encompass ten distinct projects and well over ten years of experimentation.

In documenting these, I do not intend to dictate absolutely how one should make a Hyperproduction, or even how one should put on a show. Instead, my hope is that the lessons and ideas demonstrated here will provide a fresh perspective for those putting together new multisensory musical work— that this will help light a path to the successful integration of technology, music and live experience in a way that is more emotional, more meaningful, more accessible, and more human.

2 Background

To imagine and explore a new future, it is important to understand the past. In this section, we will look at typical approaches to live musical production and performance — techniques, workflows, and organizations of personnel that are traditional in the world of production — as well as at some current research which has challenged and stretched existing approaches. It is important to recognize that there has been significant academic interest in the area of experimental live music production since the early 1950s. We must keep in mind that many experimental techniques never get to leave the laboratory. For this dissertation, we are primarily concerned with techniques that have been used in large-scale, professional, internationally-recognized projects. We learn a significant amount by testing operating principles in real-world scenarios, and this testing informs the development of new work more completely. There is a difference in style and scope of deployment between Prof. Machover's and Jacob Collier's productions, which will be discussed in detail. Productions for Prof. Machover are often developed experimentally, tested and refined in a lab setting, and then rehearsed and performed very few times at scale in a fully professional setting. Each project and performance varies quite substantially from all existing projects, as new compositions, ideas and processes are refined. On the other hand, Jacob Collier's productions are often developed very quickly, in professional studios or rehearsal environments, borrowing many elements from his existing work. We then tour the production, in many cases repeating it in live performance over 200 times. On tour, the performance has a chance to settle in and elements of it change very slowly over time. We also gain a deep understanding of which elements are compelling to the performers and audience. Both of these deployment cycles are crucial to evaluating this work. The theory of the work is well developed, but real-world practice is necessary to evaluate and iterate. The background in this chapter will therefore focus on real-world examples, covering how professional production has been achieved in the past and is currently achieved today.

2.1 Existing Production

Portions of this background are adapted from my Master's Thesis[10] since the concepts were originally important in the development of Hyperproduction software, and continue to play a role in this work. The sections below detail existing techniques, roles and processes for typical production.

2.1.1 Techniques

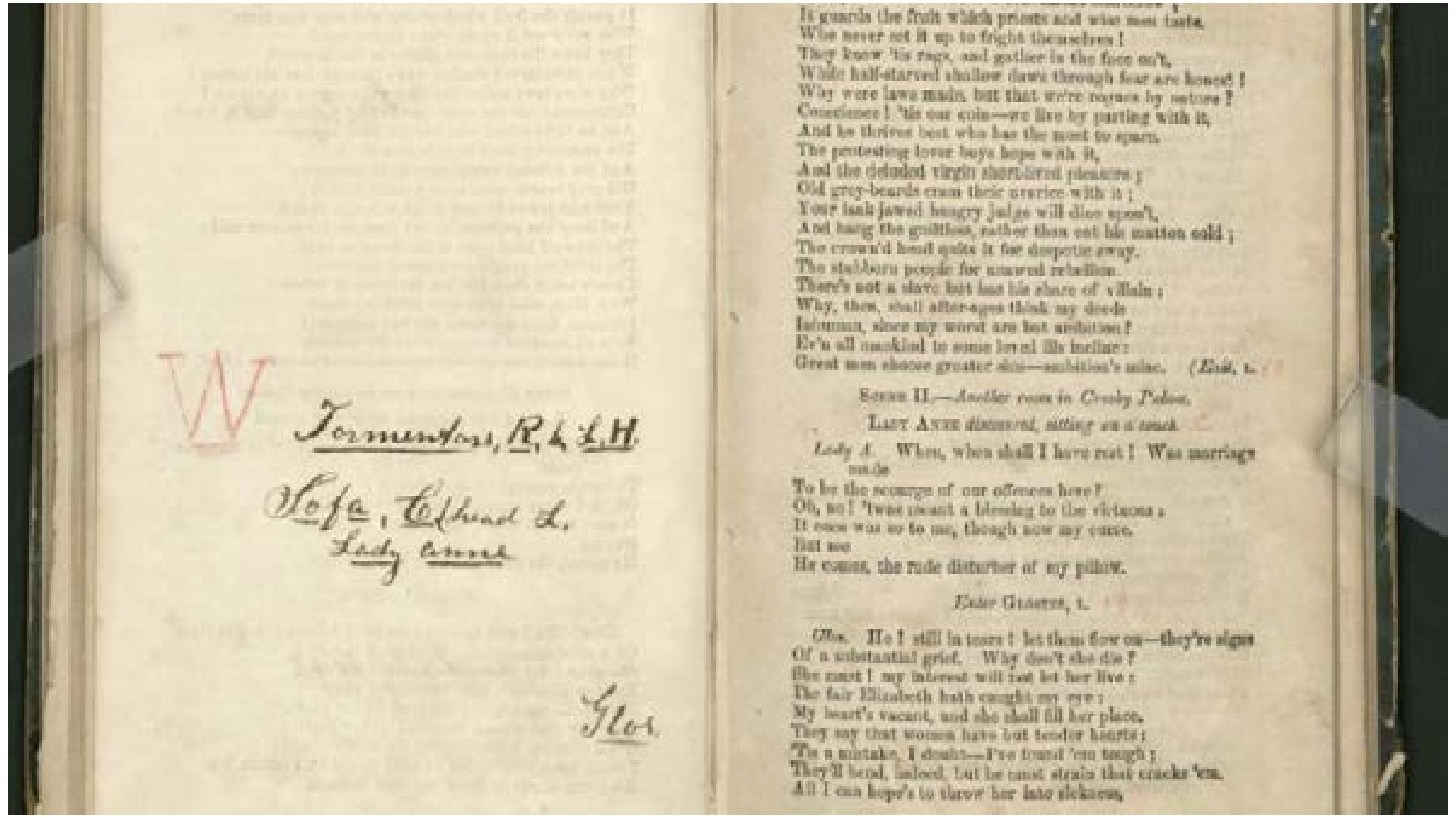
Although there are many musicians playing on stage for most (if not all) musical performances, much of what you may hear is not live. A large portion of production systems are triggered directly by a human operator using an organizational technique called cues. This type of trigger originated because it is simple to communicate the start of a cue verbally (i.e. shouting "go!" or whistling[11]) or by a hand gesture. Cues generally are comprised of small sequences of choreographed movement or changes in production elements. These sequences are triggered

[10] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (pg. 25-37)

[11] Colleary, Eric. "Why You Should Never Whistle Onstage." Playbill, PLAYBILL INC., 20 Jan. 2018, www.playbill.com/article/why-you-should-never-whistle-onstage.

("called") by a stage manager, or another operator, and can be assembled in any order and refined in isolation. It gives a natural organizational structure and allows many complicated components to happen simultaneously. In modern large-scale productions, cues are used to choreograph moments when the synchronization of human performance and technical production elements is critical. When a cue is called, the choreographies of the respective elements are run to completion, and the performers on stage must follow the timing as closely as possible.

Fig. 2-1: A "W" indicates a whistle to bring in the masking curtains during a scene change in the promptbook for John Wilkes Booth's production of Richard III, ca. 1861. Courtesy *Harry Ransom Center, The University of Texas at Austin*



Cues represent synchronization points where many elements must occur in perfect synchronization. But after a cue is called, elements run on their own timeline, sometimes at slightly different speeds. For example, the combination of a set change made by humans, a lighting fade to black, and a recorded sound cue may drift in synchronization even if all three are started at the same moment. One would think that in 2019, it would be possible to have systems run perfectly aligned, but even minute variations in speed can result in “drift” that causes systems to end up offset after a period of time. With the advent of motion-picture with sound, a new synchronization standard called “timecode” was developed for completely aligning multiple playback systems, rather than just the starting point. Timecode was first introduced in 1967 by the EECO company to synchronize tape based video editing systems, so that the edit points, picture and dialogue could remain locked together continuously. The protocol was standardized by the Society of Motion Picture and Television Engineers in

Fig. 2-2: EECO Video Character generator for burning timecode into analog video signals.

1975 as SMPTE 12M.[12] [13] The mechanism for tagging frames ensures audio and video do not drift out of sync over time because the playback speed is kept identical between systems and regulated so any change is propagated from one to the other.

This technique became a popular tool for live performance as early as 1983 for David Byrne’s *Stop Making Sense* performance.[14] [15] For live *music* performances using timecode, often the drummer is given aural or visual cues to follow the tempo exactly. In other cases, a conductor is responsible for listening to both a backing track and a live orchestra, and must try his or her best to keep the two closely aligned. Another common arrangement is for a stage manager or musician to trigger preset materials, by starting a tape machine or a projector at a specific moment to attempt to align them with the live performance.

These methods of control have had a strong influence on live production since their introduction. It is these timecode and cue systems that are the oldest and the most robust. When it comes to large-scale, high budget productions, producers are loath to trust any systems that are not well tested and proven to be reliable, since failure cannot be tolerated. Because these two types of control are the most common, much effort has been spent conforming humans’ ability to perform to work within the bounds of their unique requirements.

With a stage manager calling cues, it is possible to achieve compelling and nuanced timing. In this case, the performers focus on fostering an emotional connection with the audience, taking arbitrary amounts of time, and the stage manager closely follows their actions to trigger each element at the right moment. On the other hand, timecode allows an experience to be refined and perfected like a film or album because each run of the system is identical. It is possible to have hundreds of separate components perfectly arranged together and complexly intertwined, but there is no flexibility in timing to maximize emotional connection. Both methods of control provide a great deal of capability for live systems and experiences, but they each have strengths that do not overlap: timecode gives precise synchronization of many elements, but with rigid timing, and cues allow more fluid

[12] ST 12-2:2008 - ST 12-2:2008 - SMPTE Standard - For Television - Transmission of Time Code in the Ancillary Data Space - SMPTE Standard, ieeexplore.ieee.org/document/7290702.
[13] Wirth, Richard. “SMPTE Time Code - Virtually Unchanged After Almost 50 Years by Richard Wirth.” ProVideo Coalition, ProVideo Coalition, 3 Nov. 2014, www.provideocoalition.com/timecode-virtually-unchanged-after-almost-50-years/.
[14] Hamilton, Jack. “What Do You Call a Machine That Hangs Out With Musicians?” Slate Magazine, Slate, 5 Dec. 2013, slate.com/culture/2013/12/beat-box-review-drum-machine-gets-its-due-in-joe-mansfield-book.html.
[15] Stop Making Sense [Motion Picture] United States: Cinecom Pictures, Demme J. (Director)

timing but are difficult to reproduce and rehearse complex sequences. This leaves us with a gap in capability that is challenging to bridge.

2.1.2 On the pursuit of “Liveness”

Opera of the Future Group: Professor Tod Machover’s research group at the MIT Media Lab, originally called *The Hyperinstruments Group*, of which I am a part and for whom this dissertation was written.

With this work, I am investigating the right way to build a bridge between these capabilities. The perfect synchronization that timecode offers is inherently mechanical and less nuanced, and even a musical performance by many members of a live orchestra can never approach this level of precision. This is due to the laws of physics: the acoustic delay from one side of an orchestra to another is generally 60-100ms. Even while perfectly following the conductor, it is impossible for every player to be as aligned as the edits on a record, which are often carefully sculpted down to the millisecond. Since our ears are familiar with the sheen of a perfectly edited music recording (orchestral and pop alike), we react negatively to the same musical perfection if experienced live. This is because we sense immediately that such a perfect performance must be artificially augmented somehow, so that it is disingenuous, and this has a way of emotionally distancing us from the performers.

Technological systems default to extreme precision, since any imprecise behavior of technology must (ironically) be defined precisely. This is a fundamental challenge when adding technology to human-centric endeavors, typically solved by adding some element of uncertainty, randomization or other elements of chaos to the system. However, this is not a complete solution and does not truly capture what makes a performance feel human. There are many other elements that make a live performance feel special. Many of these come down to the mannerisms of the performer, but the technology and the show design can have a strong impact on this sense as well. Audiences have a keen sense of what is real, emotionally, in terms of stunts and gimmicks, and in terms of talent and performance. It is immediately evident if technology is too slick and glosses over all the human imperfections, or makes the performer seem like a show-off, or keeps the performer occupied so he or she cannot consider the fact that there is an audience present (let alone acknowledge or interact with them). As soon as these things happen, the audience members feel as though they are watching something disconnected from them, more like a recording than a performance. Synchronization and precision play a large part in this phenomenon and must therefore be managed quite carefully.

We have so far determined that cue-based solutions may not allow enough synchronization or complexity, but timecode is too perfectly synchronized; so what is in between? *Opera of the Future* productions address the challenge of building technology that retains human nuance by discarding timecode and instead building systems that are heavily influenced by sensor driven measurement. By attaching the control of production elements to live measurement of the performers, a system can be synchronized *just the right amount* while maintaining flexibility, since the system is actually driven by human behavior. We will discuss this in detail in the documentation of two projects in particular: *Death and the Powers* and *Fensadence* in Chapter 3. Towards the end of the dissertation we will also see how it is possible to do this even without using sensors for the *Djesse* World Tour. While these projects are a deep investigation of this concept, the idea of incorporating subtle nuance from humans directly to achieve musical connection at production-scale (as opposed to instrument-scale) is a core philosophy of every project documented in this dissertation, and it represents a fundamental contribution of this work to the field of professional live production.

2.1.3 Roles

Along with production techniques, production roles have also settled over time. Each discipline has its own set of titles and hierarchy of authority. Generally, composers, directors, and designers hold the most highly regarded creative positions. In recorded music, the producer also has an important role. These “creatives” typically imagine the intent of the piece broadly through gesture and artistic expression. This may include writing music, coordinating action and perspective for storytelling, choosing how elements will look and feel and sound, or picking the texture and arc of an album or song. The roles involve the conception of new material directly from the imagination of the creator of the work; so this work is held in high regard throughout the world.

At the same time, a wide range of support roles (engineers, electricians, builders, programmers, animators, etc.) exist to help the creatives execute their vision. These supporting technicians perform physical, conceptual, and virtual acts of construction to turn a work into reality. For music, theater, and other time-based media, these roles also help the execution of the performance to meet the intent of the creatives. There is generally a large separation of status between those who conceive and those who execute. A number of organizational roles (production managers, show producers, etc.) facilitate communication, action, and setting of expectations between the technical and non-technical staff. Some translation is required to help the artist specify enough detail for the technical team to formulate an accurate design and to ensure the technical designs will meet the creative expectations of the artist.

There is another important class of roles: the performers. The role of a performer is generally non-technical, but involves some artistic interpretation of the intention of the work of the creator to bring a human element to the live execution. Performers are regarded highly, like creatives. They provide the human-emotional attachment for audiences and must be able to improvise and share themselves in a way that is vulnerable to evaluation and criticism.

2.1.4 Process

Techniques have evolved to allow teams to efficiently put together a production once it is fully designed. Rehearsals are typically structured to allow autonomy between departments so that the many pieces of a performance can be independently improved and rehearsed, without significant dependency upon one another. As productions become larger, the difficulty of having full-scale rehearsals drastically increases. So, it is critical to be able to refine parts in isolation so they can be assembled later.

In practice, this results in there being very little time for technology to be tested. For large performances, the first run through with technology must be essentially perfect. Much time is taken before creatives are present to refine the technology in isolation in order for there to be no stops or required changes due to misbehaving technology. Because of this arrangement, many systems are made to be programmed and designed in isolation without the features that would make them flexible in a typical rehearsal scenario. For example, many are limited in the ability to start and stop at arbitrary times; they can only be stopped and then reset to the beginning. This is sometimes due to technological limitations, but it is also frequently due to safety considerations. For robotics or moving scenic elements, engineering the ability to jump backwards and forwards on a programmed timeline is an extremely complex task, as the automation system must be able to track the show carefully and certain actions are only one-way executable (e.g. confetti cannot be "undropped"). For long sequences, these inflexibilities can result in substantial time inefficiencies as the entire team must completely reset a system to refine a small segment within a larger piece.

When developing new technologies for live performance, it is important to consider the creative momentum of the rehearsal process. There is a concept of flow, defined by Csikszentmihalyi[16] as when "people typically experience deep enjoyment, creativity, and a total involvement." Often it takes time for one to achieve a creative flow, and any sort of distraction, especially by technology, can impede one's ability to generate creative ideas.

[16] Csikszentmihalyi, Mihaly. Flow: the Psychology of Optimal Experience. Harper Row, 2009.

For systems to be truly useful in creative environments, we must consider all the patterns that can take us away from creating, and try to remove these. Many of the systems documented in future chapters have special features to offer functionality without the need for flow stopping behaviors such as reboots, excessive waiting, non-idempotent operations, and compilation. Eliminating these behaviors allows a system to "rehearse well," and it is therefore no longer necessary to refine elements in isolation. Instead, the entire piece with all of its elements can be rehearsed together, which is where interesting new interactions can be discovered and refined. Therefore, the rehearsal process is documented for each project in the coming chapters. Wherever possible we attempt to combine systems early on to get a sense of how rehearsal will flow, and what interactions are possible.

2.2 Composition and Sound Design

Conservative descriptions of the act of music composition such as those by Hindemith[17] constrain the art-form to the use of concepts from classical western music theory (ex. notes, chords, melodies, counterpoint, harmony, etc.) The sound of classical compositions was refined through the design of instruments and halls. Slowly over time, the characteristics of both evolved to the instruments and venues we have today. Members of large orchestras are now trained from an early age to listen for balance and timbre, and in both, to strive for very particular requirements. Conductors have the responsibility to listen to an ensemble and maintain a balance by communicating offsets to the players who cannot hear the full relationship of levels between instruments. These balances, timbres and orchestrations are all notated in detail in the composition, and there is a general understanding of how an ensemble should sound in a concert hall.

More recently, the writings of Ross[18] expand the definition of composition to include methods which go well beyond classical harmony and counterpoint. These include algorithmic composition and techniques such as serialism, pioneered by Schoenberg, Stockhausen and others.[19] These methods focus on gesture and texture through their use of twelve-tone rows, instead of classical harmony and counterpoint. This requires the application entirely different sonic strategies, balance and aesthetics. In the 1970s, Grisey was one of the composers

[17] Hindemith, Paul, and Arthur Mendel. The Craft of Musical Composition. Schott & Co., 1948.
[18] Ross, Alex. The Rest Is Noise: Listening to the Twentieth Century. Harper Perennial, 2009.
[19] Grant, M. J., and Imke Misch. The Musical Legacy of Karlheinz Stockhausen Looking Back and Forward. Wolke V.-G., 2016.

who helped to define yet another style of composition called spectralism, which focused primarily on the aesthetic of continually evolving timbre.[20]

As time progresses, the relationship of sound and composition have become inextricably linked. With the advent of the record album, compositions today may exactly specify the sonic identity of a piece. However, this process of sonic specification is not necessarily achievable by the composer alone, since the sound of a piece is highly dependent on the technical implementation of the systems creating the sound. Typically a composer will work with a group of engineers or designers to consider sonic implementations for their work. This can take place in a studio for recorded work, but it is also discussed for live performance when equipment choice and operation have a large impact on the audience's perception of a composition.

One of the original designers of live sound was Abe Jacob, who mixed the Beatles' last performance at Candlestick Park, and worked on *Sergeant Pepper's Lonely Hearts Club Band*.[21] Having worked with live artists such as Jimi Hendrix and Peter Paul and Mary, Jacob was asked to help a choreographer with the 1975 production of *A Chorus Line* and effectively became the first theatrical sound designer. Theaters and concert halls present unique challenges when electronic sound systems are added, and Jacob became the expert in solving many of these challenges.

There are several concepts which the implementation of the sound systems may affect, but that also impact the perceived result of the composition:

Timbre
The timbre of the composition, which refers to the character or quality of the musical sound as opposed to its pitch or volume, is heavily influenced by the spectral performance of the audio system on which it is reproduced. Timbre can be manipulated through the use of harmonic distortion as well.

[20] Spectral World Music: Proceedings of the Istanbul Spectral Music Conference. Pan Yanincilik, 2008.
[21] Coakley, Jacob. "The Godfather of Sound." Stage Directions, stage-directions.com/5865-the-godfather-of-sound.html.

Imaging
The imaging in a piece is given by the perceived location of each individual part. Some audio systems are capable of generating "phantom sources" which may be perceived to originate from a location where there is no speaker present. Other systems create a sense of imaging by using numerous speakers to simulate many individual sound sources. The spatial separation and diversity of sources defines whether a group of sounds is perceived as a single texture or many individual components.

Balance
The balance is a determination made by the composer and sound designer and technicians. It is often quite complex to communicate what the balance should be, and I have found that sometimes textural descriptions are most useful. When articulated carefully, the balance of many elements can be thought of as a timbre. As a mixing engineer, I memorize timbre rather than balance to try to achieve a specific feel when many elements are involved.

Acoustics
The acoustics of instruments and space greatly determine the ability to achieve one's sonic desires with electronic audio equipment. Where halls have been refined along with acoustic instruments to achieve a very specific sound, speakers often will not complement the same space in the same way. Care must be taken to consider the characteristics of the piece, the acoustics of the instrument and of the room, and the technical specifications of the audio equipment being used. Dispersion and frequency response of audio equipment both play a large role in how equipment supports the natural acoustics of the space.

Timbre, imaging, balance and acoustics factor heavily into the assembling of tools and shaping of systems during live performance. The projects documented in this dissertation all contain audio equipment where selections were made based on how to achieve a specific quality of sound desired by the artists. The equipment falls into roughly two categories: one class of systems that allows us to wrangle and choreograph different forms of content, and a second which allows us to reproduce that content successfully given the environment and performers.

2.3 Hyperinstruments

The motivation behind the Hyperinstrument project was to allow virtuosic performers the ability to produce complex experiences across many mediums in a live performance setting. Previously, these experiences could only be achieved with postproduction in a recording studio, because a large number elements had to be carefully crafted and fit together. The aim of a Hyperinstrument was to give performers control over these elements via an interface which was a natural extension of their expressive tendencies. An example of this is the original Hypercello created for cellist Yo-Yo Ma, which used sensors on and around a specially built cello and bow to allow him to control additional layers of texture and sound simply by playing the way he naturally would.

A Hyperinstrument system has a set of sensors which capture the performance in addition to audio analysis. These sensors take a variety of forms: accelerometers, infrared (IR) beacons, microphones, resistive or capacitive sensing, cameras, radio frequency identification (RFID), etc. The original Hypercello used a combination of capacitive sensing on the fingerboard, IR proximity sensing between the bow and the instrument, accelerometers on the bow to capture movement, and audio capture analysis of the sound of the instrument itself. Data analysis provides a way of deriving "meaning" from all the captured data. We cannot extract this meaning directly because the raw data from the sensors are often incomplete. For example, it may be necessary to understand how a bow is being moved. An accelerometer sensor provides a measurement of the bow's acceleration, but additional interpretation is necessary to understand position or speed. The best analysis systems use data from many sensors, and through interpretation and disambiguation they provide an accurate representation of the performer's expressive and gestural intention. It is this representation that instructs the control for additional layers of the performance. There are many types of analysis techniques ranging from simple scaling and Fast Fourier Transform (FFT) to very complex classification systems that take advantage of machine learning techniques.

2.4 Illustrative Production Examples

Given the ideas in the previous sections, we will briefly discuss the typical organization of existing production styles. For the first type, Broadway Theatre, we have included a generic description of how a production is typically set up. This is possible because there are so many Broadway productions that the process of creating them has become somewhat standardized. Following that, there are examples of some specific non-traditional performances that are put together in interesting ways.

2.4.1 Broadway Theatre

For the typical Broadway show, a person or group of creatives writes the script and composes the score. A director takes the script and brings it to life through a series of dramaturgical rehearsals with the cast. At the same time, the musical director rehearses the cast, band and singers (if the production is a musical). Towards the end of the rehearsal period there are a series of blocking, music and technical rehearsals to give the cast a chance to rehearse the show with full sequencing and technology. During this period there is a rehearsal called a cue-to-cue, where the technical team steps through the entire piece,setting and documenting every piece of production technology along the way with no performers present. The director typically has ultimate creative control over the final rehearsals.

Each department has a separate staff to design the components of the production to the director's specification:

Sound Department

This team supports all of the sound-related requirements for the show including managing and playing sound effects, mixing the live performance, handling sonic reinforcement of instruments and singers, supporting communications and basic video needs. The operation of a Broadway production is achieved with a very specific technique called line-by-line mixing. Since there are many microphones on stage, the goal of the mixing engineer is to keep the smallest number of them "open" at the same time, which results in greater gain before feedback and phase coherence. Sound effects are called by the stage manager and triggered by the primary mixing engineer, called the A1, from the mixing desk. Monitors for the performers to hear themselves on stage are typically controlled from the same mixing desk that is used to balance the sounds in the theatre. Many speakers are positioned around the performance venue, and care is taken to ensure they are all aligned based on the speed of sound so that voices sound consistent and realistic in every seat. In certain productions, an AB audio system is used which actually uses two complete PA systems to allow two microphones to be "open" without causing phase cancellation.

The sound department personnel consist of the following:

The **Sound Designer,** who has creative control of all sonic elements of the production.

The **Associate Sound Designer,** who helps the sound designer to design, draw, specify and install all the equipment on the production

The **A1,** who mixes the show and handles day to day operation of the sound equipment.

The **A2,** which may be one or more people who handle all stage related duties, including miking the cast and orchestra before each performance.

Lighting Department

The lighting department handles all lighting needs for the performance. Sometimes this department handles projection or video work as well. The designer and assistant designer typically work together to develop a set of drawings which represent the creative vision of the designer and director. These documents are used by the master electrician and assistant designer to install lights in the correct location. A programmer sits in rehearsals with the designer and assistant designer to develop a set of cues which control the choreography of the lights throughout the piece. An operator listens to a stage manager throughout the production and triggers the cues at the correct moment in the script.

The lighting department personnel consist of the following:

The **Lighting Designer (LD),** who has complete creative control over all lighting aspects of the production

The **Assistant Lighting Designer (ALD),** who helps the designer to design, draw, specify and install all the equipment on the production.

The **Master Electrician (ME),** who is responsible for installing all the lighting and cabling in the performance space, and for designing and executing power distribution

The **Electricians,** who focus the lights in their correct orientation and location as directed by the the ALD and ME, and who hang lights as directed by the ME.

The **Programmer,** who programs the lighting desk during rehearsals, working with the LD and ALD.

The **Lighting Board Operator,** who operates the lighting board for performances, often much a much less complex task than programming.

Scenic Department

The scenic department handles all carpentry-related tasks. This department may handle automation and rigging as well. Typically the production and set designers come up with a design that will satisfy their creative vision as well as that of the director. A master carpenter builds and installs the set on the stage with the help of the crew. Flymen and carpenters listen to "deck calls" from the stage manager during the performance to move elements of the sit or rigging.

The scenic department personnel consist of the following:

The **Production Designer,** who has complete creative control over the look and aesthetic of the performance.

The **Set Designer,** who designs the set in collaboration with the production designer. This role may be combined with the production designer.

The **Master Carpenter,** who builds the set as specified by the set designer.

The **Carpenters,** who work on the construction or installation of the set as directed by the Master Carpenter

The **Riggers,** who handle rigging of infrastructure, effects and scenic elements

The **Flymen/Automation Operators,** who handle the operation of moving elements during the performances

The Broadway style of production is effective in allowing relatively complex show production that is not based on timecode, but is still reproducible every night with similar working principles and roles developed across a number of shows. This has resulted in a booming business where many performances have been produced and successfully run for decades. However, since the system works so well, a conservatism has developed and the Broadway production style has become quite insular. As a result it is very difficult to achieve certain forms of complexity— an example of this is the recent spiderman production, which would have been quite straightforward for a non Broadway style organization like Cirque du Soleil to produce, but on Broadway suffered from a significant number of production problems, delays, and injuries. There is also a specific sonic aesthetic to these productions — very loud and nasal lead vocals with less attention typically given to instrumental reinforcement — that is not desirable for other types of performance. Nevertheless, many production techniques developed for Broadway musical theater have now been adopted by opera and even television, such as "line-by-line mixing" and hidden microphone placement. These techniques can be utilized without the traditional Broadway sound, and they have revolutized many other areas where complex mixing is required.

2.4.2 Orchestra

A symphony orchestra is the largest form of professional classical performance organization, other than some opera companies. The conductor is the artistic leader of an orchestra and is responsible for the coordination of all musical activities. The ensemble of players is divided into major sections: strings, winds, brass, and percussion. Within those sections, players are grouped by instrument. And the instruments are assigned a part or multiple parts in the score. Where there are multiple parts, the first part is typically the most complex and the highest in pitch, so that "first violins," "first violas," etc. are typically the top notes of the chords which are the most difficult to play. Within the first violins, there are desks, so that the first violin, first desk sits closest to the conductor. Each desk has two chairs. The first violinist at the first desk in the first chair is the concertmaster, who is responsible for tuning the orchestra.

Since members of the orchestra sit according to their part, desk, and chair, it is much easier to hear one's own part rather than other parts and instruments. For this reason the conductor gives feedback to try to maintain the balance called for by the composition. This feedback augments the written instructions contained in the score and parts that each player reads.

When composing a new piece, a composer will traditionally write the basic melody and harmony first, along with a bass part. This sketch is then arranged and orchestrated to form the full score, with parts determined for every member of the ensemble for the duration of the piece. The conductor is given a score that shows all the parts simultaneously, but each individual player is provided only with the part that he or she plays. This means that players do not have a sense of how their part aligns with other parts in their section. As a player it is important to develop a relative sense of how one sounds in context, imagining what the conductor is hearing and asking for to ensure that balance is correct not in one's own seat, but for the audience.

Orchestra rehearsal is very expensive, and there are strict rules governing how long performers may work. Therefore, by the time a new piece is rehearsed by the orchestra, there is little opportunity to iterate on the composition or fix any errors. Often what the orchestra plays the first time a new piece is read is therefore representative of how that piece will sound in performance. Composers may adjust their piece for a future ensemble, but rehearsal time is not used to incorporate any significant changes. Rehearsal time is often limited to one or two complete runs of the piece, with a short period of time for questions and minor corrections. It is therefore extremely important for a piece to be playable and to work the first time it is read by the orchestra. Composers are trained to imagine the orchestra playing in their head while creating a piece. Their imagination is refined, with years of training, to be able to "hear" the qualities and performance techniques of real orchestras. Similarly, conductors rehearse and study pieces without any sound, entirely in their minds.

This style of working has many implications: first, it does not support any significant degree of variation; there is no time to rehearse anything that cannot be easily sight read. It also means there is little to no experimentation. A piece must be finished to be performed, and the composer must be able to audiate the piece in his or her mind without needing to test timbral or arrangement ideas. This results in a rather conservative repertoire and has made it quite difficult for orchestras to remain relevant and economically successful in the contemporary arts marketplace.

2.4.3 Immersive Theater

Sleep No More is an immersive theater experience based on Shakespeare's Macbeth and Alfred Hitchcock's 1940 film *Rebecca*. The piece was written by the theater collective called Punchdrunk and was first produced in Brookline, Massachusetts in 2009 by the American Repertory Theater. It has taken up permanent residence in New York City since 2011. Audience members experience the performance by walking through a 90,000 square foot building with 6 stories of extremely detailed set, while the cast members have interactions and play out scenes throughout the building, sometimes involving the audience. The audience members wear masks and do not speak for the duration of the 3 hour event. There are 20 hours of content in the performance. Users cannot see everything in one night and are encouraged to make their own version of the performance during each viewing by exploring and following the characters they find interesting.

Sleep No More is a unique example of live immersive theater. The audience is mobile and the environment is carefully controlled. The quality of the performance makes one feel as though everything happening is improvised, when in truth the action is choreographed down to the second. Several factors contribute to this: At the start of the performance, a 3 hour long soundtrack is triggered. This plays throughout all the spaces in the production with 19 channels of different audio feeding many different areas of the performance space. Adjacent areas have similar but not entirely identical sounds playing. Some parts of the sounds may be the same, but others may change. This results in a smoothly transitioning sonic environment across nearly 90,000 sq feet of performance spaces. The timecode from the recording feeds a lighting console which controls all of the lighting in the entire facility as well.

Performers listen to the soundtrack and use it to time their movements very precisely. Although there is no steady rhythm, the soundtrack has fragments of a pulse that start with clearly identifiable material (such as a voice, a beat or a swell). The performers memorize these moments and time their movements and traversal of the space accordingly. The movements are essentially a fully choreographed dance, as there is very little speaking by the cast and no speaking by the audience.

There are moments of the production called "one-on-ones" where a performer chooses an audience member at random to bring into a private room to perform an interactive monologue, which might involve physically touching the performer or other intimate actions. Because the world of the performance is set up like a real environment, there are aspects of it that work just like the real world. For example all the phones on the set can dial one another, and the characters often make calls from one part of the set to another. The designers use scent and temperature to help the audience navigate the spaces, with certain rooms smelling of roses or being very cold. Because the space is so large, this contributes significantly to one's sense of spatial awareness and memory by involving additional senses. Smell especially is used to form specific scent memories that aid in this navigation.

Sleep No More is so unique because it feels improvised even though it is not. This is achieved by a combination of the agency of the audience and the vast amount of content, which prevents one from seeing repeated material or understanding how the cast's movements are repetitive. Additionally, there are moments where micro-improvisations are possible within the constrained structure of the piece. While the choreography is prescriptive, there is still room for the cast to alter very small elements of their performance to customize and enhance a connection to the audience. This is an important idea: that a structure which is entirely constrained can still leave

room for this kind of connection. It is a very different concept from a pop tour, even though both use a fixed soundtrack. The pop tour leaves no room for this style of connection through micro-improvisation, whereas the Sleep No More production achieves it powerfully. A challenge of this approach is the rigidity of the macro structure of the piece. Because all the pieces are so carefully intertwined and specified, it becomes difficult to change or alter any one part of the performance without affecting anything else. This is less critical for a theatrical experience where the script and story do not change, but in a musical context it can be powerful to vary the larger structure of a performance.

In the next chapter, there will be a brief discussion of a research project undertaken to extend this production and the feeling of improvisatory immersive storytelling onto the internet. I will also detail the implementation of Jacob Collier's one-man show, which has many similarities in structure and design applied in a completely different medium.

2.4.4 Pop Tour

A recent pop tour that I was involved with provided a chance to see how some of the larger productions are put together. This particular tour was a mid-sized tour in arenas, which generally seat 15,000-40,000 people. Stadium tours are even larger, with 100,000 or more seats. This experience is representative of an up and coming young artist who may be a household name, but does not have the resources of a "pop icon" to spend at the highest level on production.

For this style of tour, a music director will rehearse a band of professional session players along with backing tracks operated by a playback operator and a monitor engineer. The music director usually plays an instrument live with the band as well. These backing tracks are often modified versions of the album Pro Tools sessions. They contain vocals, effects and any other elements that are not played by the band. This enables the band to completely rehearse without the artist present. The artist will arrive towards the end of this rehearsal period, run the songs with the band in a rehearsal studio and make comments on the arrangements. A creative director and choreographer will go through the show and add dance and transition elements, and the arrangements and show timing are modified to support these. Special verbal cues and count-ins are added to the backing tracks, but these are only heard by the performers through in-ear monitors.

Once the timing is set, timecode is added to the backing tracks, and this timing is provided to lighting, video and audio departments. Multi-track audio is recorded of the rehearsals to give both the monitor engineer and the front-of-house engineers time to refine their mixes. As a rule of thumb it is desirable for the concert and monitor mixes to sound as much like the album as possible, so engineers will often try to recreate elements of the studio production on the live mixing desk or with outboard processing.

Once the tracks and timing are determined, all of the design and production teams develop lighting, sound and video content and programming offline in previsualization (pre-viz) suites. These suites have a 3D model of the set on a large screen with all lighting and visual aspects of the performance simulated. Pre-viz allows all designs to be finalized before anything physical is constructed.

Following this production period, the full stage and set is constructed for a week of rehearsals in an arena. This allows the technical teams to refine and program, as well as the dancers and band to rehearse together. Since the backing track is from the album, the album vocals may be used if the artist is not available for all of the rehearsals. The production manager oversees the logistics and personnel to ensure that the stage and set can be constructed and dismantled in the required amount of time. For many arena tours, the entire stage must be constructed and dismantled in 4 hours. Special strategies are used to achieve this. For example, the crew can build the stage at the opposite end of the arena with all special elements such as elevators and pyrotechnics while simultaneously hanging the sound and lighting systems. Then the crew will roll the entire stage with elevators and effects into place under the sound and lighting systems.

For the performance, dressing rooms and production spaces built underneath the stage are identical every night. The playback operator and monitor engineers are located under the stage and communicate with the performers via microphones and in-ear-monitors. Displays show the technicians underneath the stage what is happening. A control position is set up in the center of the arena with light, sound, video and effects consoles.

During the performance, the playback operator triggers the start of each song, then the ProTools systems send timecode to all other systems, which are programmed to trigger cues at specific times. The playback system may also have live effects which are triggered based on timecode, such as vocal correction. On much smaller tours, all playback might be loaded into a digital drum machine and triggered by the drummer. For larger tours,

many more mixing desks may be used with separate operators for the band and artist and with many playback systems and operators.[22]

Although functionally similar to the infrastructure used for *Sleep No More*, this execution represents the most constricted form of production— the vast number of elements that are choreographed and added to the backing track make it very difficult for performers to have any lee-way in what they do on stage. The band can add micro-improvisations, but they are often constrained by the structure of the music arrangement, processing and effects added to the live instruments. However, this rigidity makes incredible effects such as pyrotechnics, flying, optical illusions, video mapping, and other spectacle possible.

2.4.5 Indie Artist Performance

Björk's *Biophilia* tour functions a bit more like an experimental version of a pop tour. Rather than using audio and timecode to provide synchronization, a pair of redundant Logic Pro laptops send MIDI to many of the instruments on stage to control and synchronize them. Some of these instruments include a pipe organ, a Tesla coil, and Andy Cavatorta's[23] gravity harps.[24] Performers on stage play these non-traditional instruments (the Tesla coil being controlled by a Reactable[25]) or a set of iPads generating effects free from time code.[26] In addition, many traditional acoustic instruments are used. The tour also featured an Icelandic choir. The large number of performers, interactivity and instruments gives a more natural feel despite still being timeline based. This type of performance requires a much more involved, collaborative, all-hands-on-deck style of rehearsing. And there is significantly more flexibility in performance. This affords a much more natural feel simply because of the large number of people working together. In many ways this approach has some similarities to orchestral performance. A large ensemble by definition infuses some amount of human timing. That said, this approach

[22] Staff, ProSoundNetwork Editorial. "All Hands on Desk for Beyoncé and JAY-Z's On The Run II." ProSoundNetwork.com, 21 Sept. 2018, www.prosoundnetwork.com/live/all-hands-on-desk-for-beyoncé-and-jay-zs-on-the-run-ii.

[23] Cavatorta, A. "Nervebox: A Control System for Machines that Make Music." M.S. Thesis. MIT Media Laboratory, 2010.

[24] Marantz, Andrew. "Inventing Björk's Gravity Harp." The New Yorker, The New Yorker, 19 June 2017, www.newyorker.com/culture/culture-desk/inventing-bjrks-gravity-harp.

[25] "Reactable -." Music Knowledge Technology, reactable.com/.

[26] Kaganskiy, Julia. "Meet Max Weisel: The 20-Year-Old Behind Björk's Interactive Live Set-Up." Vice, 31 Jan. 2012, www.vice.com/en_us/article/pgzwjn/meet-max-weisel-the-20-year-old-behind-björks-interactive-live-set-up.

still prevents, for example, a situation where the artist may wish to repeat a section or alter the macro structure in any way.

2.4.6 Jam Band

Snarky Puppy is a popular jazz band that tours with a relatively large number of musicians. At the Ground Up Festival in 2017[27], they had 47 musicians on stage playing simultaneously. These types of bands have no pre-recorded materials at all, with all inputs to the mixing desks coming from instruments played by performers. This requires other departments (lighting and video) to carefully follow the music and the action on stage and improvise. The mixing engineer can't easily prepare, because almost every performance has slightly different instruments and the band will change what they are doing as well. This results in some remarkable musical spontaneity and highly emotional audience interactions. The band can stretch solos out as long as they feel is musically appropriate in the moment. The macro and micro structure of arrangements can be modified on the fly, and there is no need to verify that technology will adapt or follow along. If the audience and band are feeling collective excitement, extra choruses, different chord progressions, and other alterations may be included spontaneously. Musically this requires the band to be curiatorially conscious of what is happening. It could be possible for a solo to go on for too long. With no confining structure for the performance or individual songs, the artist must rely on his or her own sensibilities to guide and shape the performance in real time. This can be exceedingly difficult, even for something as simple as consistent tempi. This type of show also dictates that lighting and video (if video is used at all) are very basic, since it is difficult to make these systems adaptable in the face of so much flexibility. In these cases, the operators often have a deep understanding of the music and the artists and follow along as if they are additional members of the band. Still, there is a limit to the ability of an operator (most do not have the musical abilities of the performers) to trigger elements perfectly and at a large scale.

[27] Almeida, Celia. "GroundUp Festival 2017: Five Unmissable, Under-t he-Radar Performers." Miami New Times, 4, 8 Feb. 2017, www.miaminewtimes.com/music/groundup-festival-2017-five-unmissable-under-the-radar-performers-9121528.

2.4.7 Live Television Broadcast

For a television broadcast such as *The Tonight Show with Jimmy Fallon* that has a house band and guest bands that perform live-on-air, the workflow is quite different, because the show is technically identical each night. For situations like this, it is essential to make the workflow as efficient as possible. For example, a task that requires 10 button presses and must be done 10 times a day, every day, may be reduced to a single macro button to save the operator a lot of effort. Mix suites for television have couches and comfy chairs, because rather than being a portable setup, this is someone's office each day. Very specific steps are followed for differing parts of the show, such as guest bands, to enable the operator to have as much of a head start as possible to get a good sound. The bass always plugs into the same channel regardless of the guest band. Rehearsals are important because each second on live air counts and the television crew must understand where to point the cameras.

For audio reinforcement in the studio for the live audience, a large number of small speakers are used rather than a traditional PA. By using a high density of speakers closer to the audience, each speaker can be quieter with lower overall room sound. This helps the sound of the microphones feel close, like a recording studio rather than a live performance. When Fallon runs up into the audience, there is a special user interface that allows an operator to move a small puck over a drawing of the seating. The operator moves the puck to follow Fallon as he moves around the audience and its location causes the speakers in the area to dim significantly.

Television is an interesting medium because timing must be very carefully constrained due to the broadcast schedule. This means that bands must take all necessary precautions to ensure their songs fit in the exact amount of time needed. This often means playing to backing-track, and some artists even lip-sync. After an infamous episode in 2004 where Ashlee Simpson's vocal track was triggered incorrectly, many artists now sing live on Saturday Night Live.[28] However, there are often additional sounds and tracks that are triggered via backing. Mixing for television is a challenge because the bands must attempt to do a studio mix in real time. In a live venue, there is some lee-way for the performance to sound slightly different than the recordings because the environment is so vastly different from one where fans would listen to an album. For broadcast, the audience is listening on exactly the same equipment, and it is possible to directly compare the sound of the live performance to the sound of the album recording. There is another challenge in that often the mixing desk for the TV show that is broadcast is not the same as the one that is used for live performances. A band may have

[28] Leung, Rebecca. "Michaels: Lip-Sync An 'SNL' No-No." CBS News, CBS Interactive, 1 Nov. 2004, www.cbsnews.com/news/michaels-lip-sync-an-snl-no-no/.

worked carefully to program processing and effects, but these are difficult to integrate on the television studio mixing console. In many cases a single operator will run the desk for all guest bands and not their usual mixers. This makes sense in some ways, because different skills are needed to craft a mix that sounds good on consumer television speakers. However, this mixing engineer often does not know the music of the guest band well.

2.4.8 DJ Set

At a large DJ performance (such as at the Coachella music festival) there are several methods of production, so that it can be difficult to know which technologies are contributing to the listening and viewing experience. In certain performances timecode is used for very advanced and complex lighting. In this case, the DJ is not able to have much room to improvise. On other types of performances, the DJ uses a computer or a CDJ-based[29] system to read the crowd and choose songs and mashups at will. For these types of performances, there are amazing lighting operators who have perfected a technique called busking, that enables very complex lighting effects to be used in an improvised way. This will often be combined with other effects such as LED video walls, carbon dioxide cannons and lasers that result in a completely immersive improvised experience. This works well because the music is generally quite similar in timbre, structure, rhythm and tempo, so the lighting operator is able to have a good idea of what will happen next and run the production elements accordingly.

This is a powerful example of advanced production being improvised and applied in real time, but this works only because the music is somewhat repetitive and therefore relatively easy to understand, and this makes it easier for operators to predict what will happen next. Still these designers[30] are at the forefront of using spectacle and production in creative ways. This is the type of production that we would like to achieve for musically complex work— it represents the highest level of scale and complexity seen today that is not exclusively timecode driven.

[29] DJResource.eu. "Ultimate CDJ Comparison Chart." DJResource, www.djresource.eu/Topics/story/25/The-Ultimate-CDJ-Comparison-Chart/.

[30] "Los Angeles." Production Club, production.club/.

2.4.9 Summary of Experiences

These experiences are summarized in the table below. The experiences with the highest production value forbid improvisation of any kind. By sacrificing certain production components, adding musicians who have mainly technical roles (who may even function solely as an operator), or structuring the timed content to allow micro-improvisations, it is possible to achieve some form of improvisation and production. All completely free timed productions have relatively low production value.

Table 2-1: Summary of existing production experiences and their limitations.

Type of Production	Time-code/ Timeline Based	Micro-Improvisations Possible	Cue/Trigger Based Production	Free Timing	Operators Require Musical Ability	Production Value
Broadway Theater	No	Yes	Yes	Yes	Yes	Medium - live mixes are not often as refined as pop tour.
Orchestra	No	Yes	No	Yes	Yes	Low
Immersive Theater	Yes	Yes - show format lends itself to micro-improvisations in dance, music would be much more difficult.	No	No	No	High
Pop Tour	Yes	No	No	No	No	Very High
Indie Artist	Yes	Yes	No	No	Yes	High
Jam Band	No	Yes	Yes	Yes	Yes	Low
Live TV Broadcast	Yes	No	No	No	No	High, but with difficulties translating stage mixes to television
DJ Set	Yes	No	Yes	No	Yes	Very High

Now that we have reviewed examples of previously existing performance systems, structures and technologies, we will focus on new works that attempt to break the rules outlined above. As mentioned earlier, the work in this dissertation investigates methods of integrating technology and live music performance in such a way that the technology allows for flexible musical expression, sound and meaningful connection to the audience, while allowing for exciting, sophisticated and magical production values. The coming chapters will address important concepts to consider when developing new approaches, technologies and structures for large-scale entertainment productions by documenting project development in great detail. The work is categorized roughly into two tracks: Opera of the Future and Jacob Collier. Much of the work for Opera of the Future has required bringing truly new systems to light, while the work with Jacob Collier utilizes existing technologies and ideas, many of which I learned through the process of creating productions in Opera of the Future. Following that documentation is a short description of a production system which was installed a private home, and which I believe may be a blueprint for future personal production. Finally, I will reflect upon and tie together concepts relating to this work.

3 The Opera of the Future

Productions in Professor Tod Machover's MIT Media Lab research group *Opera of the Future* often involve many technological innovations typically considered by professionals to be too risky for live performance. As researchers, we introduce and carry these risks as an attempt to challenge existing notions of performance and production, and to defy and expand conventional wisdom. This chapter presents a deep look at these intentional risks through the documentation of several large projects spanning from 2007 to the time of this writing.

Through these projects, we have found a dependable method of successfully producing risky work: approaching the act of designing and building a production from the ground up, using the most basic engineering principles, rather than typical entertainment industry patterns. Many professionals in the entertainment industry have very specific expectations, processes, and timelines, as this produces quality work efficiently. In *Opera of the Future* we begin very few assumptions, if any, and instead set our timelines and development processes using the most basic ideas about invention. Fundamentally, I believe that creating live production with newly invented technology requires this mindset. The ideation and initial development on a such a project requires potentially unlimited time, which can be daunting, as it is impossible to know how long it will take for the project to advance, let alone be completed. The best creators, such as Tod, understand that this is the true process of developing new software and hardware. It is nearly impossible to imagine something significant, design it, build it, and then have it work as intended. The reality of this process is much messier, with problems popping up at every turn, requiring significant iteration. It is a challenge to be conservative about planning and executing without stifling the creative possibilities of the project. This chapter attempts also to show, by example, how to balance the pursuit of creativity with the added risk and complexity of inventing new technology.

3.1 Early Projects

Before arriving at MIT, my production experience was in live sound for staged musicals and rock and roll shows. Tod started his Media Lab research group in 1985, and I started working in the *Opera of the Future* research group my freshman year, in November of 2007. Once at MIT, I quickly learned that contemporary classical performance was unlike musical or rock performances, but had some interesting qualities that were loosely related:

It is not just the refined acoustic sound of instruments which is important but also the articulation and performance techniques used.

1. Through-composed pieces would generally consist of electronic and acoustic sounds mixed together, with the electronic sounds coming from one or many speakers, and the acoustic sounds coming — amplified or not — from live players performing with classical instruments.
2. Synchronizing the electronic and acoustic sounds was challenging. In pieces from the 1950s–1970s, the musical score would be notated with inches and feet of tape, corresponding to the exact location on a reel of tape where the playhead should be positioned for that musical moment. In more recent pieces, fragments of sound might be loaded into a sampler and triggered via MIDI notes. Some new pieces used sensors to automatically trigger sounds at the right part of the score, with a computer program operating as a "score follower" or analyzing the live performance of the musicians.
3. Some pieces also used microphones to capture and transform the acoustic sounds, turning them into electronic sounds.
4. Some pieces would use microphones and sensors to add other elements to the performance, such as lighting or images projected behind the performers, or the manipulation of real-time audio effects. Sometimes these additional elements were made ahead of time and triggered or timed to go with the performance, just like tape or samplers were used for audio.

When I joined, there was also a good deal of institutional knowledge that complimented my own observations:

- Generalized score following[1] was understood to be temperamental and not typically trusted. It resulted in awkward moments where the computer would misinterpret the score or performance, resulting in unmusical, unartistic interjections during the piece, or producing uncomfortable desynchronization. Humans were just better at keeping the technology in lock step— this could involve playing notes on a MIDI keyboard at designated parts of a piece to trigger electronics, running "tape" at variable speed to keep it aligned properly with the live performance, or putting real time systems in different modes at specific points in the piece to change their parameters. In all cases, performers or technicians were trusted with this rather than computers.

[1] Barry Vercoe, The Synthetic Performer in the Context of Live Performance, Proceedings of the International Computer Music Conference, Paris, 1984.

- There was a strong focus on reliability. Often someone would build something and have it somehow fail right before the performance. The visual programming language Max MSP 4.6 was a common culprit for many of these performance problems. Max uses graphical boxes with connecting lines to allow users to build up functionality by creating a mesh of processes. Several times I saw someone spend weeks building a very complicated Max patch only to have it crash or do the wrong thing at the moment it was supposed to work in front of an audience. This was due to the general instability of the software at the time, combined with a language that made it easy to produce very confusing and convoluted designs. Depending on how the patch was structured, debugging these systems could be a nightmare. We used the term "spaghetti" to describe patches that were unreadable because of all the connecting lines. In recent years Max MSP has added functionality to enable clearer organization of patches and it has become much less crash-prone.
- Since many of Tod's pieces used traditional Western instruments (i.e. those found in a symphony orchestra), another important philosophy was to build from the acoustic sound of those instruments, which has been refined for hundreds of years in concert halls. For example, it is extraordinarily difficult to amplify a violin with a microphone and speaker in such a way that feels natural and affords the player all the familiar expressive tools that work acoustically. In most halls amplification is not needed for a violin, and the magic of the sound is lost when put through speakers. For many of Tod's pieces, the natural sound and texture of acoustic instruments was preserved: not modified significantly, but rather augmented by additional electronic content played through speakers.
- Many systems that work in a lab setting do not function well in performance situations. Microphone bleed, stage lights, temperature, humidity, and tempo may all change unpredictably in performance. For systems to work reliably, they must be designed to be robust in a variety of environments and situations.
- There was much consideration about the best way to layer electronic and acoustic sounds. A lot of projects had portions featuring subtle layering of many different audio sources. The exact relationship between the layers and sonic qualities of sources

A piece can be designed with strict rules but still seem very organic, dynamic and improvisational from an external perspective.

was a part of the composition just like notes. The mix, balance and acoustic relationship of these sources was written into the piece. Because of this, the choice of equipment and the exact process of production and live reinforcement was extremely important to the proper realization of the piece. Immediately it became evident that there were specific challenges with this form of composition when producing performances for classical venues. Often speakers were intended for listening in a small space, or reproducing a small spectrum (such as speech) in a large space. It was extremely rare to find speakers that worked in a large space and gave the right feeling and impact to support the compositions that Tod had written.

Even when designing for large halls, it is possible to simulate the feel and experience of the full-scale technical system in a smaller space.

With these ideas in mind, I began exploring, building and putting on live performances. Each project and performance was an opportunity to try a different approach. Afterwards we would reflect on and evaluate the approach for future projects. My first fall, I created a basic pitch recognition system based on Miller Puckette's *~fiddle* for a visualizer running with the Ying Quartet[2]. Next, I supported a performance of Jeux Deux for the Detroit Symphony Orchestra, involving a Disklavier MIDI piano and Max patch made by Michael Fabio which analyzed notes played by the pianist (via MIDI) and sent data back to the piano immediately to accompany the pianist in real time. For Jeux Deux the system had several modes of triggers and interaction, and these were switched by the pianist from a small midi controller on top of the piano. The system looked for specific pitches as triggers to start each fragment of accompaniment. Since the piano could send and receive MIDI, this was very robust and reproducible in all types of environments. The interactions gave the impression that the piano had its own personality and was reacting dynamically to the performer, despite a very strict set of rules defining the piece. This was a powerful idea: that a piece could be designed with strict rules but still seem very organic, dynamic and improvisational from an external perspective.

Around this time I also became interested in methods of surround sound that did not utilize a matrix of outputs, as was common in systems by Dolby and DTS at the time. Instead, I began to explore models for positioning sound objects in space abstractly, placing them with coordinates rather than defining send levels via traditional pan pots. I found papers and implementations for one such method called 3rd Order Ambisonics[3]

[2] "Ying Quartet." Ying Quartet, www.ying4.com/.
[3] "HOA Technical Notes - Introduction to Higher Order Ambisonics." Blue Ripple Sound, www.blueripplesound.com/hoa-introduction.

and began experimenting with various tools to work in this format. I began with the ICST ambisonics tools[4] in Max 4.6, which provided a rudimentary GUI for placing sounds and then subsequently programing speaker locations. By inputting both the location of one's speakers and where sounds should be located, these tools would allow sounds to be placed in 3D without any specific dependency on speaker location. The locations of the speakers could be changed easily, so that no matter where the speakers were located, the system could make it seem to the listener as though the sounds were in the same locations with respect to the center of the room. This made it possible to design a soundscape in a studio with one set of speakers, then take the soundscape to a large room or concert hall with a different number of speakers in different locations than in the studio. Using Ambisonic encoding, the soundscape would sound roughly the same in the concert hall and the studio, despite a drastically different sized room and arrangement of speakers. This style of workflow illuminated another powerful idea: even when designing for large halls, it was possible to simulate the feel and experience of a system in a smaller space. Trusting our sense of how an experience could feel, and then seeing that scale to larger space and audience, enabled the designer tap into and follow their emotions when designing systems and technology. The goal was to make an experience feel magical, organic, and special, then determine how to bring that feeling to a larger group of people. I quickly learned that this wasn't only a matter of taking the same equipment from a small room to a large room, or even adding more of the "small-room" equipment in the larger room. Large spaces did behave differently, and required special design and technology to help experiences translate to every audience member.

3.2 *Skellig*

In the summer of 2008, Tod was in the midst of producing his opera *Skellig*, which contained several non-traditional elements. In addition to a professional cast, there was a chorus of middle school aged children. They sang traditional musical parts, but Tod also had the idea that the chorus should make affected vocal textures to augment the electronics and music. Charles Holbrow and I spent time making sound examples of many types of noises, inspired by Trevor Wishart's *Vox Cycle* (especially Vox 5)[5]. Simone Ovsey and I traveled to the English countryside near Durham to work with the chorus using a large headphone system, allowing everyone in the chorus to listen to sounds without hearing them out loud in the rehearsal room. This was connected to

[4] "Tool: ICST Ambisonics Tools." Cycling '74, cycling74.com/tools/icst-ambisonics-tools.
[5] Milani, Matteo. "Trevor Wishart: Chemistry Of Sound • Digicult: Digital Art, Design and Culture." Digicult, 30 Apr. 2016, digicult.it/digimag/issue-041/trevor-wishart-chemistry-of-sound/.

a sampler, and we played the sound examples to the choir and had them experiment with trying to imitate the sounds they heard in their headphones. We then recorded these sounds, which were incorporated into textures triggered by the orchestra, and we also came up with several textures which the chorus would perform live.

3.2.1 Electronics

There were three types of electronic sounds in the piece:

We used the term "trigger" to describe stereo sounds which were long in length and had complex pitch content. These were designed to be slightly overlapped and were loaded into a MIDI sampler (Kontakt 3[6] running on a MacBook Core2 Duo). The sampler was set to play the sound for as long as the key was held down, with a 1.4 second release envelope. This allowed the keyboard player to crossfade multiple triggers by holding and releasing multiple keys at the same time. Sounds were edited so they aligned with a number of measures of the piece, similar to using tape. The sampler notes for each trigger were notated in a part along with the rest of the orchestral score, so the part could be performed as if it were a normal instrument in the orchestra. In each measure, the orchestra would play music, and the trigger notated for that measure would play electronic sounds which connected with the acoustic music. The duration of the note (i.e. the number of beats) in the score represented roughly how long each sound should last. The extra time at the end was provided in case the orchestra and singers were performing slightly under tempo. Tod conceived of the sounds so they shadowed the musical content of the orchestra but were not strictly dependent on the exact timing and tempo. The shadow had the effect of thickening the sound, making the timbre complex, nuanced, and most importantly continually changing and shifting. The subtle but continuous variation was intriguing and engaged curiosity in listeners. This concept of the "shadow line" is quite Machoverian and Tod executes it masterfully!

Another class of electronics was performed on electronic synthesizers played as if they were a piano. The part was notated in the score just like a piano part, with two staves. We called this the "virtuosic" keyboard part, and it required a pianist to play in the manner of a percussion part, where the triggers did not require technical skill, but rather very precise counting and rhythm. The sounds for this keyboard were generated from hardware

[6] "KONTAKT 6." Samplers : Kontakt 6 | Komplete, www.native-instruments.com/en/products/komplete/samplers/kontakt-6/.

synthesizer modules. We used three modules for this piece, the Yamaha FS1R[7], the Kurzweil K2500R[8], and the Nord Lead 2[9]. We needed a way to automate the programming of the synthesizers throughout the duration of the score. With Michael Miller, I created a Java application which would receive MIDI program changes from a controller in the orchestra and in turn send commands to each of the synthesizers to select specific sounds and mixes of sounds to be sent to the mixing desk. We called each set of sounds a "patch" after the name given to a specific wiring of an analog modular synthesizer. There were around 20 patches in the piece. I chose a Yamaha S90[10] midi controller, as it has buttons and LEDs which send program changes very easily, with a dedicated button and LED for up to 16 program changes. This allowed the keyboardist to quickly know which patch was selected and move to any patch with one button. To do more than 16 program changes, there was a bank selection switch which also had a dedicated button for up to 8 banks. Together this allowed direct access to any one of 128 programs with two button presses, and 16 adjacent patches with a single button press. It is difficult to find keyboard controllers with this interface these days; many have an increment and decrement button, or require multiple presses or a wheel to access programs. These interactions are not reliable during a performance. It is too easy to select the wrong patch by accident. The system also had to handle some special edge cases during patch changes, such as pedal down. In some cases, this can cause stuck notes, so we had to be sure to allow for MIDI note off messages to be sent to modules even in other presets. Latency is another concern for virtuosic keyboard systems. This was helped by the fact that hardware synthesizers generally had little to no latency.

The final type of electronic sound in the piece was what we called a texture. It was a piece of composed electronics that was several minutes long. Tod took fragments of the recorded voices of the cast and edited them together in his studio. We felt that this would be an opportunity to put together a part of the project in surround sound using the ambisonic workflow I had begun experimenting with earlier in the semester. To allow us to mix Tod's composition in surround sound, I created a patch in Max which would take audio from Logic Pro via Jack for OS X, an OS level audio routing program. The Max patch listened to MTC timecode from Logic and would recall preset movement trajectories at times specified in a cue-list of movements and positions.

[7] "Yamaha FS1R." Home Page, Vintagesynth, 16 Oct. 2019, www.vintagesynth.com/yamaha/fs1r.php.
[8] "Kurzweil K2500." Home Page, Vintagesynth, 19 Oct. 2019, www.vintagesynth.com/kurzweil/k2500.php.
[9] "Nord Lead 2." Nord Keyboards, www.nordkeyboards.com/products/nord-lead-2.
[10] "S90 ES - Overview - Yamaha - United States." YAMAHA, usa.yamaha.com/products/music_production/synthesizers/s90_es/index.html.

Charles Holbrow[11] worked with Tod to break his composition up into many pieces and then spent a good amount of time using Logic and Max to spatialize the piece dynamically. This was then saved as an encoded stream of 16 channels of audio which could be decoded either in our studio or anywhere else. To this day, I use that recording to test newly finished Ambisonic systems. This texture was triggered manually from a laptop next to the mixing desk, since it was just a one-off and it was many minutes long.

3.2.2 Live Production

I went to England in November of 2008 for the premier of the performance, with a Pelican case full of synthesizer modules, and some laptops and audio interfaces. The opera was performed at the Sage Gateshead theater in Newcastle, and we hired a sound designer, Chris Full, who knew the space and the sound systems there. He had organized a set of 8 distributed surround speaker channels around the theater, which seated about 1500 people. In total, we had 32 microphones in the orchestra pit and the cast of 18 all were fitted with wireless microphones, There were line inputs from the triggers, virtuosic synths and the Ambisonic texture inputs. These all had to be combined together on a large mixing console and sent to roughly 50 speakers in the hall, including the 8 surrounding groups. The Sage Gateshead had hired a technician to run the mixing console during the performance. However, since the music was quite difficult and the content of the electronics was not noted in the score or libretto, Chris and Tod suggested that I mix the show.

The mixing desk was a pre-release version of the Midas Pro 6[12]. I had previous experience with analog consoles and small digital consoles, but never a large desk like the Midas. This console was designed around pop-groups, which were configurable labeled groups of inputs that could be activated on the central faders of the console. Chris set this up so that for each scene, important channels were right under my fingers. I would later learn this was a slight modification of a common technique for mixing Broadway theater. With that configuration, we were able to do the mix with Tod looking after the trigger levels, and me mixing the live microphones. This was the first opera of Tod's that had ever been produced without a supertitle screen, so Tod and the creative team were especially concerned to ensure the text could be clearly understood. This was the only way for the audience to grasp the plot, as this was combined with the relative inexperience of the youth choir who was on stage

[11] Holbrow, C. "Hypercompression – Stochastic Musical Processing" M.S. Thesis. MIT Media Laboratory, 2015.
[12] "PRO6-CC-IP: Midas: P0ARU." Midas, www.midasconsoles.com/Categories/Midas/Mixers/Digital/PRO6-CC-IP/p/P0ARU.

making textures. The voices of the principal performers had to be balanced along with the choir and electronics.

I learned a lot from watching Tod mix the electronics. So much of the mixing I had previously done revolved around finding a good static configuration of the input levels, and making periodic adjustments. The goal was to find a specific and subjective balance of acoustic and electronic sources, and to maintain that relationship through the continuum of expression of the performers, or even to grow and change the relationship where musically appropriate. There is not a "right" balance, and it was not the goal to hear everything equally (which is the typical charge given to sound engineers). Instead, this kind of mixing is much more like mixing an album in a recording studio, where deliberate creative choices are made about the makeup of the final sound that is heard. I quickly understood that perception of volume and balance was highly dependent on timbre. Sometimes a simple change in timbre would require me to pull the fader back significantly despite no change in meter deflection. In broad terms, the choir was the most useful for adding high end detail and my goal was to fit that into the electronics and orchestra texture. Many of the sounds the choir made besides singing were based on critters moving around in dead leaves in a garage. So this high end complemented both the warmth of the orchestra and the attack and low frequency content in the triggers and synths. I began to understand that to fit all the pieces together, I had to find spectral space for each component to occupy, and the most satisfying mix utilized all parts of spectrum, extending all the way up to the highest high texture and down to the lowest low sounds.

Tod always said that one of his goals was to use technology to enable the creation and performance of studio-like experiences in a live setting. I have spent much of my time at the Lab helping to do this. Even with all the complexity and automation and technology that is used to allow many small elements to be interlinked and interactive, I believe this method of mixing is truly what brings the element of studio sound to our projects and allows us to amplify the expressiveness and emotion of the performance.

3.3 *Death and the Powers*

There is documentation on *Death and the Powers* in my Master's thesis,[13] which describes the project in the context of the application of mapping and *Hyperproduction*. In this dissertation, I detail more exactly all the systems and especially the processed used to select and design the technology components, and the process of creatively developing and then rehearsing the piece.

By the time I joined the Opera of the Future Group, Tod's largest project, *Death and the Powers*, had already been in development for almost 7 years. A patron of the Opéra de Monte-Carlo had asked Tod to develop an ambitious project as a birthday present for Prince Albert of Monaco. The opera house in Monte Carlo overlooked the Medditarean Sea. Initially, this patron had asked for a production which began on stage in the opera house, but finished in the sea, with the audience leaving the opera house for the final scenes.

The project went through several iterations, and when I arrived in 2007, there was a mostly complete libretto written by the former US poet laureate, Robert Pinsky.[14] Alex McDowell had joined the team as production designer, having worked on many landmark films such as *Fight Club* and *Minority Report*, but having no experience with stage productions. The stage director was Diane Paulus, who would soon be appointed as the Artistic Director of the American Repertory Theater (A.R.T.) at Harvard, and was known for her immersive theater *Donkey Show* and for her work on *Hair* on Broadway and the West End.

A complete summary of the piece is given in a recent paper[15] excerpted here: "The story, created by Machover with librettist Robert Pinsky and dramaturg Randy Weiner and directed by Diane Paulus, opens with a prologue set in the distant future. A troupe of robots awakens and prepares to enact an ancient, ritual pageant play about the Powers family that forms the inner narrative of the oeuvre. Four of the robots come forward and are transformed into the principal human characters in a cinematic sequence of images representing the characters' backstory. The action begins with Simon Powers, an ailing, wealthy entrepreneur and inventor, preparing to enter The System, a technological infrastructure he has created with his adopted son and research

[13] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (Pg. 46)

[14] Robert Pinsky. "Death and the Powers: A Robot Pageant". In: Poetry (2010), pp. 285–327.

[15] Jessop, E., Torpey, P., and Bloomberg, B. "Music and Technology in Death and the Powers." Proceedings of NIME. Oslo, 2011.

assistant Nicholas. The System promises to keep Powers's essence and consciousness alive beyond the moment of his death, which occurs at the culmination of the first scene. Powers is then seen transforming into his non-corporeal and omnipresent form. The drama continues on to explore the reactions of Nicholas, Simon's third wife Evvy, and Simon's daughter Miranda, as well as the outside world at large, to Simon Powers's new, technologically-immortal form. Nicholas and Evvy ultimately choose to join Powers in The System, though Miranda is hesitant to leave behind those suffering in the world. A simulacrum of Powers emerges from The System to implore her to join him and her loved ones in the realm beyond, but Miranda chooses to favor humanity and remain in the corporeal world as Simon disappears and The System sublimates. The robots appear again as the opera closes and comment on their failure to understand the themes, notably death, treated in their pageant play."

3.3.1 Initial Prototyping

Tod had an idea that the sound for *Powers* would be physical and tangible. The audience would have a sense of many sounds and be able to pinpoint their exact location. To try to test our ability to locate sounds on stage from the audience seating, we worked with Garry Geaves and Stuart Neville at Bowers & Wilkins (B&W) to design a large test rig. We built a scaffolding that was 10 feet by 15 feet, roughly the size of one of the largest set pieces in the show. We covered the front of the scaffold with acoustically transparent projection surface. Behind the screen we placed 9 custom designed B&W speakers. The speakers used aluminum 1 inch tweeters from their 700 series and 6.5 inch woven kevlar mid range drivers from their 800 series speakers mounted inside acrylic cases (laser cut and glued, then filled with cotton balls). The speakers were biamped with a series of Lectrosonics multichannel amplifiers fed by a Yamaha LS9.

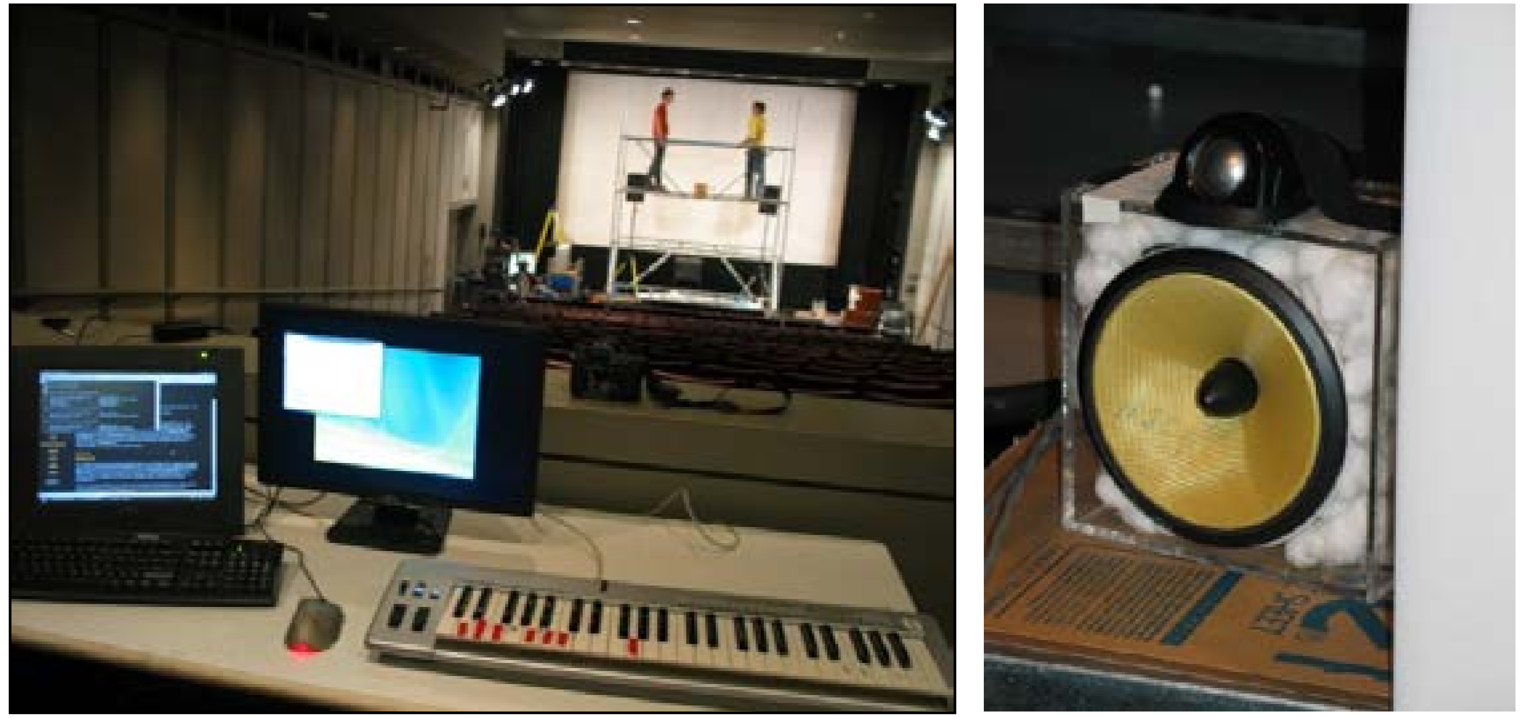


Fig. 3-1: Prototype spatialization system for Death and the Powers testing.

The LS9 was fed by a Kontakt system which was configured to round-robin its outputs via a set of MOTU 2408 interfaces. Each new sound played was routed to its own input into the mixing console, and then "panned" across the 9 speakers based on the location on the wall where that sound was supposed to originate. So, for a sound coming from the top right, the output for that speaker would be routed to the four top right speakers with balances between the four based on the closest speaker to the desired location. We projected a bookshelf on the front of the scaffolding and lit up books to simulate that book making a sound. The sounds were played through the speakers at the same time a book was lit up via projection.

Simultaneously Simone Ovsey, who was also an accomplished percussionist, climbed into the wall along with all the speakers. She would play percussion instruments from behind the screen corresponding with the location of the book lighting up.

Even with the slight differences in timing, it was overwhelming to all of us that from any reasonable distance (the audience would be at least 25 feet away) the speakers did a very good job of abstracting the position of the sounds. The percussion instruments really did not provide any greater degree of localization. Had the results been closer, we could have arranged a study, but it was clear that, especially with visual feedback, our brains tend to attach any nearby sound to the visual element being shown.

In parallel with this, I was starting to explore flexible audio systems that we could use for the performance. After working on *Skellig*, I became aware of some of the limitations of mixing desks. It seemed that for systems with large numbers of speakers, it was easy to run out of both output busses and input channels. With the performance more than 2 years away, many manufacturers I spoke to would ask "How many inputs and outputs do you need" — I couldn't answer that question without knowing more about the piece, the orchestra and the performers. Would we put speakers in the bookshelves? Would we build an ambisonic system? The performance was slated to tour, so I anticipated needing different numbers of outputs in different venues. There was no sense yet of the largest and smallest venues we would play— at one point, our rehearsal venue had

Fig. 3-2: Abstracted panning signal flow concept created and used for the ART workshop.

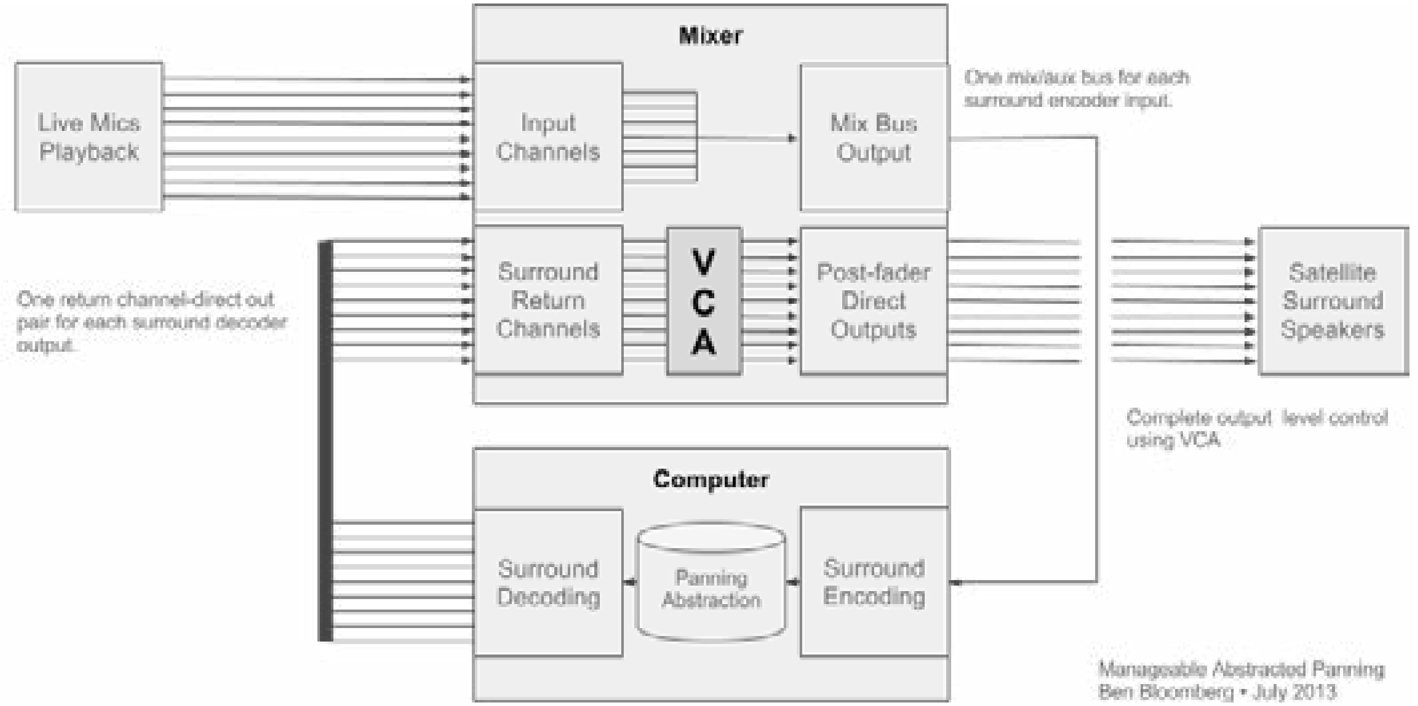


Fig. 3-3: Tod at A.R.T. with a MIDI control surface located at the center of the hall, controlling electronics for the *Death and the Powers* workshop.

Pattern control: The way a speaker transmits sound into the environment, generally plotted as spectrum and amplitude with respect to angle. See the figure below for an example of such a plot, courtesy of d&b audiotechnik.

several hundred audience seats and our performance venue had several thousand. I began to search for systems that could enable us to be flexible with our technology design as the creative vision and logistics materialized.

3.3.2 A.R.T. Workshop

To test run the libretto, the music and the production, we performed the opera with the cast and electronics at the American Repertory Theater's Loeb Theater main stage. This was the first chance I had to concretely think about the audio system for the piece. Back then we went to Parsons Audio in Wellesley, Massachusetts for each project and asked what they had in their demo stock, always trying to get things done as economically as possible. They arranged to provide an SC48 by Avid for the workshop. At the time, the house mixing desk at A.R.T.'s Loeb Center was an analog Allen and Heath console. We had to remove the mixing desk and get the SC48 installed. The house speakers were a combination of JBL and EAW, all very old models. In a rented truck, we brought over seven B&W 800 series speakers, two large 800D and five small 805D. This made up a front system for the section of the audience before the house break, which was comprised of about 8 rows of seats, or about 250 seats total. We added Mackie HR 824 speakers around the perimeter of the audience seating for surround sound. This was a point of discovery: speakers which sounded fine in smaller rooms did not work well in larger spaces. Specifically, the high end fell off the farther back one was from the speakers. After making this observation, I became quite interested in a quality of speakers I started calling "throw" but which would turn out to be dictated by ***pattern control***.

The next challenge was to determine the best way to fit all our inputs and outputs for the show on the mixing desk. The desk had many inputs but relatively few outputs. For computer IO I used the Core Audio AVID firewire driver which provided 18 inputs and outputs and additional MOTU interfaces for stereo sounds such as triggers. The ambisonic system required some inputs and outputs routed directly to the speakers. Since we were still using Max MSP, I was concerned about how to have a reliable way to control the output of the Max. The mechanism I chose to use took computer returns and routed them via input channels (which were relatively plentiful compared to outputs). Post fader direct outputs took the computer returns, which were ambisonic decoder outputs, directly to the speakers. By attaching a VCA to the channel strips, I could effectively control all the speaker outputs going to different places from a single fader.

The instrument systems for the workshop were quite similar to Skellig, consisting of a virtuosic keyboard part, a keyboard with triggers and some longer ambisonic textures. Rather than trigger the textures from the mixing desk, we set up a version kontakt with 16 outputs and loaded the textures into it, so they could be played exactly the same way as a trigger. For the workshop, it was quite difficult to hear from the rear of the house where the mixer was located. We tried to solve this in two ways: First I put a set of MIDI faders in the center of the house next to Tod's seat. This allowed him to continue mixing his own electronics. It also enabled him to trigger harmony clusters, which were sets of transpositions that were engaged on the voice of the main character, Simon Powers. We used Antares Harmony engine to do the transposition, which had amazing low frequency content with the voice of James Maddalena, who was the singer playing Simon. We also used an effect that Charles Holbrow had designed which had many sliding taps along a single delay line, meaning that the same audio signal could be dynamically delayed by different amounts. The tap points and slides could be randomized based on a few parameters making it possible to dynamically change the delay. By taking the output of this and encoding it ambisonically, it felt like a fracturing of Simon's voice all over the theater. Charles and Tod sat in the middle of the house, and Charles had a laptop that he and I used to run a text chat. When Tod had mixing comments, he would tell Charles, who would text them to me. In this way, I got a sense of how the mix should be compensated based on what I was hearing from my location. At the end of the piece for the finale, the team was able to get one of the robots functioning for the first time. It came out to take a bow, which was fitting.

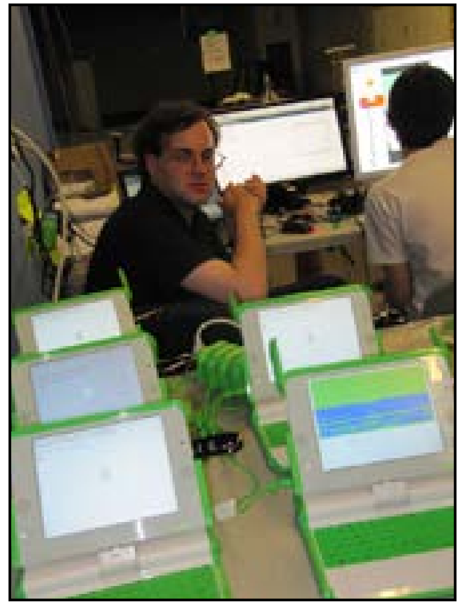


Fig. 3-4: Peter Torpey and Michael Miller provisioning OLPCs which would be placed inside the Operabots. Photo Credit: Peter Torpey

3.3.3 Bringing the Full-Scale Production to Life

After the workshop, we spent a significant amount of time regrouping and modifying our designs. I now had a much better idea of how the piece would fit together, and I was even more interested in avoiding MaxMSP. At the final workshop performance it crashed right before the first notes of the piece, and this caused an awkward pause. Diane shouted into the dark theater "What's going on!?" It was quite stressful. So I began to think about how to make the system more reliable while allowing it to function well in much larger spaces. We had been provided some photos of the hall in Monte-Carlo where we would premiere, and with these I began to understand the scale of what we would need to build.

I continued to focus primarily on audio, but there was so much to be done that I ended up helping in many different ways. We needed a reliable communication system for the entire production. Peter Torpey was, at that point, working on his Masters degree and had proposed, with Elly Jessop, to derive many parts of the production from sensors attached to James. They called this concept Disembodied Performance.[16] The sensors would capture James' movements with the intention of measuring his performance and the emotions of his character,

Fig. 3-5: The Salle Garnier at the Opéra de Monte-Carlo

Simon, and use this computational model of Simon's emotion to control may of the elements on the stage: the visuals that the audience saw, lighting for the robots, etc. To create this model, Peter developed a mapping system in Java which took all the raw sensor data and would output values such as "weight" "intensity" "complexity" - these values were fed to many systems, but mainly to another one of Peter's custom programs for creating dynamic visuals. We determined after several visual experiments that these visuals could be shown on the large set piece bookshelves, and with the right imagery the books would appear to move. Originally Alex McDowell had hoped that the books would physically move, but after some calculations it was determined that the weight required to keep a bookshelf of actuated books from toppling was on the order of tens of tons. A visual experiment determined that the books could be made to look more like they were moving by using projection on a flat book, than by actually moving the books physically. This was one of the many discoveries we made throughout the development of the production: that something faked to look real could feel more real than the real thing. This had to do mainly with the large distance from the set. Everything had to be magnified in intensity to read from the audience (similar to how stage actors are taught to exaggerate).

Since we were deriving the control of many systems from Peter and Elly's emotional mappings, we thought it would be good to carefully structure the networks and communications protocols for the performance. This became my secondary responsibility along with audio. We wanted to have every system in the performance speaking the same control protocol so they could all be linked together. For almost all existing professional theatrical equipment, there was no common control protocol, so we were breaking new ground in attempting to link different departments (sound, lighting, robots, video) together. At the time, professional production equipment utilized several competing types of time code, show control protocols, MIDI specifications, RS232 and RS485 specifications. Many devices had to be specifically integrated and very few had any kind of TCP/IP connectivity. Rather than employing these existing systems, we chose to build our own systems which would all communicate via Open Sound Control (OSC)[17]. At this point (it was 2009) OSC was popular on experimental and academic platforms like Max and SuperCollider, but did not have any adoption in the professional entertainment world.

Michael Miller had just started his Masters of Engineering in the group, with the official task of building the robot control system. He based this system on OSC as well, and we began to think about how to structure communications between his system, the sound system, and Peter's Disembodied Performance System. The Robots themselves were powered by laptops that were extra units from the Media Lab's One Laptop Per Child project. They had a 400Mhz AMD Geode. I built drivers for the CPU architecture and spent some time

[16] Torpey, P. "Disembodied Performance: Abstraction of Representation in Live Theater." M.S. Thesis. MIT Media Laboratory, 2009.

[17] Opensoundcontrol.org an Enabling Encoding for Media Applications, opensoundcontrol.org/introduction-osc.

optimizing the wireless performance of the units. We ended up running them with 5Ghz Cisco USB sticks, partially to allow the antenna to be located outside the aluminum chassis of the robots, but also to avoid the problematic 2.4Ghz. I found an access point with a beamforming antenna array inside made by Ruckus. The combination of the custom drivers, the access point and the external USB interface meant that we were able to maintain a roundtrip ping time of 0.8 ms on average to all of the 12 robots simultaneously.

Safety should be considered from the beginning; in the stress of rehearsal and production, all systems need to be fool proof and with zero possibility of harming people, as problems will inevitably occur due to the high pressure nature of the creative process.

Mike Miller designed the Java-based software stack on the robots to interface with a customized motor controller designed by Gavin Lund. The software itself was very similar to a DAW with sequences for each cue. Motors could be automated with drawn in behaviors or piloted manually. It was also possible to script Lua blocks, which could execute programmatic movements. This handled translation on omni-wheels, a motor for

Fig. 3-6: Kelsey Brigance and Karen Hart piloting Operabots above the stage at the Salle Garnier, Photo Credit: Jonathan Williams

the robot's height, and articulation of the head. Eleven addition channels of onboard lighting were included, with control automated or able to be driven directly via OSC. At times we used this to allow audio to drive the lights. Several touch screen machines with XBox controllers served as operator stations. The screens showed a top-down infrared camera shot of the stage, and each of the four operators could select a robot on the display to control.

For all parts of the control chain, the software would monitor for communication from the operators and if interrupted in any way— on wifi, connection to the motor controller, network access— the software would halt the affected robot. Ultimately this wasn't the most robust implementation. In a perfect world, we would have implemented a separate emergency stop. I had designed an implementation of this type of system: an FM transmitter with a steady tone would be connected to beam break sensors at the front and sides of the stage and E-stop buttons. Any trigger of the beam break or E-stop would cause the tone to stop, which would trigger a power kill relay on all the bots. We did not have time or resources to implement this system. During a rehearsal, one of the robots almost drove into the orchestra pit, and it was intensely traumatic for the operator (Mike) and the creative team. He had had very little sleep, and Diane Paulus asked him to execute a very complex set of manual movements close to the edge of the stage. Thankfully no one was hurt, but it was an extraordinarily good lesson: safety should be considered from the beginning; in the stress of rehearsal and production, all systems need to be fool proof and with zero possibility of harming people, as problems will inevitably occur due to the high pressure nature of the creative process.

While development on the robots was in progress, Mike and I also focused on a new Ambisonic system for the piece. I wanted to scale the system to be able to handle over 50 outputs with 32 inputs. Judging by photos of the space, we would need that many outputs to feed the various areas of the opera house. Much of the VIP seating (including the royal family's box) was sunk into the back wall of the balcony and required completely separate sound systems to allow those patrons to feel immersed in the same way as those seated in the hall. In the end we determined that installing several separately decoded Ambisonic systems, which we would then time-align together in the space, would be ideal. At this time, I had also reached out to Chris Full, who had helped with *Skellig*. He was instrumental in setting up collaborations with a Dutch speaker company, Duran Audio, and the Swiss mixing desk company, Studer. He took on the design of these elements of the system, based on the software that Mike and I had developed.

We went through many iterations of the surround system architecture, but settled on a set of Audio Units that Mike wrote, implementing the same DSP from the ICST Ambisonics Max Externals. These audio units handled encoding and decoding and were hosted in the Digital Audio Workstation (DAW) Digital Performer 8. Rather than doing summing within the DAW, the plugins would write their audio to shared memory. This was important because at the time there was no commercially available DAW with a 16-channel summing bus. The encoders would take audio from the DAW, generate 16 channels, and write those to shared memory summing in the process. The decoders would read the shared memory and use it to generate a signal for each individual speaker based on the 16 channels. To make this work, we had to trick Digital Performer into processing the plugins in the correct order. We were able to do this by setting up a dummy stereo summing bus inside the DAW which was not actually used, routing from the encoders to the summing bus to the decoders. Additionally, a command line daemon ran separately listening for OSC on the network to control the parameters of the encoders remotely. We were able to use an app called 3D Audio Scene made in MaxMSP to send OSC (for which it was relatively reliable) to control the DSP running inside Digital Performer. We used an SSL optical MADI card to connect the DP system to the mixing desk, so we were capable of 64 encoders and 64 decoders. Running the system at 128 samples of latency, we were only able to achieve about 32 encoders before seeing buffer underruns. The decoders could be processed in parallel, and since all encoders and decoders were audio units, DP handled parallel processing of the plugins quite efficiently. Since we did not look at the systems during the show, we were able to optimize parameters of DP such as "work quanta" to get more performance out of the systems. The computers running the DSP were 2009 Mac Pro towers.

We also worked with Evert Start at Duran Audio to implement his Wave Field Synthesis algorithm[18] used on Duran Audio's beamforming column array speakers. They donated a modified line array with 64 four inch speaker cones spaced very close together. Their algorithm used phase and fractional delay to simulate a virtual source in front or behind the speaker. Mike created more audio units using shared memory implementing a real time dynamic version of Evert's algorithm. Previously, their implementation only worked for static sources. Ours could take incoming coordinates and the coordinates could be changed over time to "pan" virtual sources around the stage area. The algorithm for WFS did not include summing, so the system was more efficient with each decoder calculating it's output based on all the inputs. We had 32 encoders and 64 decoders, also hosted in Digital Performer and connected to the mixing desk via MADI.

[18] Evert Start. "Direct sound enhancement by Wave Field Synthesis". PhD thesis. Delft University, 1997.

To track the robots and walls on stage, we used an Ubisense ultra-wide-band RFID[19] system which was connected to a system running software by Matt Berlin and Jesse Gray that filtered the output to remove noise. We were able to have 1 meter accuracy on the robots and people and 10cm accuracy on the walls, which were much larger and had multiple RFID tags. This location data was fed to the WFS array and used to acoustically reinforce the singers (and sometimes robots) with realistic delay and spatialization, based on their location on stage. Because the people and points were moving, and the encoding algorithm was based on delay, we had a similar effect to Charles's sliding delay lines, where a constant velocity of movement would result in a pitch shift. This was especially problematic for the singers. If they stood in one place singing and then started walking at a constant speed across the stage, their reinforcement would shift in pitch. To combat this, we limited the speed objects that were allowed to move in the system and implemented an S curve on the velocity. This effectively limited the amount of pitch shift and ensured it was never a constant. For the distances and speeds that the singers were moving on stage, this worked well as a solution making the pitch shift almost non-existent.

The mixer we chose to use was a Studer Vista 5. Unlike many of the mixers I had used before, it was extraordinarily configurable in terms of physical capabilities through expansion cards and processing capabilities in software. Based on the physical DSP cards present, one could allocate processing and IO completely flexibly. This was quite a specialized and expensive type of mixing system in 2009, although there are many desks with these features today. We could configure the number of channels, busses, auxes, groups, matrices, VCA, etc. and the processing allocated to each audio path, the number of EQs, type and number of compressors, sends, and processing emulation. This would then compile for almost an hour, and then we could load the DSP file into the desk and it would function. The maximum hardware and software configuration was on the order of 1700 inputs and outputs, so we would not come close to running out of room and could scale the configuration up and down based on the size of the hall and the rehearsal process.

Ultimately we utilized roughly 350 input paths and 250 output paths in the mixer DSP. Each of two DSP frames (D21m) was limited to 192 inputs and outputs. Five MADI cards were used, one for the stage boxes (two more D21m) fitted with 48 inputs and outputs on analog XLR connections, two MADI connections for WFS and Ambisonics processing, a third for playback and effects, and the remaining one for additional surround sound outputs via a set of SSL converters. Some hacks were required to get both stagebox inputs and outputs running on a single card. Traditionally the stageboxes required two MADI cards, since both input and

[19] Enterprise Location Intelligence. URL: http : / / www . ubisense . net / en/ (visited on 07/21/2014).

Fig. 3-7: Ben Bloomberg entering coordinates into the Ambisonics processing computer in Salle Garnier

Fig. 3-8: The arrangement of computers and mixing desk for *Death and the Powers*. Computers from left to right: remote access to Synth, Effects, WFS, Ambisonics; WinControl and 3DAS, laptop with mixing notes and DCA Assignments, Studer Vista5 Control Console

output boxes needed bidirectional communication. In our case, we were able to use the redundant MADI output port on the first card to feed the input frame, while the primary output fed the output frame. This meant that the preamp control information could be sent to the input frame. Each MADI card was set to a maximum input and output channel count so as to keep the frame total under 192 channels. The ambisonics and WFS systems only required 32 outputs each from the desk, so we could allocate the extra outputs to the SSL surround converters and effects systems. We had two computers handling keyboards, sample playback and effects. These were connected to the desk via MADI, but we used an RME MADI to ADAT matrix to distribute the channels among the two machines.

The playback machine managed the keyboards and samples for the production. We used several kontakt instances to handle stereo playback and ambisonic textures. The playback machine also ran several virtual instruments, mainly instances of Cameleon 5000 by Camel Audio (a precursor to Alchemy, now included with Logic). Rather than use the Java application that Mike developed, we transitioned to using Digital Performer and it's built in "chunk" feature to automate the keyboards and VIs[20]. DP also had a feature called "virtual rack" that allowed us to host plugins running continuously while only changing MIDI routing. DP was also able to intelligently handle transitions between patches, which meant the keyboard player could hold the pedal

[20] Digital Performer's Chunks Will Blow Your Mind, 23 Nov. 2019, www.admiralbumblebee.com/music/2019/11/23/DP-Chunks-Blow-Your-Mind.html.

down change patches and then play notes in the new patch, with the pedal release sent to the previous patch, so there was continuous sound. The effects system ran all the live instances of Antares for the harmony clusters. It also ran Charles' delay spatializer and a 250GB East West piano library. The size of the orchestra and orchestra pit meant that we could not have a real piano, so we tried to find the best sounding piano sample package that was available. All of these plugins were hosted inside another instance of Digital Performer.

Two more machines were used, one for recording multitrack versions of the performances (this was actually a Mac G5 running reaper) and another machine running audio analysis for the Disembodied Performance system. A final machine drove the visuals system connected to a large array of LEDs inside the walls, which meant that all together there were seven desktop workstations required to run the production for performance. Another 7 computers were used for the robots and walls control, with 15 laptops in the robots and walls themselves. We had two computers at the mixing desk as well, one to run the 3D Audio Scene automation software and one to run WinControl and the Studer DSP programmer which was able to program both the Duran Audio speakers and the mixing desk. All the speakers for the main PA system were connected via RS-485 which allowed us to configure their crossover voicing, EQ, time alignment and other parameters.

All together there were about 45 computers which had to work in concert to make the production successful. To allow the computers to communicate I also had the responsibility to determine the best way to structure the networking and communications. We ultimately specified seven independent networks[21] to handle various systems:

1. UbiNet - For tracking systems including all the UWB-RFID equipment
2. CoreNet - Handling all set and robot automation and control communications - including 5Ghz Wireless network for Robots
3. OSCNet - Handling communications between all systems (robots, sound, video, visuals, etc.) sharing performance control data derived from sensors. Handling tracking control data for surround sound and WFS (i.e. non safety critical production infrastructure)
4. CobraNet - handling digital audio distribution to the WFS array. This physical network required two separate VLANs to allow proper clocking between the WFS array elements and the

[21] Bloomberg, B. Death and the Powers Systems Detail Workbook. URL: http : / / web . media . mit . edu / ~benb / statics / POWERS % 20Majestic %20Documentation%20v2.1.pdf (visited on 07/21/2014).

mixing console.

5. StuderNet - handling communication and remote control of the studer mixing desk, DSP cores and control surface.
6. AudioVNCNet - Remote control access for system control laptops and iPads
7. KVMNet - a closed loop system of IP KVMs by Aten. These allowed us to pull up any workstation from a variety of consoles around the theater negating the need for VNC to the core audio computers, since VNC uses up system resources.

When we traveled to Monaco to premiere the piece, I asked my good friend Ben Maron to accompany us to manage the networks so that I could focus on the audio aspects of the production.

Once the systems were planned we had to determine how to spec the cable. This involved carefully determining every piece of equipment and the cable connections required. We were able to create a map of where components were located and which cables needed to run between those locations, which allowed us to specify the lengths for each of the paths from location to location. This became part of a much larger effort to create documentation which would make it possible for the production to tour effectively. I created a workbook with all this information, and we would modify the book for each venue where we planned to install the show. You can see a version of this book in Appendix <??> to understand the full system documentation.

3.3.4 Rehearsals and Development

The development and rehearsal process for *Powers* was a massive learning experience. A system so large and complex was riddled with bugs and it made even the most basic tasks take an extraordinary amount of time and effort. With a large portion of the team from Hollywood used to large budgets and sky-high expectations, another portion from Broadway expecting a very certain type of tech and rehearsal process, and the remaining team from MIT with no experience in either scenario and only basic production knowledge, the rehearsal and production environment became quite tense. The team came together once in the spring and then for an extended period in the summer for rehearsals.

The spring period involved full scale props with many stagehands manually puppeteering fake versions of the walls and robots to choreograph the singers along with the set. During this time, many design meetings were held to try and solve rigging and technical issues as well. The goal was to leave the rehearsal period with a clear idea of what would need to be accomplished before the summer rehearsals. The summer rehearsals would take place at the Majestic Theater in Boston with the full set, robots and orchestra.

Fig. 3-9: Robot rehearsal at the warehouse rehearsal space in Cambridge, MA. Robot operators and designers in the rear of the photo and Carol Armitage addressing them from the chair in the center.

The preparation period for these rehearsals became frantic. It was rare for the team to get more than 3-4 hours of sleep per night. The team was having a lot of trouble finishing the control for the robots. The initial design for the robot system called for their driving to be mostly autonomous, so they could be programmed to run choreographies automatically, using the UWB tracking system to have reproducible paths and trajectories. It turned out that even with filtering, the UWB wasn't accurate enough to allow the bots to drive autonomously. This was discovered several weeks before the full rehearsals were due to start, so a choice had to be made whether to try to add more location sensing to the bots or to make their translational movement completely manual. To add sensing, we had found a magnetometer and gyroscope that could be fitted to each robot, but they were several thousand dollars each and it would take significant time and resources to verify if they could solve the navigation problems. At that point, Bob Hsiung and Mike Miller made the choice to abandon autonomous translation capability of the robots and focus on stability and reliability. Even this was taking a significant amount of time.

One of Diane's assistants came out to run robot choreographies before the main rehearsals started. These rehearsals were stressful and problem-ridden. Often the rehearsal would start, a bug would be discovered and that would be the end of the day. The team would work for the rest of the rehearsal and all night to attempt to be ready for the next day's rehearsal. That rehearsal would start immediately and another bug would be found. Many rehearsals were spent this way, even into the theater. The stage management team quickly lost patience with the robot team who were, for the most part, unable to make it through a rehearsal without some problem or bug that prevented them from doing what Diane wanted to rehearse.

Fig. 3-10: Wave Field Synthesis array mounted above orchestra pit.

This was compounded by the fact that Diane was very frequently changing the choreography and blocking. The technical team would spend hours trying to make the robots do a specific movement, and then the movement would be changed to something which uncovered another bug. This happened quite a lot in the theater, with the crew, orchestra, and creative team sitting and waiting for the bug to be fixed, or having to change to another section of the piece, or rehearse without technology.

This kind of trial by fire led everyone to modify their systems to be able to better cope with Diane's style of rehearsal: rapid changes, jumping around in the score, running small segments over and over. Many existing production automation systems needed to be run on a long timeline. In these commercial systems there was not a simple way to restart from halfway through a sequence of movement; it could only be done by restoring the entire system (all set pieces, etc...) to a starting position and running from the beginning of the sequence. Because our systems were custom they did not require recompilation or reboots to modify programming, and could jump around. Ultimately, this actually saved a huge amount of time. Slowly, the bugs became fewer and fewer and the blocking stabilized, and the robot operators grew more comfortable with the music and their movements.

During this period the audio was fairly well behaved. The first orchestra rehearsal was a special experience; we had the orchestra play and slowly unmuted the sound system. It felt incredible, partially due to Chris's hard work tuning and time aligning the audio system. The PA design consisted of three main pairs of speakers (Duran T2115) hanging to the left and right of the stage. Subwoofers for this system were mounted along the sides of the stage, two B-07 15 inch subwoofers per side. And additional four 21 inch subwoofers were used for electronics and effects and a complement of 36 small U12 six inch soft-dome tweeter speakers were used for surround. Three Duran T2112 were mounted on stage and served as the main reinforcement for Simon inside The System. Finally the 64 channel 10 meter wave field array replaced the traditional front fills and was used for localization of vocals and robots. Along that array were two more U12s for music front fill.

The microphones we chose for the orchestra were a combination of classics such as the Sennheiser MKH40, Neumann U87 and KM 84 and some newer choices, such as Mojave MA200, Applied Microphone Technologies clip-ons for the strings, and others. We used reverbs from DP running on the effects machine. Our goal for the sound of the orchestra was to make it sound larger than it was, and also to allow it to feel natural, despite close micing and a challenging acoustic environment. It was instructive to experience that with good microphones, positioned properly, a good mixing desk and good speakers, not much was needed in terms of EQ. We were able to get quite subtle with the balance of levels and the use of the reverb to open up the space or make it feel intimate. After working with the orchestra alone, we added the singers. They were amplified with Sennheiser SK5212 and EM1046 wireless microphone systems, as well as MKE1 subminiature omnidirectional microphones hidden in their hair (as was common for broadway productions).

The mixing desk was configured just like a Broadway console, with VCAs assigned based on what was needed for a particular segment of the piece. We had 8 programming scenes and each scene would bring the correct faders to my fingers and group others together if necessary. I had faders for surround textures, stereo triggers, the orchestra, reverb and effects, and channels for each of the cast microphones arranged by who was present on stage at any given moment. I learned that the proper way to mix a performance like this, especially given the loud volume of the opera singers on stage, was to keep only one cast microphone open at a time. This prevents singers from leaking into one another's microphone. When this happens and both singers are moving, the phase relationship changes between the two microphones and it causes comb filtering which sounds like a flanger when movement is also involved.

To ensure only one microphone was open at a time, I had to memorize the libretto and keep my fingers on the faders at all times. I would push the fader up before each line of the piece for the correct singer, then pull it down afterwards. With two or more singers at the same time, I would have to either use one microphone (if the were close together) or both watching the relative levels (if they were across the stage from one another). This type of mixing became quite exciting to do while also shaping the electronics, orchestra and reverb.

3.3.5 Premiere in Monaco

The entire production was shipped in three 50 foot cargo containers to Monte Carlo. We arrived a month before the premiere to install the set and tech in the opera house. This involved mounting speakers in the hall while not disturbing the gold leaf detail work on the walls. For the VIP seating areas, the cable runs had to go through gold leafing, so we used 22 AWG speaker cable and the crew carefully snaked it through the plaster and gold leafing to keep it hidden. The power in the hall was turned off each night at 11:00 pm, and since there were 45 computers running, we had to install battery backup units to allow us to save our work each day and shut down while still getting the maximum amount done each day.

The robot team had been rehearsing between the Boston rehearsals and the Monte-Carlo arrival by moving physically around a large temporary venue. Having been shipped by boat, the Monte-Carlo arrive was the first time they were able to set up the robots since the Boston rehearsals a month prior. A major challenge with the deployment was the fact that the Monte-Carlo Casino was in the same building as the opera house. The casino ran special jamming equipment to block unauthorized radio communications, and all their communications equipment ran in the 5Ghz band. We ran a several day scan on the radio spectrum and determined the ideal 5.8Ghz channel to use. This changed once during the month long rehearsal period, but was ultimately reliable for the performance.

The power for the LED displays in the bookshelves presented issues: there was a set of 24 breakers which had to be turned on. Occasionally, turning one breaker on caused all the others to trip, since the initial power-on of the LED product would produce a spike of current to ground. The team discovered that using a random ordering of the breakers was the most effective way to get them all on without causing a trip, but any failure would require them to start over, since all the breakers would shut off.

We also learned that the annual Monte Carlo yacht show would be overlapping with our premiere. This was an incredibly concerning piece of news. Many of the yachts were international with long range UHF radios, exactly in the same band as our wireless microphones. There was no way to know what frequencies the boats would

Fig. 3-11: Load in in Monte-Carlo for *Death and the Powers*

use, and they would only broadcast occasionally. We did our best to monitor the spectrum, but there were a few broadcasts which interfered with our microphones during the performance. The marina was right outside the opera house and the boats came and went for two weeks during our performances. Digital wireless systems were not affordable or common yet. Today there are options to allow both transmitter and receiver to change frequencies in case of interruption. There are also systems that use multiple frequencies. We were using a Sennheiser system typical of Broadway productions, but it was not able to cope with the bursts from the yachts.

There were also some unique requirements for the premiere. The stage management team had a special camera pointed at Prince Albert's seat. The performance was required to start as he sat down in his seat. It was said that he could be up to 15 minutes early or up to 15 minutes late, so we went into standby for the top of the show 30 minutes before the stated start time and everyone waited at their posts until the Prince arrived and took his seat. This was quite suspenseful.

The premiere and following three performances were ultimately a success. All the robots and technology behaved as intended. It was quite a testament to the rehearsal and development process that was developed along the way. Each performance improved as the crew, cast and orchestra became more familiar with the piece.

3.3.6 Touring

Performances were scheduled in Boston and Chicago following the premiere in Monaco. This run was scheduled several months after the premiere in September to allow the equipment to ship back to the US by boat. I created workbooks for each city's version of the production with updated documentation to allow us to easily record positioning of all the surround sound speakers.

Boston was familiar because it was the same venue where we held our rehearsals. The Majestic Theater at Emerson was a challenging acoustic environment. The lower levels had a significantly lower reverberation time and the upper levels were quite live. We did our best to address this via matrix mixing ambience to the lower levels more than the upper levels. Because our systems were time-aligned copies of one another, this couldn't be accomplished with ambisonic panning and it brought up an interesting question of whether all audience members should hear the same content. Typically, the goal of the sound designer is to make sure the entire audience hears the same thing. However, this doesn't work so well with textures or 3D sound environments. In this project, I began to adopt a slightly different notion, that the audience should have a shared listening experience that was cohesive, but not necessarily the same. For example, if a crowd of 10,000 people witnesses a helicopter flying left to right, of course those people on the left hear more helicopter at the beginning and less at the end, while the people on the right hear more at the end and less at the beginning. In a collective experience, it is

acceptable for the audio to be inconsistent from seat to seat as long as the lack of consistency creatively supports the piece. This is an idea that is especially important for large format surround sound systems.

The performance in Chicago took place in the Harris Theater, which was a very large theater compared to the Majestic and the Opera de Garnier in Monte Carlo. Because of this, we added another 12 speakers to the PA

Fig. 3-12: First day of load-in at the Harris Theater

and surround sound systems. The goal of maintaining a natural sound in a much larger space was difficult because we had to depend more than usual on the PA system rather than the acoustic sound of the orchestra and singers. We tried several different balances of amplification and acoustic sound, and ended up pulling back amplification which made the piece feel quieter, but more natural. For the loudest moments, we were able to lean on the PA a bit more. This resulted in greater dynamic range overall which was an interesting experience, almost like the audio equivalent of an HDR photo.

In Chicago, there was also an incident with the tracking system for one of the performances that required the walls to be piloted manually instead of autonomously. As a result, the wall choreography during that performance could not be executed exactly or in the same amount of time as was typical. This was also a safety problem and had to be resolved for the following performance.

3.3.7 Remote Theatrical Experience with Powers Live

A basic description of this infrastructure is given in my Masters thesis[22] but much more detail on the development and design of the experience is provided in this section to demonstrate some of the lessons learned throughout the project.

In 2014, we were approached by the Dallas Opera to produce *Death and the Powers* one more time. The opera house in Dallas was quite a new facility that was appropriate for the piece— its stage could fit a full scale Boeing 737 jet airplane and it had many convenient infrastructural features such as easy cable runs, a special elevator for the mix position, and accommodations for projection. We had previously been approached by many other individuals about doing the piece, but many theaters did not have the right systems or were limited in height, preventing us from installing the set. Some opera companies did not have enough funding to put on the piece. The Monte Carlo production required a touring company of 72, and many opera companies did not have the resources to support that number of personnel. When it became apparent that Dallas would be able to support the production, there was also a question of bringing the production to some of the smaller venues or companies that had previously been interested in doing the show via simulcast.

We first considered some of the precedents for opera simulcasts. In general, they were quite simple, with 2-4 cameras aimed at the stage, and a pair of directional microphones picking up the acoustic audio in the space. The quality of these simulcasts (even for high budget productions by the MET for example) was still lacking, especially when compared to being there in person to watch the show live. The simulcasts were "cheap seats" that had a reduced quality of experience. This was the setup that Dallas initially proposed, but there were two

[22] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (Pg. 65)

reasons we thought this approach would not work to simulcast *Powers*. First, we feared that a simple capture of the piece with a few cameras and microphones would not be nearly sufficient to recreate the complexity of the live experience, with hundreds of loudspeakers, projection and LED visuals, and robots. Second, we fundamentally objected to the idea of "cheap seats."

With Tod we wanted to develop the idea that the remote venues should be seeing the piece from a different perspective with additional experiences and narrative to support their experience. In this way, the simulcast wouldn't feel like a lesser version of the live show, but instead a special version of a broadcast with features that weren't part of the live experience in Dallas. It took time to develop a conceptual and technical approach to achieve this, and we started with a basic narrative premise that the audience in the remote venues had already uploaded themselves into The System and were watching the story unfold from the inside out. We wanted there to be additional streams of information presented to the remote viewers to support the notion that being inside The System made one omniscient, with the possibility of having access to many vantage points simultaneously and insight that the humans in the story did not possess.

We wished to create an interactive experience for the remote audience's mobile devices to provide these extra perspectives. Peter Torpey also had the idea to make visual overlays for the broadcast that would accompany the live video footage from the stage. With these goals, it remained to be seen exactly how we could use available tools and software to achieve this reliably and with a good end result for remote audiences. In imagining the experience, we had some constraints:

- We believed that creatively the opera was already powerful by itself. We did not want to change the piece drastically or add elements that would detract (or distract) from work that had already been created and refined. This became a question of orchestration of attention of the audience and how to weave the new and old narratives together. We went through the script and determined parts of the piece where we felt the mobile devices should be dormant, parts of the piece where we thought it was acceptable for the mobile devices to be intentionally distracting, and parts where the mobile experience should be tightly tied and connected to the piece, almost as an extension of the piece.

- We wished to use the audience's mobile devices for conveying our additional narrative, but also as figurative "pixels" of The System, with a visual language that included the piece "trickling" from the screen and stage onto the devices in the remote audiences. It would therefore be important to use the same philosophy for rendering imagery and processing control data on the mobile devices and the existing production infrastructure.
- Since the existing production infrastructure was tested and well understood to be robust, reliable and predictable, we did not want to add or change any elements which would diminish the existing performance technology. We wanted to keep interfaces and connections to the remote experience as small as possible and designed so they would not interfere with existing infrastructure in any way.

We began to experiment with broadcast video technologies and mobile internet technologies to see how they could help to move us towards these creative goals.

For the mobile experience, we tested streaming video and audio technologies to see if it would be possible to send live content over the internet to audience members' smartphones. While there were many promising technologies, they were all quite dependent on the available internet connection, and we worried that trying to do live streaming to a full audience (of as many as 1000 people) would require too much bandwidth. Systems that had predictable latency could be synchronized down to a frame or two of latency, but factoring in various internet connections meant that we really had to leave a second or two buffer to account for packet loss and variable latency. This would not coexist well with the broadcast which was transmitted via satellite at a constant 400ms latency. Additionally, there was not a multicast WiFi 802.11n implementation, so we would need a very high density of access points, and this would require complex radio frequency management. We did not trust remote venues to have this expertise locally, and we did not wish to send personnel to every one of the 9 venues that had signed on for the simulcast.

Instead, we looked at other low-bandwidth real-time protocols and found WebSockets, which at the time were gaining wide acceptance in modern web browsers. Both the mobile version of the WebKit rendering engine and NodeJS had good support for WebSockets. Since they were low bandwidth, latency was simply the amount of time it took to traverse the network from the server to the device. We chose to create the experience based on cached assets that could be triggered with low latency (almost like internet triggers). We also added elements of the experience that took real-time data from the sensing in the production. This could easily be forwarded as there were already a number of libraries for translating OSC to WebSockets and back.

We designed a production and cueing system based on Java/Objective-C and WebKit for Android and iOS, NodeJS for the server and a special cuing language that Peter Torpey designed to manage assets and design of the experience. Garrett Parrish was in charge of creating the mobile application that audience members would download. We came up with the following specification for the application:

- Audience members could download the app anytime before the show.
- A versioned JSONbased show file would be downloaded that contained cues which would be executed with each trigger note played by the keyboard player in the orchestra. These triggers happened every few bars.
- The cues could have several behaviors: displaying a fragment of video, playing audio through the phone's speakers, making the phone vibrate, showing generative imagery on the phone's display, showing interactive imagery on the display where the audience could manipulate the imagery, allowing interactions with the app to be sent back to the main server to be used somehow in the live production. All of these behaviors could be chained together, crossfaded, and combined in sequences. For example, a single key trigger might start a video playing, with a generative (canvas based) overlay reacting to live data sent at 15FPS representing the amplitude of the singing voice, then all that might fade out and the phone would vibrate while a sound comes in with a 10 second fade.
- The cues could be spread probabilistically across devices, so a certain percentage of devices could be specified to run a set of cues rather than the full group.

- At home on their personal internet connection, the app would download necessary video, graphics and audio elements. This was based on what was required for the latest version of the cue list. All assets were versioned as well, with previous versions stored, so it was possible to roll back a cue list instantly if there was any error (either technically or dramaturgically).
- The app contained two modes, a pre-show mode, which allowed people to read about the experience and update their app with latest assets and cues, and a show mode which we would enable just before the performance started and would run the active cue list.
- Once at the venue, the app would assist users in joining the correct wifi network.
- Each user was given a generated ID which could be used to form groups for testing or for venues in each city.

Garrett designed the mobile application implementation, which enabled all of these features. Peter worked on the cueing structure and design of the content, and my responsibility was to create the server and internet infrastructure which would enable the production equipment to reliably send WebSocket data with Peter's content to Garrett's application. This was a difficult proposition as there was truly no way to test the server at scale before the actual production. We could not recruit several thousand testers to try the experience at the same time. Satellite time was extremely expensive, on the order of tens of thousands of dollars per hour, and without the satellite broadcast, it would be impossible to know if the video and experience were synchronized. There was no way to set up the full production beforehand because that required the full crew of 72 people and several days. How does one design a system for a truly high stakes performance (with thousands of paying audience members) having never tested the system and with only a single opportunity for it to work? We approached this by separating the system into modules and testing each of them independently at scale, then testing very carefully the interfaces between each category of infrastructure to ensure we would know how to troubleshoot and diagnose the interfaces.

To begin, I created a virtual version of the performance with video and multi-track audio from the Monte Carlo production. We extracted all the elements we would need from RAW recordings so that they would closely mimic the audio analysis we would expect for example, from the microphone in the isolation booth where James Maddalena performed as Simon in the System. In addition, we replicated the MIDI coming from the trigger keyboard used to control the samples and visuals on stage. This gave us a version of the show we could work with to design production elements and content and a "simulated" satellite broadcast. We could give the impression of satellite latency by delaying the video content in our timeline by 400ms. With this setup, it was simple to design content and test with a small number of devices. We could also be certain that if the systems we developed for the live production generated identical data to this "virtual" show, the mobile app infrastructure would be well understood and well tested.

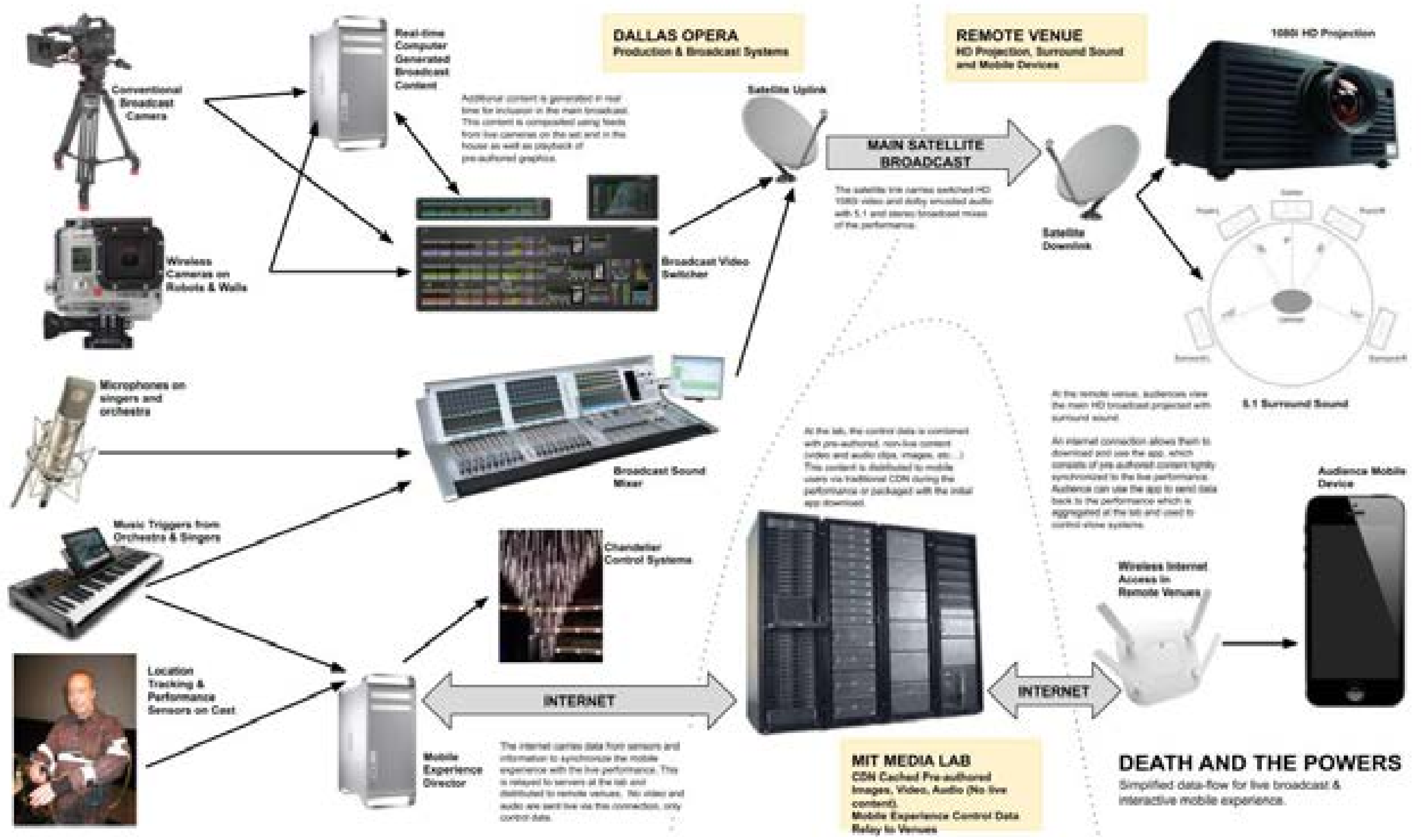
The design of the server for relaying the control data from the show to the public was intentionally quite simple, keeping no state per client and with a single configuration file). The server would spin up worker threads for each core of the CPU and read the configuration file which contained global parameters for the app. The server would then connect to an interaction host, which was located on site at the theater in Dallas. This host was responsible for sending and receiving all data from the live performance, such as trigger numbers and real time control data for audio amplitude and other analysis. The server would then relay exactly what was sent from the interaction host to all connected clients. It would remember the active trigger in case the process needed to be restarted, so that the show could resume immediately rather than waiting for the next trigger from the interaction host. This trigger was also sent if a client had to reconnect. An ID for each client was generated upon installation of the app, and added to a database of clients in Redis. The database allowed the clients to be assigned to testing groups. For testing groups, the show could be run and tested while other groups were kept on a pre-show holding page.

The following configuration options were given on the server:

- STATE_SAVE_TIMEOUT — the amount of time between saves of the active trigger
- ADMIN_SEND_INTERVAL — the frequency of updates sent to admin clients
- ADMIN_PORT — the port for interface of administration monitor

Fig. 3-13: Signal flow diagram for *Powers Live* system.

- CLIENT_PORT — the port used for connection via clients
- INTERACTION_HOST — the IP and Port for interaction host (a static IP for the show in Dallas)
- HOLDING_SCREEN — whether to display a very basic screen "Welcome to Powers Live" on all clients (asset updates and caching happen in the background)



- VENUE_WIFI — requires the client app connect to the venue's wifi before app will run. The name of the wifi network has to match a pre-approved list and the users select based on the city where their venue is located.
- WEB_VIEW — this forces all apps to display background information the projects and an onboarding process for signing up for mailing lists, showing the progress of the asset downloads, etc...
- TESTING_ONLY — this causes MIDI and performance data input from the live show to be ignored. Triggers can be sent manually via the administration interface.
- CONTENT_VERSION — the version of the cue list that should be used by all clients
- DEFAULT_PROG, DEFAULT_NOTE — in the absence of any stored state, what MIDI note should the server relay to newly connected clients
- DB_HOST, DB_PORT, DB_KEY — the Redis host connection information
- TESTING_DB, TESTING_KEY — the DB ID for test clients that should be sent cues and the full production experience even before the performance

We used two virtual machines to run instances of the NodeJS server handling websockets. An instance of HAProxy load balanced between the two servers, while internally NodeJS distributed connections to its worker processes. Since the only client state was given by the server configuration file and incoming MIDI notes, clients could disconnect and reconnect to a different worker or machine with no consequence. If something went wrong, it was possible to take an entire machine offline, reboot or change configuration and all clients would seamlessly move to the secondary virtual machine.

Another virtual machine was hosted by NeCSys at the Media Lab on a 10 gigabit internet connection. This machine contained all assets and content for download by clients served statically over HTTP. It also contained the JSON cue lists. Because we were confident that clients would download assets mainly ahead of the date of the performance, we did not use a CDN or other content cache. The content on the server was immutable, so any published cue list's assets could always be downloaded, which made it possible to easily roll back to previous cue lists.

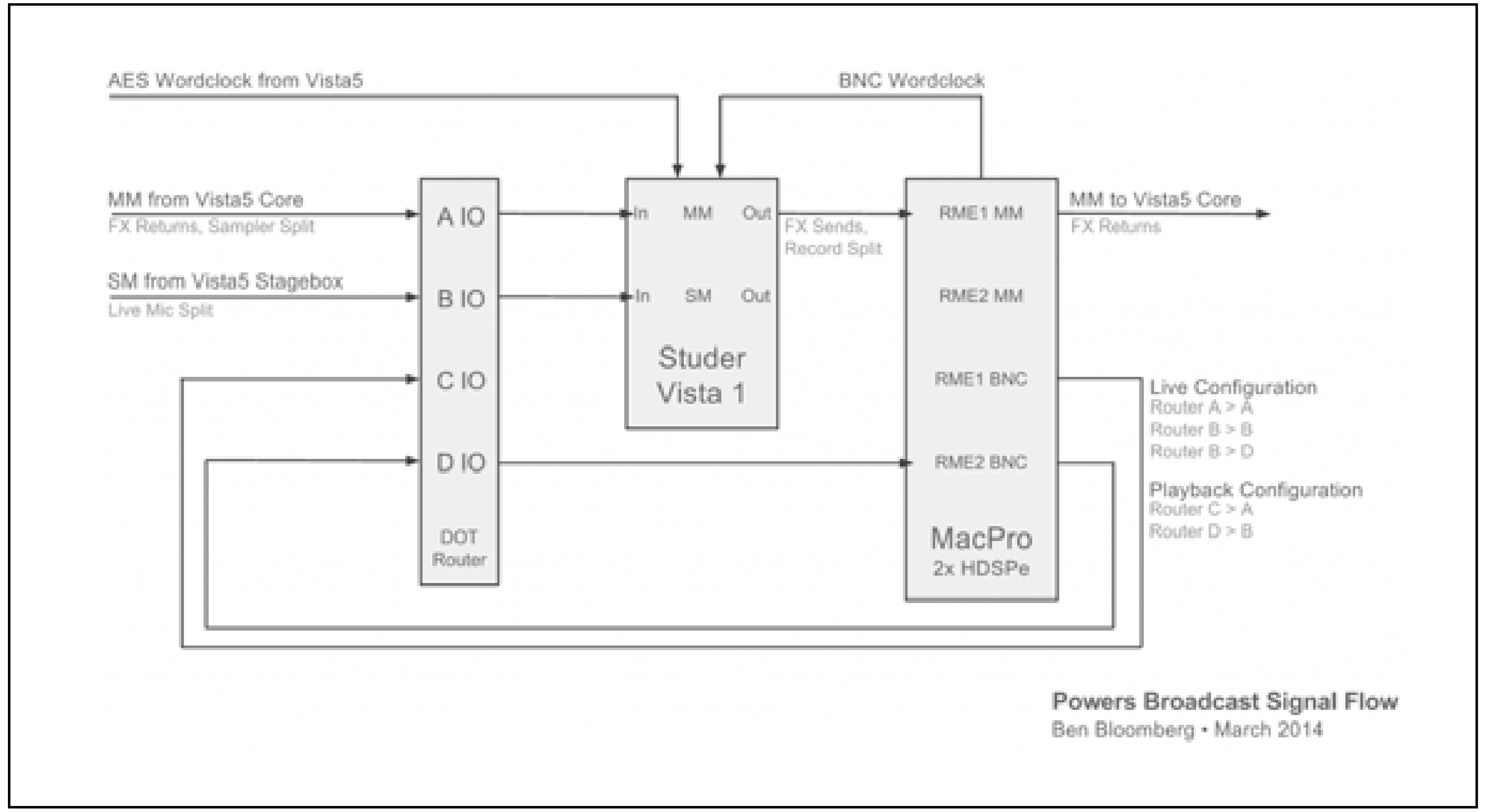
I needed a way to ensure this server infrastructure would be reliable under load from thousands of clients, so I wrote a dummy client (also in NodeJS) that implemented the same API as the real client (see client API documentation in Appendix ??). I also created an interface to monitor all connected clients, and we added reporting to the client software so the server would receive each client's state and display it, color coded based on whether the client was "in sync" with the show. Using the dummy client, we were able to spin up thousands of endpoints which would all connect to the server and request data. This gave us a way to measure bandwidth used for control data. We found that we were able to serve 8,000 clients from a single instance of Node with CPU being the bottleneck rather than network bandwidth on a 1Gb internet connection. We conservatively estimated that two VMs under load balancers would handle 14,000 clients with no problem and this was sufficiently above our remote venue capacity (roughly 4500 seats) that we felt confident in the system architecture. Tests for the mobile client included fuzzing to ensure that the client would not fail and multiply network traffic somehow, creating a loop or other issues. We then tested average latency of various networks to see how the dummy clients performed on LTE and on wifi.

All of these tests and diagnostics were used to develop a protocol for validating the internet infrastructure of the remote venues. To test latency and bandwidth, I would remote desktop into a machine running on the wifi network and ping our virtual machines, also running iPerf to check available bandwidth along the entire path from the venue to MIT. I would have the venue operators download the app, connect to the venue wifi and check for their device in the administration console, and add it to a testing group. I would trigger cues on their device as a final test to verify connectivity.

The satellite broadcast system posed its own host of challenges. We determined that to accurately capture all the action in the performance, we would need 11 cameras, including 1 wireless camera on one of the robots. Care had to be taken to ensure the camera system would not interfere with wireless control of the robots, since both systems operated in 5 Ghz. Peter's overlayed visuals had to be sent via SDI to the broadcast OB truck and embedded on program feed. We wanted the visuals to be very low latency, but completely reactive to the performance data from sensors. To achieve this, we used a quad SDI capture card feeding Quartz Composer. We used an NVidia GTX 770 Ti GPU to accelerate Quartz composer with input and output fed by the SDI card (Blackmagic Decklink Quad). The signal was sent to the OB truck and the broadcast director was given certain moments of the score where it would be featured.

To handle the sound for the broadcast, we set up a makeshift control room with a second Studer desk and 5.1 monitoring. Extra channels were added to the Ambisonic system to decode for 5.1. The record machine was replaced with a Mac Pro running a 192 channel RME MADIFace FX PCIe card. This machine would multi-track all inputs to the broadcast desk, which could be used later for playback for rehearsal of the mix. We chose Charles to mix the broadcast since he knew the text and music, but since he had never mixed the show before he would need more than just the single dress rehearsal to be able to do a good job with the mix. I had spent the

Fig. 3-14: Virtual Soundcheck routing for *Powers Live* satellite broadcast system.



previous year designing a private studio in Cambridge centered around a Euphonix System 5 mixing desk. This space was set up for live internet broadcast for musical performances with "studio quality" sound. Part of the development of the workflow for that space was the realization that it drastically improved the performance to give the broadcast audio engineer time to take rehearsal recordings, no matter how short or rough, and work with them to practice the dynamic, mix, fix EQ and compression and to find problems by soloing tracks. This drastically improved the quality of the broadcast audio and gave a sense that the recording had been mixed as if it were a studio recording. I wanted to use this concept of "virtual soundcheck" for our performance, especially since it was the first time Charles was mixing.

In an effort to save money we rented the mixing desks for the Dallas show from a 'budget' rental company as part of a large package including lighting and rigging. The desks were delivered with two separate kinds of optical cards, despite our clear specification that they would be used together for broadcast. On top of the usual card and slot configuration for the live performance, I had to switch half of the cards from one desk to the other to allow the two desks to share the stage boxes and recording system. This took two days to solve using a combination of BNC, single mode and multimode fiber, and a Digital Router, which we used to switch from live input to "virtual soundcheck" mode. The final configuration is shown in the next figure.

As a final component of the experience, we wanted to represent the presence of the remote audiences in the live stage show. Our intention was to give a sense of connection to the live experience, to make it feel larger than just the physical manifestation of the hall and venue, and to show the size and scope of The System with all of its remote inhabitants. Because interactions with the remote app were sent back to the interaction host, we could collect input from the remote audience. Our aim was to somehow integrate this control into the live performance so that the remote audiences were actually controlling the live performance. The physical dangers of the robotics and automation meant that it would be extremely difficult to allow the remote audiences to move pieces on stage. On our first site visit to the hall, we found that the space had a large chandelier which typically retracted into the ceiling before performances. The chandelier was made of 380 acrylic rods, and although it had only been used in a single color and physical arrangement it was addressable with 380 RGB channels of LED lighting and 44 separate winch motors. This made it capable of many more configurations, but the Dallas Opera did not have any documentation for how to control the lighting and motors, only a box with several presets programmed into it.

To solve this, my colleague Elly Jessop wrote an application which would spam the chandelier with data and use a camera to determine which commands controlled each of the 380 RGB channels in the chandelier. This patch was then fed to Peter Torpey's mapping and visual system, and the interaction host would aggregate the incoming control data from the mobile devices to send to the mapping and visual systems. In this way, the remote audiences directly controlled parameters of the chandelier. This raised two creative questions: how would users know what they were controlling, and how would the audience in Dallas realize there was a remote presence? Since there were not enough rods to give any single remote user individual control over a small piece of the interaction, and that would not make a particularly compelling visual effect, we thought that the next best thing would be to display an aggregate representation of activity while giving the remote user local feedback. This would result in users seeing the chandelier become more active over time as they saw other audience members around them interacting with the app. By simply tapping the screen a glow on the screen was displayed that increased in intensity with each tap. In the hall, the chandelier was increasing in frenetic energy that was appropriate for the moment, which was the climax of the piece when Simon's daughter chooses not to enter The System herself. I do not think it was clear to the live audience that the activity was actually driven in real time by the remote audiences. However, narratively and stylistically the movement made sense: as the others in The System called out along with Simon for his daughter to join him in The System, the environment around the stage and audience came to life.

3.3.8 Reflecting on *Powers*

Death and the Powers was an immensely ambitious project. Almost every aspect of the performance was designed from the ground up. Coming from so many backgrounds, the teams spent a good amount of time just learning how to communicate (i.e. Diane asking for the robots to go "as fast as possible" resulted in robots that were capable of literally killing humans). We found that it was quite important for the technologists and the creative team to communicate continuously. Left to their own devices, both would veer off and make decisions that were detrimental to the production. In a rehearsal without working robots, Diane would decide on choreography or blocking that was not possible with the robots. When making technical decisions, the teams would occasionally make assumptions that would severely limit the creative use of an object. Because of my previous experience in theater and opera, I ended up as a translator of sorts between the creatives and the engineers. I was able to immediately and clearly explain which creative requests were difficult to achieve and what small modifications would result in big time and cost savings. At the same time, I was able to spot overly ambitious promises made by the engineering team to the creative team and try to help set expectations.

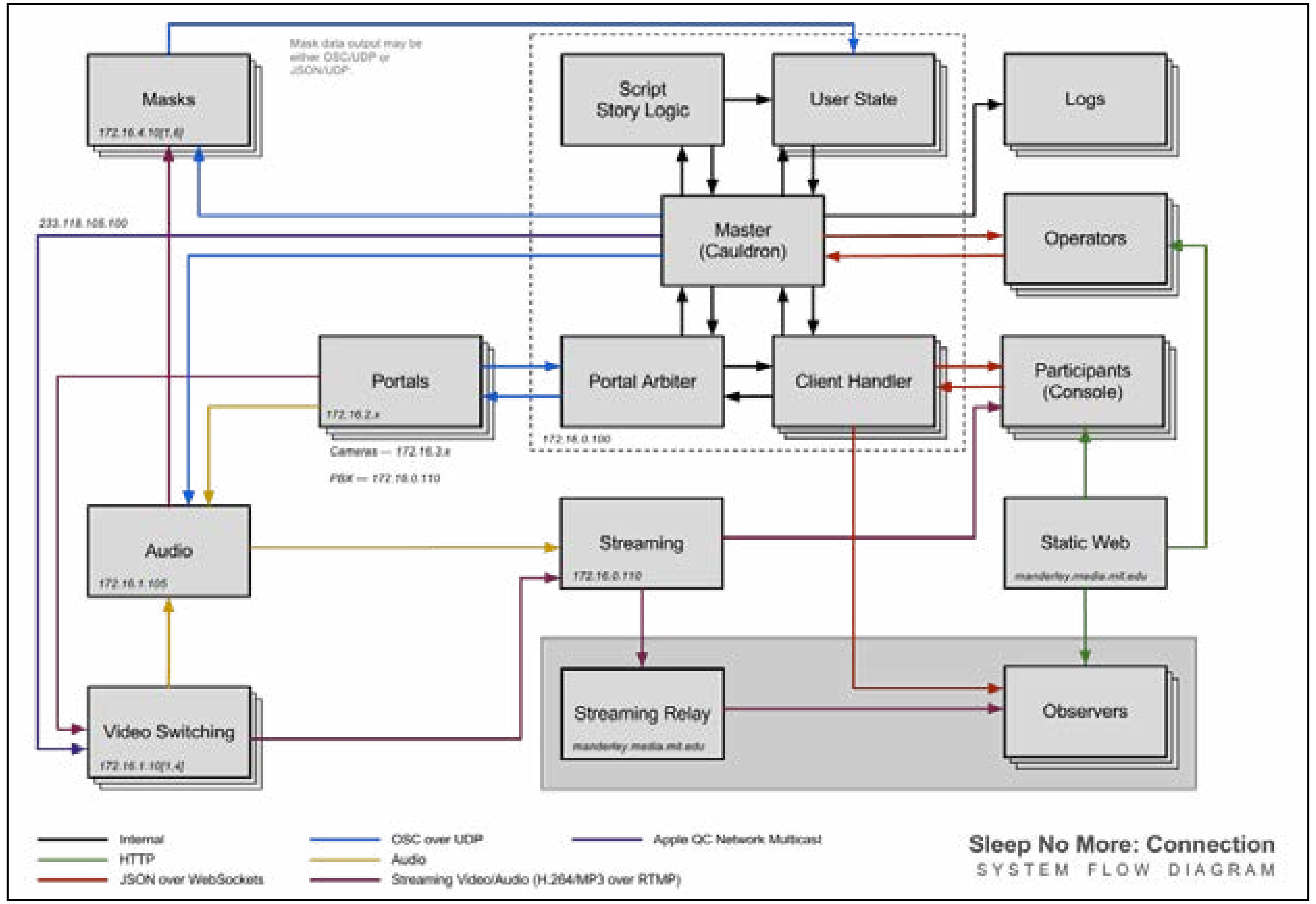


Fig. 3-15: Sleep No More system diagram. Created in collaboration with Peter Torpey

That said, the project was still extremely difficult, with almost everyone who worked on it pushed to the very limit, both emotionally and in terms of abilities. It gave me a serious understanding of how incredibly important it was to plan for surprises. It also demonstrated with great clarity how important it was to have the right combination of teams and people. By the end of the project we became good at spotting jobs that required extra people. We added a dedicated networking person (my friend Ben Maron) and a crew manager (my friend Thom Howe) to the Monaco crew to ensure we could meet the strict schedule once we arrived in Monte Carlo.

Given all of that, the technology and systems created for the piece represent a pinnacle of creative technical design, especially for the time. The systems were amazingly subtle, yet so clearly connected to the live performance and the music. Many audience members who were interviewed spoke about how the technology was not the focus of the piece, yet they acknowledged that it was the basis of much of what they felt and heard. This is what was so special about the systems in *Powers*; they were truly there to tell the story and support the creative vision. The technology was a medium, a canvas on which the story was told, customized for the exact narrative and piece, and grown along with the creative vision. I believe that growing the technology and creative vision together is the only process that results in an experience so tightly intertwined. *Death and the Powers* was the first project where I was able to see how such an interrelationship could happen and the scale which was possible to achieve. I also began to understand how critical the design of the teams could be when it came to pulling off such large projects.

3.4 Extending *Sleep No More*

Between the time of the original premiere of *Powers* and the simulcast version of *Powers* in Dallas, the creators of the New York production of the play *Sleep No More (SNM)* asked the Opera of the Future group to explore the idea of remote participation for their show. We had used the simulcast version of *Powers* to explore remote participation, and many of the lessons learned about how to design and structure the Powers Live remote system were a result of the failures of the SNM system. For that reason, I think it is instructive to document it here as well. The SNM production had a version of a score that was somewhat non-linear. It was as if each audience member were witnessing a performance of a piece in a slightly different order. This was the case because the creators of the live production (which was very carefully scripted) wanted to bring the impression of a completely free and unscripted production to an internet audience.

The challenge of the SNM system was to connect online users in an immersive personal environment (i.e. on their laptop with headphones) to on-site patrons at the live performance. Since both were given agency to explore however they liked, which was a core part of the *SNM* experience, the production elements of the online immersive experience had to be able to follow around the participants as they explored the space dynamically. In essence this required putting on a separate production for each pair of participants, one in the live space, and one at remotely at home.

3.4.1 System Overview

This system overview is excerpted from my Masters thesis.[23] However, this document will go into greater detail on the technical implementation for the systems. The Masters thesis documents the control architecture from the perspective of a non-linear mapping system which influenced the design of the Hyperproduction software.

The figure below shows the complete system layout. We'll briefly discuss how media distribution fits in with other aspects of the performance system, and then talk in detail about video, audio and telephone systems used for orchestrating the online and onsite experience.

Story Logic and User Console

A master story logic controls all aspects of the performance experience for both online and onsite participants. It sends control messages to all systems to ensure visual, aural, physical and virtual experiences stay synchronized with the script. The engine is flexible so that the story unfolds in a non-linear fashion.

Online, the user interacts with a web-based console which provides text, images, video and sound depending on the story and how online and onsite participants interact with the system.

[23] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (Pg. 52)

Portals and Masks

On site, the participants wear a mask that has been instrumented with an Android device to deliver sound through bone-conduction speakers and to collect biometric and location data. These masks lead the participants to various portals throughout the space. Portals allow online and onsite participants to communicate. The bandwidth of this communication varies greatly; some portals are simply a book flying off a shelf with specific words circled. Some send video or audio to the person on-line.[24]

3.4.2 Media Distribution

The components of the media distribution system for *Sleep No More* consist of subsystems for video, audio and telephones. Each of these subsystems has similar components:

- Input for live material from cameras, microphones or telephones
- Input for pre-recorded video or audio
- Processing of live and pre-recorded material
- Mixing of multiple processed-inputs to multiple outputs
- Encoding for distribution over the internet
- Distribution server for sending content to participants
- Client for receiving content in the browser of online users or on the android device of onsite users.
- Control interface for decoding a universal set of messages and triggers sent by the story logic into system-relevant control messages. See Appendix B for a complete listing of show control messages.

[24] Akito van Troyer. "Enhancing Site-specific Theatre Experience with Remote Partners in Sleep No More". In: Proceedings of the 2013 ACM International Workshop on Immersive Media Experiences. ImmersiveMe '13. Barcelona, Spain: ACM, 2013, pp. 17–20. ISBN: 978-1-4503-2402-1. DOI: 10 . 1145 / 2512142 . 2512150. URL: http : / / doi . acm . org / 10 . 1145 / 2512142.2512150.

Together, these components enrich the participant's experience beyond text and still imagery.

3.4.3 Video Systems

Camera Input

IP cameras placed throughout the space were used to capture live video of the performance. Cisco, Lilin and Foscam cameras were used. MJPEG encoding was chosen as the encoding format for retrieving video from these cameras on the internal network. This is because MJPEG has low-latency. Although the bandwidth required for MJPEG is greater than H264 or MPEG4, the network was designed with the intention of supporting several high-bandwidth low latency video streams.

Streams were received by video rendering nodes which handled switching and processing of the live video. A quartz composer patch running the Kineme Video Tools module accessed the streams via HTTP.

The Cisco and Lilin cameras used delivered MJEPG streams over RTSP instead of HTTP. These were remuxed on the master streaming server before being sent to the video nodes for switching. The re-muxing was accomplished in real time by VLC. This did not add any additional latency through transcoding, it simply rebroadcast incoming packets using a different media container.

VLC uses a custom boundary chunk for its MJPEG over HTTP broadcasting. This means the re-broadcasts could not be viewed in a web browser.

Playback, Switching and Processing

Playback, switching and video processing was executed on video nodes running Quartz Composer. Each node subscribed to a multicast group to receive control commands from the story logic. A clip-store on each machine played back specified pre-recorded material based on cues from the story logic. Real-time video processing was used on both live and pre-recorded clips. Quartz composer allows this processing to be accelerated by the GPU. This allows a single machine to easily render as many as 9 client streams simultaneously.

Several other methods were explored including open source projects (including LiVES, FreeJ, CamTwist, Jitter, and QLab). However, none of these solutions were capable of mixing to multiple outputs simultaneously. We would have needed to run multiple instances of the same software. Most video switching software solutions are not intended for this. Running a switching application for each user would also have prevented sharing resources in the mixer. Only one copy of each clip should be stored in memory. Processing on live inputs should only take place once for all outputs. Quartz composer allowed this flexibility.

To save physical space, each video render node was a quad-core i7 2.0Ghz mac mini with 8GB of RAM.

Encoding and Streaming for Browser Delivery

This was one of the more complex pieces of the video system. There are many methods of delivering live video to the browser but none are particularly well suited for low-latency video.

Initially the goal for this was to adhere to open standards and use HTML5 for live video streaming. However, it quickly became evident that it would be impossible to control the buffer size and thus latency for live video quickly became 15-20 seconds if there were any issues with the connection. This was not ideal for a real-time interaction.

On top of those issues, we discovered several deficiencies in the OGG container format. It's not well suited for live streaming because it uses variable length data segments. This led to freezes and a stuttering video which would stop for several minutes and play faster than real-time to catch up with itself. Both Mozilla and Google had announced their intentions to drop support for H264, and no realtime WebM encoders could easily take input from any of the matrix video mixing solutions we found.

For this reason (and with great emotional pain) we reverted back to Flash, H264 and RTMP for our real-time video distribution. It is still the standard and the tools are robust.

For encoding H264 we opted to use VLC to capture pixel-accurate areas of the screen. This was perfect because there was no direct interaction with the video nodes during the performance. In fact, there was not even a keyboard and mouse connected to those machines. The displays served as a "heads-up" monitor of all the cameras in the space and video program output being sent to participants.

A shell script launched an instance of VLC for each online participant. Their specific slice of the screen was rendered to by Quartz Composer. A raw UDP H264 videos stream was broadcast to the master streaming server, running Wowza Streaming Server.

Wowza encoded the video as RTMP and it could be accessed by clients in the browser. Wowza was chosen for its extreme reliability. Both Darwin and Icecast streaming servers were also tested. Wowza was the most performant and the only server capable of RTMP and HTML5 streaming.

Browser Client

The browser flash client for receiving video was based on a reference design by Wowza. It was modified to pull in specific video streams when triggered by javascript from the console or story logic, and also to perform GPU accelerated rendering and scaling. The video client ran as a constant layer in the background of the web console. Stylized text and background imagery were positioned above the Flash background, to obscure it when desired.

3.4.4 Audio Systems

Microphone Input

Live sound was captured in the space by microphones. Initially it was thought that live sound would be much less of a critical component for the experience. Because latency was important, special hardware was used to run the microphones directly over Cat6 cable already in the building. An audio interface connected to the main sound mixing computer captured input from two studio microphones in real time.

Once rehearsals began, it quickly became evident that we would need to find a way to accept more live audio from other parts of the space where we did not have the luxury of additional cables for studio microphones.

We used small Intel Atom development boards to capture and encode live audio from small USB microphones. The encoding was performed by DarkIce, a minimal command line Icecast broadcaster. DarkIce streamed audio from the ALSA hardware inputs to an Icecast server running on the main streaming server. The stream was then received and decoded by an instance of VLC on the sound mixing computer.

Mixing, Playback, and Routing

Audio mixing and DSP was achieved inside traditional digital audio workstation (DAW) software. In this case

Fig. 3-16: Mixing console for Extending Sleep No More Project

we chose to use Reaper for its scripting capabilities. It also can isolate DSP to multiple processes, which makes it ideal for the control infrastructure we created.

In Reaper, there were channels for each source and destination. Sources were microphones in the space and a large sampler (Kontakt 5 by Native Instruments). Destinations included each console and mask. In total there were 65 channels of audio.

Reaper ran on top of Jack Audio Router, which was responsible for all IO on the system. Streaming encoders and decoders and telephone clients connected to Jack, which could then be used as live channels inside Reaper.

Encoding

Ten instances of DarkIce were used on the mixer as well, to encode streams for the browser clients and masks. These streams were broadcast to an Icecast server. In the case of the web client streams, Wowza rebroadcast the Icecast stream so it could be picked up by the same Flash client in the browser receiving live video. With this approach, it was possible to get audio latency down to 2 seconds on the client interface.

For the masks, the native android media player API was used to retrieve the stream directly from Icecast. It would have been better to write a custom client (even running Flash) because it was not possible to control the buffer size of the native media player. This meant the latency was 10-15 seconds on the android device, causing a lot of trouble with real-time audio sent to the onsite participants. Unfortunately, there was not enough time to accomplish this.

3.4.5 Control Middleware

Using custom control middleware, the story logic was able to abstractly specify all levels of all possible channels. Thus, the story could send an event "hotel-zone-on" via OSC which would enable all audio sources for the hotel, live and pre-recorded and possibly alter effects parameters. Cues were executed in a pile-on configuration and only channels, samples and effects specified in the cues were affected by a running cue. Cues cannot be executed simultaneously; they are put into a queue and run in sequence. A future version of this automated mixing software will have simultaneous cues. The system was based on a single threaded version of the PyOSC server with custom handlers for reading in XML and executing modifications to Reaper with ReaScript.

Cues were written in XML (affectionately named BLEML after its creators) with names corresponding to their function and a specification for which channels and samples to fire. Samples were triggered via MIDI notes. A channel's volume is specified with a fade time and desired level.

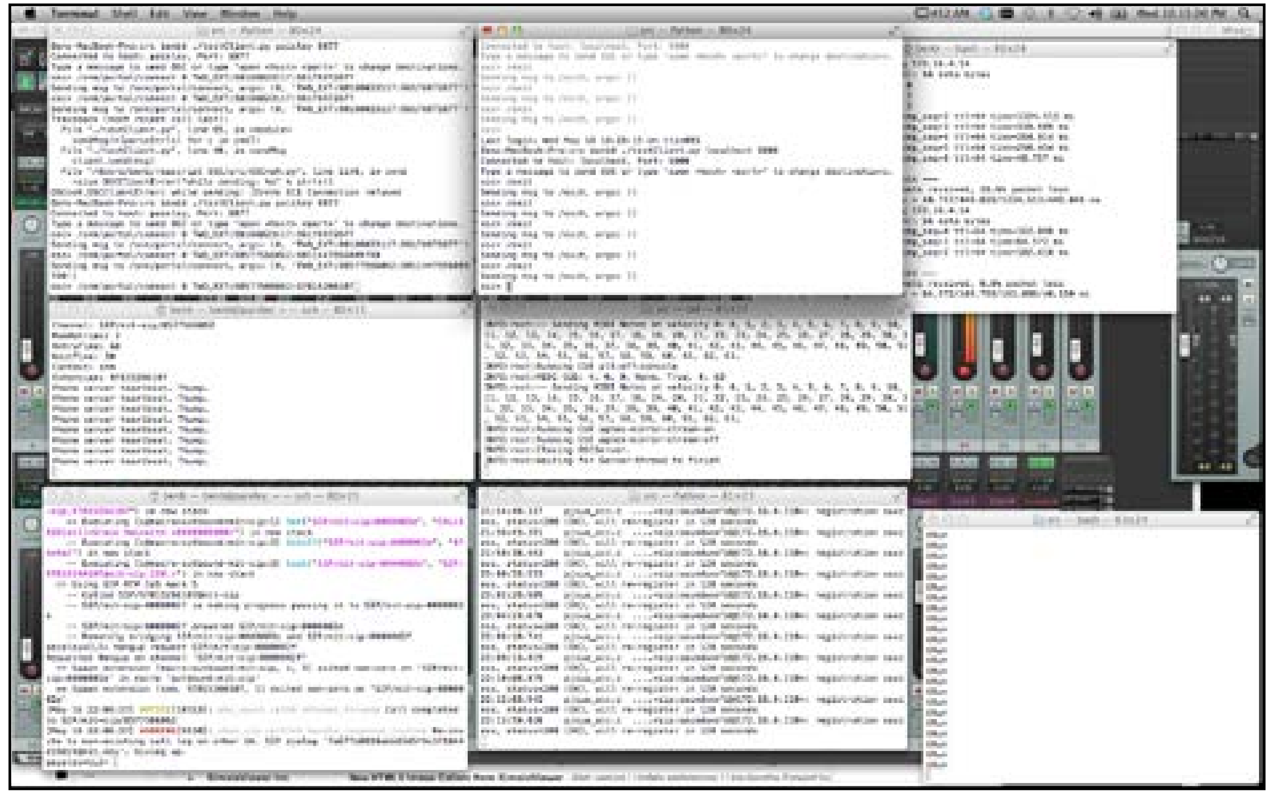


Fig. 3-17: Telephone and mixer automation console for Extending Sleep No More Project.

For each cue, a pair number is also specified. This number corresponds to the output channels the cue should be run for. For the purposes of this project, the output was a pairing of mask and online console. Each track in Reaper has a prefix which assigns it to a specific output pair. A special pair designation X means that the channel should be available to all pairs. MIDI notes to trigger samples are fired on different channels for different pairs.

3.4.6 Telephone Systems

Asterisk System

The main PBX for the telephone system was based on the open source project Asterisk. Asterisk has many types of connection protocols that allowed for very flexible integration with other subsystems.

The primary role of the Asterisk system was to initiate telephone calls to various endpoints. These included online participants' cell phones, onsite participants through several working 1940s telephones in the space, and even cast members at times.

This interfacing was achieved through a variety of software and hardware interfaces. All the interfaces used in the production communicated using SIP. For several testing sets and clients for the iPad and computer, IAX2 (Inter-asterisk Exchange Protocol) was used.

Control messages received via OSC trigger the writing of a small file which initiates a call. Calls are made from a channel device (physical gateway or SIP registration) to an extension. The extensions execute a set of commands (Dial, Wait, etc...) on a given channel. These commands can set call variables, such as caller-id, or perform operations based on the call state, playback audio files, create conferences, or even spy on other calls or extensions; these give the system its flexibility.

Physical Interfaces

ObiHai SIP gateways were used to connect physical phones in the space to the Asterisk PBX. These gateways allow one to configure a custom Tone Map. This is especially important because the 1940s telephones in the production do not use standard dial, ringback, disconnect or ringtones. Some time was spent optimizing the tone recognition so that the telephones could successfully interface with the script logic to make and receive calls from users at home.

A second ObiHai interface was installed for a standard touch-tone telephone used for testing and during the show as a hard line for the actress playing Grace, a character introduced to specifically for the virtual extension of the show. When we needed to have two simultaneous calls, the first would appear on her headphones and the studio microphone, and the second would ring in on the hard line.

Software Interfaces

Software SIP interfaces allowed integration with MIT's VOIP system and the sound mixing computer. This allowed the Asterisk PBX to initiate calls internationally and access pre-recorded and live content from microphones in the space.

MIT VOIP configuration is fairly straightforward. IS&T created an account for the project and our system registered with their servers. In order to seem mysterious, we modified outgoing caller-id so the calls originating from the show had the number 0000000000.

To integrated with the mixer and sampler, the PJUSA python wrappers were used. These have to be built from source, but once setup, they allow very flexible configuration. The telephone client based on PJUSA automatically connected itself to the Jack Audio Router whenever a call was established. This allowed for it to interface with the rest of the audio distribution systems.

3.4.7 Network Infrastructure

Network infrastructure was crucial to the project because it allowed all subsystems to communicate. The internet for the project was provided on two redundant 50Mb/5Mb Time Warner Cable connections to the performance site in New York City. Online clients were load balanced between the two internet connections and back-end servers with a Cisco 2900 router connected to a variety of 3600 and 3700 series switches. An Aironet 3600 wireless system was installed using a 2500 controller and 24 3602 access points we installed throughout the building.

Switches were interconnected using LACP over multiple gigabit links. This allowed end-to-end Gigabit Ethernet and wireless N performance for the clients and servers on the network, including IP cameras, masks, portals, and operator systems.

3.4.8 Design Process

The design was carried out over a number of weeks with the producers and directors of *Sleep No More* and the students working on the project. The specification for the system was in constant change as the needs for the project shifted and the artistic intention of the piece took concrete form.

To ensure the required flexibility each system was designed to be nearly limitless in capability and programmatically flexible, so that the specifics could be changed. A good analogy is building a theater to put on any type of show rather than a specific performance. The audio system used XML definitions (BLEML) which could be reloaded while running. This allowed us to change and edit the show as it was being created and rehearsed.

While the systems are running they can be changed and there is no difference between building a performance and running it.

Another key element of the systems was the ability for all the setup to be automated. Python scripts and shell scripts on each computer automated the process of starting programs in the correct order and in the case of audio, performing the initial signal routing between applications. Having a one-click start meant that it was possible for anyone to start and operate the systems running the show. I had to leave early (for final exams) and another student was able to take over and operate the audio systems during the remaining performances.

3.4.9 Conclusions

This was one of the most ambitious projects we had attempted. The media systems had many components that all needed to work together reliably. Much of the work putting the system together was finding programs and frameworks that would interoperate correctly. After this was completed, it was possible to script the interaction and control in Python.

The media systems were some of the most reliable (relatively) when it came to performance testing. The system never crashed during operation and it was able to cope with the instability of other systems. Specifically the live streams coming from the atom boards were notorious for disappearing in the middle of the performance. In that case the specific instance of VLC decoding the stream would die. However a Python script could quickly determine which Jack port needed to be reconnected and it would instantiate another VLC (manually triggered once we verified and restarted the atom board).

If Jack had been running for an entire day, it would sometimes lose sync and the audio would have clock issues. In many cases this could be fixed by restarting the Jack server. However, in certain cases, it required a reboot to fix. This is most likely a bug in the version of Jack being used. Restarting a process should clear out any bad state, but it did not. Each program connected to Jack had 32 virtual ports. Given that the system had 18 applications connected, this meant Jack was summing as many as 576 audio channels depending on the routing configuration. Considering that, it performed quite admirably.

The scripting interface to Reaper which was used to receive OSC and trigger cues which could change parameters or fire MIDI events based on an XML cue list was single threaded. This meant that the process listening for messages on the network would block all GUI interactions. This is where the plugin process sandboxing was very useful. It would be good to find a way to keep the script from blocking the UI. That way the mixer could be used and monitored during a performance. To get around this limitation, I created extra OSC messages which could be sent manually to monitor and operate the mixer while it was listening on the network.

The telephone system had several small bugs which we could not fix in time for the performance. First of all, although the SIP clients sent appropriate hangup events, the server had trouble disconnecting clients when calls had completed. In certain cases, this required an operator to manually hang up calls after they were finished. In the case of the physical phones, it took an extra step to verify that the person in the space had in fact hung up the receiver.

However, these are all fairly minor issues given that the entire system was designed and assembled with no prior knowledge of this type of workflow or tools suited to it. Essentially every component created had never been used or tested, so it was a good exercise in building a reliable system from fairly unknown components. The system performed well, serving content for five performances to online and onsite participants. Tweaks to the cues, audio and video stores were made throughout without causing issues or slowdowns. The streaming servers and encoders ran for the most part, continuously, without issue.

Beyond the technology, the project was a very interesting platform to explore exactly what was required to create a sense of humanity for an experience where all interactions were conducted indirectly via technology.[25] The audience had to use intermediary props, software and devices rather than experiencing the performances directly. In our evaluation (post performance interviews), we found that this sense of humanity and even the basic points of narrative were understood by some participants perfectly, while others did not understand at all that there was a person on the other end of the experience, or even the basic narrative. We found in running the experience that we needed a human operator to mediate each pair of participants. This operator also took on the role of performer, filling in for the interactive story engine where it couldn't come up with proper responses. The operators had to understand the world of the performance inside out and put themselves in the head of different characters to be able to reply as them through the chat interface. Without the operators, the experience

[25] J. Rogers D. Dixon and P. Eggleston. Between Worlds: Report for NESTA on MIT/Punchdrunk Theatre Sleep No More Digital R&D Project. University of Dundee, University of West England Bristol, 2012.

would have been completely impossible, and even with them it was still quite difficult to communicate exactly what was happening. This provided valuable insight for Powers Live where the experience of the remote audiences was also completely mediated by technology but we did not have the luxury of providing an operator for each of the thousands of audience members.

3.5 Hyperproduction

Hyperproduction is the software that I created for my master's thesis. That document provides a detail on the technical implementation of the platform and its specific advantages over existing approaches, as well as theory about data flow for analysis in performance.[26] In this document, I will provide a brief excerpt to give a technical overview of the platform, and then we will focus and reflect on the usage of Hyperproduction in practice. Much of the implementation of the advanced and "theoretical" features in Hyperproduction took place after the writing of the masters thesis, so the projects documented, especially *Fensadense,* reflect the application of Hyperproduction to real projects and its ability to actually achieve what was proposed in the masters.

Hyperproduction was born partially out of the need to modernize the Disembodied Performance mapping system (DPS) designed by Peter Torpey for *Death and the Powers*, and partially because there were many projects right after *Powers* that suggested some changes to the fundamental architecture of Peter's system could help us work more effectively. Peter's mapping system worked by allowing the user to create a graph of mathematical processes which would allow the designer to analyze incoming real time sensor data at video rates. Many other platforms were intended to achieve a similar task (MaxMSP, ROS, SuperCollider, PD) , but were not specifically intended for performance design with sensors (i.e. Hyperinstruments) and as such required additional effort or came with many inconveniences. In addition, DPS incorporated a presetting system that made the use of modes, triggers and cues much more flexible. We used Peter's system for many projects following Powers, including *Spheres and Splinters*, and Elly Jessop's thesis performance *Crenulations and Excursions*. After a few years of using the system, we realized the following:

- DPS was written in Java and core changes to its functionality or processing nodes required compilation. This meant completely reloading the system during rehearsals if changes needed to be made, and it meant working both in the application, but also in a full Java IDE along side. It also made installing the system quite difficult as it require specific versions of Java and libraries. As Java became less prevalent in modern OS, the system became harder and harder to set up, especially integrating with the OS audio and MIDI implementations.
- DPS was based on a tree architecture, which meant very efficient computation of outputs, but it also required that all outputs be accessible at the launch of the software. The result of an output node was computed by calculating the output of all nodes feeding it synchronously on every frame. If a processing graph was too large, the system couldn't maintain the proper frame rate. It also meant that circular graphs were not allowed as these could cause infinite loops.

In 2012, I began to work on a toy implementation of a similar data processing system based on NodeJS. I was so impressed with the performance of the WebSocket Powers Live server that I wanted to build something similar to Peter's system but involving significantly less code and based on an interpreted language that could be dynamically loaded and unloaded. My goal was to include an editor that would allow one to change the functionality of the program without needing to restart or recompile. I was also interested in building in all the components of the Powers Live system and making the mapping system work over the internet. Since Javascript is commonly written with asynchronous patterns, I based the system on forward propagation with no global frame rate. Nodes would simply process data when their inputs changed and update their outputs. The system would provide a global "tick" which could optionally be used to synchronize data processing or generate output sequentially. Input and output nodes could interface with other systems, such as OSC, Serial Devices, more complex protocols like ArtNET, RDM and others. The NodeJs Package Manager provided easy access to many already-working implementations, which made it quite simple to add functionality.

The following is excerpted directly from Chapter 6 of my masters thesis to give a broad overview of the architecture of Hyperproduction at the time it was created:

[26] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (pg. 57)

3.5.1 Data Architecture

The system is separated into nodes that process data, devices that connect the system to the outside world, and containers that contain nodes and devices and have some interesting properties. Connections and ports facilitate the passing of data between nodes, device and containers. The system uses a "push" model to ensure that outputs are updated as their inputs change. Let's look in detail at each component.

All objects— ports, connnections, nodes, devices and containers- have a unique ID. This ID is used to associate objects with each other and create connections that pass data. When hand-writing mappings, it is possible to specify these IDs. Future versions of the system will have a descriptive name associated with nodes and containers that does not need to be unique.

All objects have a JSON representation as well, which provides a simple means of integration with frontend or monitoring systems. Internally, the JSON represenation is used as a template to create deep copies of nodes and containers.

3.5.1.1 Nodes

Nodes are the smallest building block of the system. A node takes a definition: a number of inputs and outputs, a text description and a processing function. When created, the node instantiates ports for each input and output, and connects them to the processing function from the definition. When inputs are updated, the processing function is called to process data to outputs. The processing function has a state that can be maintained across calls. This allows the function to implement processing that takes previous input or output into account.

Keeping the definition of the node's behavior separate from the node itself facilitates simple copying of nodes. The node keeps the defintion used to create it and can therefore be duplicated very easily.

Nodes are intended to be incredibly simple to write. During production, a node could be added to the system dynamically and replicated as many times as needed, wherever the node's functionality is required. The system does not need to be restarted to add nodes; this is a benefit of using a scripted language. The intention is that a library of nodes could be accumulated over time to provide many types of processing.

Data entering and exiting the node on a port can take any form. This provides support for complex data types such as multi-dimensional input and output, strings, ENUM, etc. In the current implementation it is up to nodes to do their own type checking. This functionality is left to the node and is not executed in ports or connections to prevent overhead. A map may have thousands of connections, ports and nodes. By checking type only in nodes that require it, it is possible to have greater control of the overall performance of the system.

3.5.1.2 Connections

A connection is a unary object that connects a single output and input. An output port may have many connections, but an input port may only have a single connection. When an output port's value is updated, the port object updates all its connections, which in turn update respective input ports.

Connections can be created between any two port IDs as long as one port is an input and one is an output. Methods exist to create connections between ports by their ID directly or also with two node IDs and port names.

3.51.3 Devices

Devices are special nodes which connect the system to the outside world. A device node takes a definition that maps inputs or outputs to some control-data for a production system. Currently there are OSC Sender and OSC Receiver nodes. OSC is an increasingly common protocol that is found on many production systems—mixers, video switchers, lighting consoles, playback systems, DSP, etc. The definition for an OSC device node lists possible OSC addresses that the sender and receiver can use to communicate. The arguments for each address can be associated to ports on that node.

The OSC nodes are based on NodeJS OSC-min library and work in several modes:

Receiver nodes may:

- work in a subscription mode, where a message is sent to suscribe to data sent to the system at periodic intervals. When data is received, the corresponding output ports on the device node are updated. Ports may be defined for each argument of every input message.

- work in a polling mode, where a message is sent for each OSC address at a specified timeout when that message is received. An update command sends messages to all defined OSC addresses for each address initially.
- work in a passive mode, where no attempt is made to subscribe or poll.

Sender nodes may:

- work in an all-update mode, where any input causes all defined addresses to send their current state.
- work in single update mode, where input to the node causes only the associated OSC address to update.
- work in message send mode, where input on a port causes a specific message and arguments to be sent.

We have created basic OSC maps to prove interoperability with Behringer's x32 mixer, a 48 input 24 output digital mixing console that is OSC controllable. The x32 OSC Device node can read and send values of faders and read values of meters. There is also a basic OSC map for Blackmagic's ATEM HD-SDI video switcher, which is currently able to take a camera on the program output.

Future implementation might involve the creation of DMX, ArtNet, MSC, MIDI and Time code Device nodes. The goal of the system is to have as much interface capability as possible with as many existing control systems as possible. This facilitates the connection of many different types of systems together using this platform.

3.5.1.4 Containers

Containers hold internal nodes and connections which can be duplicated or used together as a logical grouping of functionality. Since these containers are mappings themselves and can be represented as JSON, it becomes possible to store groups of nodes that can all be instantiated together in a textual representation.

Inlet and outlet nodes are special nodes whose connections appear on the outside of the container and on the Inlet or Outlet object inside the container. These make it possible for nodes outside a container to connect to nodes inside a container. This is achieved with a special port type, EXT, that when instantiated is added to the container's port list instead of the node's. An Inlet node has a standard output Port and an EXT input port, and Outlet node has a standard input port and an EXT output port. Containers themselves behave like nodes and can therefore be embedded like a node inside other containers. A root container is instantiated automatically when the system is started to contain all devices, nodes and containers.

It is important to note that while containers may have Inlets and Outlets to facilitate external connections, it is possible to connect any input and output across the entire system through many containers, as long as both port IDs or node IDs and port names are known.

3.5.2 Advanced Features

Several more advanced features are planned for this framework and have not been completely implemented and tested. These are described in this section.

3.5.2.1 Cue-Stack Containers

- An enhanced container is planned with the following additions:
- The container has many sub-containers.
- Inlets and outlets are kept in sync between the master container and all sub containers so that all the sub-containers have the same inputs and outputs.
- A special interpolator node has all inlets and outlets of the master container.
- The connections to these inlets and outlets are switched to directly send and receive from a single sub-container based on input to the interpolator node.
- The interpolator node takes input to specify which sub-container should be used and a value to interpolate outputs to the newly selected sub-container.
- The interpolator node decides how to interpolate based on the data-type.

Given this implementation it would be possible to create a sequence of mappings and cross-fade between them. This effectively creates a structure where a list of cued interactions and be embedded inside a group-able, copy-able object. With the whole object represented in JSON, it is possible to programmatically design and instantiate cue lists. Cue-stack containers might contain specific input-output mappings or even just simple constants.

3.5.2.2 Threading

By default, NodeJS operates on a single thread. This provides good performance and in practice works well for very large mappings. It would be possible to run multiple versions of the system and have them communicate. This communication could be using standard show control protocols (OSC, MSC, etc...) or a more tailored protocol. A device or node could be designed to manage inter-process communication.

3.5.2.3 Timed Nodes

In the current implementation, nodes pass data to their outputs on any change of input. This causes the system to run ``as fast as possible.'' While this is desirable in many cases, one can easily imagine a scenario where systems should be pipelined and a desired frame-rate must be carefully maintained.

A timing node passes input to output at a specific rate rather than on any change. Both external and internal clocking mechanisms are planned. Because timing is not guaranteed in NodeJS, this sort of clocking is really only suitable for control data, not for media.

3.5.2.4 Feedback

When using a timed node, it will be possible to have nodes feed their output back to control inputs. Without imposed limits on timing, the system would get itself into an accelerating loop. Allowing feedback enables the possibility for cue lists to run themselves. An action or input can be mapped and determine the next cue or sub-container, on a cue-by-cue basis.

3.5.3 The Process of Building Hyperproduction

The basic system came together quite fast, including only the data processing parts of the system, with no graphical user interface. A close friend, Louis Mustill, was doing a project with the Swedish Radio Symphony. As a first test of the processing graph they used the system to process data collected from the movement of the conductor, Marcus du Sautoy, and generated visuals which were projected behind the orchestra. The system worked well. Next, I focused on adding a user interface with Peter Torpey. This allowed nodes to contain customized user interfaces. The test of this functionality was used for a customized wireless communication system that Brian Mayton and I designed for Ariana Grande to use with Imogen Heap's MiMU gloves on her *Honeymoon*[27] stadium tour. Rather than conducting any analysis of movement or musical expression, the system analyzed a 5Ghz wireless link and a ~470Mhz UHF wireless link for stability, automatically switching to the more stable connection.

There is a significant amount of documentation on Hyperproduction that I wrote for my masters thesis,[28] that addresses the process of initially building the software and basic UI. However the software became mature on a project called *Fensadense*, which took place after my masters thesis was finished.

[27] Lipshutz, Jason. "Ariana Grande Announces First North American Headlining Tour: See The Full Dates." Billboard, 10 Sept. 2014, www.billboard.com/articles/news/6244205/ariana-grande-2015-tour-dates-north-america-honeymoon-tour.

[28] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (pg. 91)

3.6 *Fensadense*

Fensadense, composed by Tod Machover, was originally commissioned as a piece for a children's program in the 2015 Lucerne Festival[1] where Tod was invited to be the composer in residence.[2] It was performed by a group of ten Festival alumni. Tod envisioned the piece as an exploration of the continuum between playing as a soloist and playing as part of an ensemble, moving between strict ensemble synchronization and wild solo improvisation. We also wished to use the project as an avenue to develop a new generation of hyperinstruments. Previous iterations of hyperinstruments typically relied upon very expensive custom hardware to measure performers and instruments. For Fensadense, we imagined that a new, modernized type of hyperinstrument would use inexpensive, commonly available hardware instead. We imagined that advances in sensing and data analysis would allow the use of similar hardware for many different players and instruments. In this scenario, we would measure different instruments and the playing styles of the different people by changing the software powering each instrument, rather than the hardware. With this commodity hardware and custom software, the piece would be the first in the world composed for an ensemble of hyperinstruments, or a ***hyperensemble***.

Hyperinstruments are a category of musical instrument which utilize technology to intuitively enhance and extend the expressiveness of a musical performance, providing nuanced control of additional musical layers and textures. See section 2.3 for a detailed description

We chose to use a commercial product for sensing the performance, the Thalmic Labs Myo armband. The product was intended to be used with a basic AI classifier to recognize hand gestures based on the muscle tension in the arm. This allowed one to remotely control a computer by making swiping motions in the air. Rather than use the built-in functionality, we planned to create a system that would forward the raw sensor output, consisting of 9 EMG sensors and a 3 axis IMU, to Hyperproduction. To enable the raw data capture, Garrett Parrish created another iOS application for iPad that could display the music notation for the piece while also connecting to two armbands worn by each performer. We planned that ensemble would use these iPads to connect to specific bands with hard coded bluetooth IDs. With this arrangement, pairing was more robust, as the devices were only allowed to connect with one pair of bands, based on the sheet music displayed. For example, the violin 1 part would only connect to bands labelled "violin one".

1 Kozinn, Allan. "Tod Machover Named Composer in Residence for Lucerne Festival." The New York Times, The New York Times, 20 Aug. 2014, artsbeat.blogs.nytimes.com/2014/08/20/tod-machover-named-composer-in-residence-for-lucerne-festival/.

2 "Listen: Tod Machover's 'Fensadense' for Hyperinstruments and Interactive Electronics: New Sounds Live: New Sounds." Newsounds, www.newsounds.org/story/listen-tod-machovers-fensadense-hyperinstruments-and-interactive-electronics/.

All together, there were 20 armbands for ten musicians (one on each arm), and these generated roughly 70,000 messages with EMG and IMU measurements each second. The iPads with bluetooth links to the bands were wired with Ethernet and sent these messages as OSC. The DPS system was not able to handle this quantity of data, so we chose to use Hyperproduction to process the data.

3.6.1 Initial Rehearsal Period

We planned the project in January and February of 2015, determined the concepts outlined above, and developed some initial questions and hypotheses to test with the musicians in order to determine realistically what would be possible given our timing and the technology. Tod also wrote a theme for the piece during this time as well. We then had an initial 10 days of rehearsal with the musicians in March of 2015 in Lucerne. During this period, our goal was to collect recordings of the ensemble, conduct basic tests of the sensing and test some of our ideas about more advanced recognition of playing techniques and data analysis as well. We did not have a sense of what type of analysis would be needed, so instead of trying to build and test systems during the week, our goal was to capture as much as possible to bring back to MIT to develop our analysis there. I created the following infrastructure to do this:

- An audio recording system capturing direct audio from microphones mounted on the instruments. We planned to use these same microphones in the performance. I picked the particular model (DPA 4099) for its hypercardioid pickup pattern, which would result in less bleed from the room and more direct sound from the instruments. The 4099 sounds remarkably "open," which is rare for for hypercardioid microphones. Often a tight pickup pattern impacts the spectral response of the microphone, which results in the distortion of the character of the instrument's sound. In this case the DPA 4099 provided a good solution to stage bleed and natural reinforcement.
- A data recording system, based on a new Hyperproduction node which would commit any incoming data to a file with a timestamp. I used a program called Horae[3] to receive MTC timecode from the audio recording system and relay it to Hyperproduction via OSC. This was written to the file along with Myo measurements from Garrett's iPad application.

[3] Horae, sononum.net/horae.

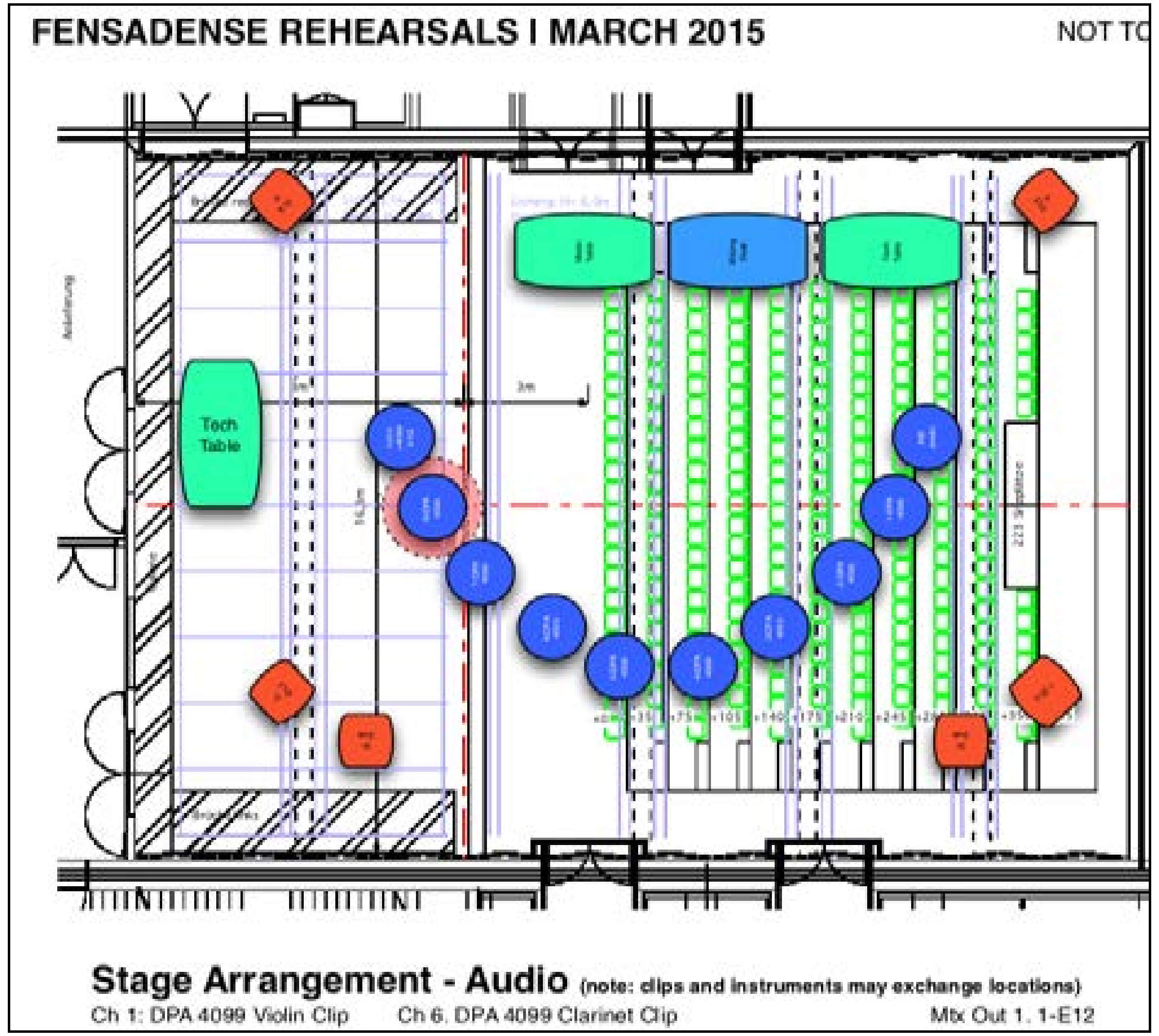


Fig. 3-18: Stage Arrangement for *Fensadense*

- A video recording system, using SMPTE LTC generated from Horae. This consisted of a DSLR camera with external audio input. On the video, the left channel contained a mono sum of all the microphones and the right channel was the LTC.

With these three recording systems, we were able to sync video, audio and data upon returning to MIT and this would allow us to test analysis on the data and audio without the performers present. In essence, this was a "virtual soundcheck" applied to data analysis.

To structure the rehearsals, we created a list of about 150 exercises, shown in the table below. For each exercise the timestamp in the data recordings was noted in the exercise sheet:

Table 3-1: Rehearsal Excersise for *Fensadense*

Morning (10:00am-1:00pm):

Goal: Gather basic individual data, test ensemble mappings on small groups.

Group 1 (10:00am - 10:40am): Violin 1, Violin 2, Viola

#	Exercise	Description	Time
	Pitch Detection		
1	Violin 1 chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	10:13
2	Violin 2 chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	10:13
3	Viola chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	10:14
4	Violin 1 chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	10:15
5	Violin 2 chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	10:16
6	Viola chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	10:17
7	Violin 1 Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	REDID TAKE
8	Violin 2 Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	10:18
9	Viola Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	10:19
10	Violin 1 Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	10:19:40
11	Violin 2 Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	10:20:07
12	Viola Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	10:20:35
	Individual Calibration		
13	Violin 1 extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	10:21:30

14	Violin 2 extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	1st take: 10:22:40 2nd take: 10:24:19
15	Viola extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	10:25:22
16	Violin 1 full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	10:26:51
17	Violin 2 full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	10:27:51
18	Viola full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	10:28:38
200	Fensadense x3 Improvised		10:34:45
201	C major scale (cued coming in and out)		10:36:30

Group 2 (11:00am - 11:40am): Cello, Bass, Tuba

#	Exercise	Description	Time
	Pitch Detection		
19	Cello chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	10:56:30
20	Bass chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	10:57:05
21	Tuba chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	10:57:40
22	Cello chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	10:58:00
23	Bass chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	10:58:35
24	Tuba chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	10:59:10
25	Cello Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	11:00:10 PITZ
26	Bass Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	11:01:20 PITZ
27	Tuba Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	11:03:02
28	Cello Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	11:03:40 PITZ
29	Bass Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	11:04:40 PITZ
30	Tuba Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	11:05:50 PITZ

	Individual Calibration		
31	Cello extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	11:06:50
32	Bass extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	11:08:02
33	Tuba extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	11:09:15
34	Cello full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	11:11:08
35	Bass full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	11:11:41
36	Tuba full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	11:12:27
202	Cello Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	11:00:55 ARCO
203	Bass Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	11:02:19 ARCO
204	Cello Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	11:04:07 ARCO
205	Bass Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	11:05:05 ARCO
206	Fensadense x3 Improvised		11:14:10
207	C major chord (cued coming in and out)		11:17:00
208	Ensemble Amplitude		11:22:00
209	Ensemble Amplitude (with smoothing)	Jack - *p*, Emilo *mf*, Justin *f*	11:28:53

Group 3 (12:00pm - 12:40pm): Clarinet 1, Clarinet 2, Keyboard, Percussion

#	Exercise	Description	Time
	Pitch Detection		
37	Clarinet 1 chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	11:57:00
38	Clarinet 2 chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	11:58:30
39	Piano chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	11:59:00
40	Percussion chromatic scale (legato) **quiet**	Quarter notes @ 120 BPM up and down. *p* dynamic. Legato connected articulation.	11:59:35

41	Clarinet 1 chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	12:00:20
42	Clarinet 2 chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	12:00:40
43	Piano chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	12:01:16
44	Percussion chromatic scale (legato) **loud**	Quarter notes @ 120 BPM up and down. *f* dynamic. Legato connected articulation.	12:01:40
45	Clarinet 1 Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	12:02:29 (2nd half F)
46	Clarinet 2 Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	12:03:50
47	Piano Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	12:04:33
48	Percussion Fensadense **quiet**	Play page 1 of *Fensadense* @ 100 BPM. *p* dynamic.	12:05:00
49	Clarinet 1 Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	12:05:30
50	Clarinet 2 Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	TAKE 1 12:06:11 TAKE 2 12:07:55
51	Piano Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	12:08:25
52	Percussion Fensadense **loud**	Play page 1 of *Fensadense* @ 100 BPM. *f* dynamic.	12:08:55
	Individual Calibration		
53	Clarinet 1 extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	12:09:55
54	Clarinet 2 extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	12:12:10
55	Piano extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	12:13:25
56	Percussion extended techniques (1 min)	Play the full range of extended techniques and effects on your instrument. Play all dynamics, tones, and techniques. This is meant so we can have an accurate picture of the types of techniques you might play.	12:15:15
57	Clarinet 1 full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	12:19:14
58	Clarinet 2 full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	12:19:53

59	Piano full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	12:22:00
60	Percussion full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	12:22:45
210	Clarinet 1 full dynamic range	Play middle C quarter notes @ BPM = 60 from *p* to *ff* to *p*.	12:21:10
211	Fensadense x3 Improvised take 1		12:28:15
212	Fensadense x3 Improvised take 2		12:30:16
213	Cued entrances		12:32:00
214	Ensemble Amplitude (with smoothing)		11:35:30
215	Ensemble Amplitude	Specific Dynamics	12:37:50
216	Ensemble Fensadenser		12:41:20

Afternoon (2:30pm-3:30pm):

Goal: Apply/test group mappings, test system robustness.

#	**Exercise**	**Description**	**Time**
	Tone Quality		
61	Play *pure* middle C x2 **quiet**	Sustain a middle C with as pure a tone as possible (close to a full sine wave). *p* dynamic.	4:39:24
62	Play *pure* middle C x2 **loud**	Sustain a middle C with as pure a tone as possible (close to a full sine wave). *f* dynamic.	4:40:18
63	Play *dirty* middle C x2 **quiet**	Sustain a middle C with a semi-pure tone (add in slight atonalities and cracks). This should reside at the middle of the pure-noise spectrum. *p* dynamic.	4:40:55
64	Play *dirty* middle C x2 **loud**	Sustain a middle C with a semi-pure tone (add in slight atonalities and cracks). This should reside at the middle of the pure-noise spectrum. *f* dynamic.	4:41:16
73	Play total noise x5 **quiet**	Play random and atonal sounds to simulate total noise. *p* dynamic.	4:41:50
74	Play total noise x5 **loud**	Play random and atonal sounds to simulate total noise. *f* dynamic.	4:42:19
	Tremolo		
75	Tremolo slow middle C x2 **quiet**	Tremolo on a middle C at a slow speed. *p* dynamic.	4:43:10
76	Tremolo medium middle C x2 **quiet**	Tremolo on a middle C at a medium speed. *p* dynamic.	4:43:47
77	Tremolo fast middle C x2 **quiet**	Tremolo on a middle C at a high speed. *p* dynamic.	4:44:15
78	Tremolo slow middle C x2 **loud**	Tremolo on a middle C at a slow speed. *f* dynamic.	4:44:40
79	Tremolo medium middle C x2 **loud**	Tremolo on a middle C at a medium speed. *f* dynamic.	4:45:00
80	Tremolo fast middle C x2 **loud**	Tremolo on a middle C at a high speed. *f* dynamic.	4:45:30
	Vibrato		
81	Vibrato slow low C x2 **quiet**	Sustain a low C and add vibrato at a slow speed. *p* dynamic.	4:45:10

82	Vibrato medium low C x2 **quiet**	Sustain a low C and add vibrato at a medium speed. *p* dynamic.	4:46:31
83	Vibrato fast low C x2 **quiet**	Sustain a low C and add vibrato at a fast speed. *p* dynamic.	4:46:51
84	Vibrato slow high C x2 **loud**	Sustain a low C and add vibrato at a slow speed. *f* dynamic.	4:47:16
85	Vibrato medium high C x2 **loud**	Sustain a low C and add vibrato at a medium speed. *f* dynamic.	4:47:34
86	Vibrato fast high C x2 **loud**	Sustain a low C and add vibrato at a fast speed. *f* dynamic.	4:47:59
	Dynamics		
87	Measure 1 of *Fensadense* (no accents, staccato) pp	Repeat measure 1 of *Fensadense* @ 100 BPM with staccato articulation. *pp* dynamic.	4:49:55
88	Measure 1 of *Fensadense* (no accents, staccato) p	Repeat measure 1 of *Fensadense* @ 100 BPM with staccato articulation. *p* dynamic.	4:50:32
89	Measure 1 of *Fensadense* (no accents, staccato) mp	Repeat measure 1 of *Fensadense* @ 100 BPM with staccato articulation. *mp* dynamic.	4:51:03
90	Measure 1 of *Fensadense* (no accents, staccato) mf	Repeat measure 1 of *Fensadense* @ 100 BPM with staccato articulation. *mf* dynamic.	4:51:30
91	Measure 1 of *Fensadense* (no accents, staccato) f	Repeat measure 1 of *Fensadense* @ 100 BPM with staccato articulation. *f* dynamic.	4:51:50
92	Measure 1 of *Fensadense* (no accents, staccato) ff	Repeat measure 1 of *Fensadense* @ 100 BPM with staccato articulation. *ff* dynamic.	4:52:30
93	C Major Scale (legato) pp	Play a C major in quarter notes scale up and down @ 60BPM with legato articulation. *pp* dynamic.	4:54:20
94	C Major Scale (legato) p	Play a C major in quarter notes scale up and down @ 60BPM with legato articulation. *p* dynamic.	4:54:37
95	C Major Scale (legato) mp	Play a C major in quarter notes scale up and down @ 60BPM with legato articulation. *mp* dynamic.	4:54:57
96	C Major Scale (legato) mf	Play a C major in quarter notes scale up and down @ 60BPM with legato articulation. *mf* dynamic.	4:55:14
97	C Major Scale (legato) f	Play a C major in quarter notes scale up and down @ 60BPM with legato articulation. *f* dynamic.	4:55:35
98	C Major Scale (legato) ff	Play a C major in quarter notes scale up and down @ 60BPM with legato articulation. *f* dynamic.	4:55:53
	~~Accents~~		

~~99~~	~~Measure 1 of *Fensadense* (no accents)~~	~~Repeat measure 1 of *Fensadense* with no accents but use normal articulation. *f* dynamic.~~	
~~100~~	~~Measure 1 of *Fensadense* (medium accents)~~	~~Repeat measure 1 of *Fensadense* with medium accents. *f* dynamic.~~	
~~101~~	~~Measure 1 of *Fensadense* (big accents)~~	~~Repeat measure 1 of *Fensadense* with large accents but use normal articulation. *f* dynamic.~~	
~~102~~	~~Measure 1 of *Fensadense* (only accents)~~	~~Repeat measure 1 of *Fensadense* and only play the accented notes with staccato articulation. *f* dynamic.~~	
	Number of People Playing		
104	Cmaj7 chord add one person at a time (L:R) x2 **quiet**	Pick and play a legato note in a Cdom7 chord and you will be added one at a time from left to right. *p* dynamic.	4:57:30
105	Cmaj7 chord add one person at a time (L:R) x2 **loud**	Pick and play a legato note in a Cdom7 chord and you will be added one at a time from left to right. *f* dynamic.	4:58:05
~~106~~	~~Bach add one person at a time (L:R) **quiet**~~	~~Play through the first page of the Bach and you will be added one at a time from left to right. *p* dynamic.~~	
~~107~~	~~Bach add one person at a time (L:R) **loud**~~	~~Play through the first page of the Bach and you will be added one at a time from left to right. *f* dynamic.~~	
~~108~~	~~Fensadense add one person at time (L:R) **quiet**~~	~~Play through the first page of *Fensadense* and you will be added one at a time from left to right. *p* dynamic.~~	
~~109~~	~~Fensadense add one person at time (L:R) **loud**~~	~~Play through the first page of *Fensadense* and you will be added one at a time from left to right. *f* dynamic.~~	
110	Cmaj7 chord randomly add/remove people x2 **quiet**	Pick and play a legato note in a Cmaj7 chord and you will be added and removed randomly. *p* dynamic.	
111	Cmaj7 chord randomly add/remove people x2 **loud**	Pick and play a legato note in a Cmaj7 chord and you will be added and removed randomly. *f* dynamic.	
112	Bach randomly add/remove people **quiet**	Play through the first page of the Bach and you will be added and removed randomly. *p* dynamic.	
113	Bach randomly add/remove people **loud**	Play through the first page of the Bach and you will be added and removed randomly. *f* dynamic.	
114	Fensadense randomly add/remove people **quiet**	Play through the first page of *Fensadense* and you will be added and removed randomly. *p* dynamic.	

115	Fensadense randomly add/remove people **loud**	Play through the first page of *Fensadense* and you will be added and removed randomly. *f* dynamic.	
	Attacks		
116	C major scale sharp attack x2	Play a C major scale up and down @ 100 BPM with a sharp attack on each note.	4:59:50
117	C major scale slow attack x2	Play a C major scale up and down @ 100 BPM with a slow attack on each note.	5:00:25
118	C major scale pizzicato (strings only, different techniques) x4	Play a C major scale up and down @ 100 BPM using pizzicato technique. STRINGS ONLY. Use different techniques each time.	5:01:05
	Myo Gesture Tracking/Calibration		
119	Everyone putting down their instrument x2		
120	Everyone assuming position x2		
123	Simon Says: Right Arm Out		
124	Simon Says: Right Arm Down		
125	Simon Says: Left Arm Out		
126	Simon Says: Left Arm Down		
127	Simon Says: Both Arms Out		
128	Simon Says: Both Arms Down		
129	Simon Says: Right Arm Forward		
130	Simon Says: Right Arm Down		
131	Simon Says: Left Arm Forward		
132	Simon Says: Left Arm Down		
133	Simon Says: Both Arms Forward		
134	Simon Says: Both Arms Down		
135	Simon Says: Both Arms Forward		
136	Simon Says: Swing Arms Back		
137	Simon Says: Both Arms Down		
138	Simon Says: Both Arms Up		
139	Simon Says: Right Arm Twist In (as far as can go)		
140	Simon Says: Right Arm Twist To Normal		
141	Simon Says: Right Arm Twist Out (as far as can go)		
142	Simon Says: Left Arm Twist In (as far as can go)		
143	Simon Says: Left Arm Twist To Normal		
144	Simon Says: Left Arm Twist Out (as far as can go)		

	SEPARATED	
230	Clean C	5:05:30
231	Dirty C	5:05:58
232	Noise	5:06:22
233	Legato Fensadense	5:08:20
234	Staccato Fensadense	5:09:30
235	C chord adding people L:R	5:10:20
236	C chord adding people LR	5:11:26
237	simon says - hey jude	
239	hey jude 1	
240	hey jude 2 second half	

We spent one day recording data, another day recording textures and sounds that were improvised by the ensemble for Tod to use in the studio, and the remaining days were spent with Tod working on the music, developing improvisations, and discussing the piece with the musicians. Tod's goal was to develop the structure of the piece collaboratively with the musicians. I will not go into detail on this process of composition here, except to mention that it required the musicians to be able to take the parts provided by Tod and notate on them all the changes and developments as they evolved. It also meant there was not a set structure or piece for us to work from when developing the technology. For this reason, much of the infrastructure we created following these rehearsals was intended to be generally applicable rather than dependent on particular music.

3.6.2 Developing the Analysis and Production

Upon returning to MIT, we had 135 GB of analysis files, including video, audio and data. The first task we undertook was to build a system to playback the data. For the rehearsals I created the recording system, but we did not have time to make a playback system. Peter Torpey worked on this and it was able to take timecode from the audio playback and synchronize the data playback. Once this was functioning, we were able to inspect the data and begin to develop technical elements of the piece.

Interestingly, we discovered it was incredibly difficult to use the data to create meaningful mappings based on the recorded data. We realized that when we developed analysis with performers present an important part of the process was trying the same actions over and over again. Although we had captured a lot of exercises, we didn't have enough data to be confident that we were creating mappings that would work well. Instead, much of our experimentation happened in the barn attached to Tod's studio. Tod would spend time in his studio writing the piece, and we would develop mappings by putting the armbands on ourselves. When we thought we

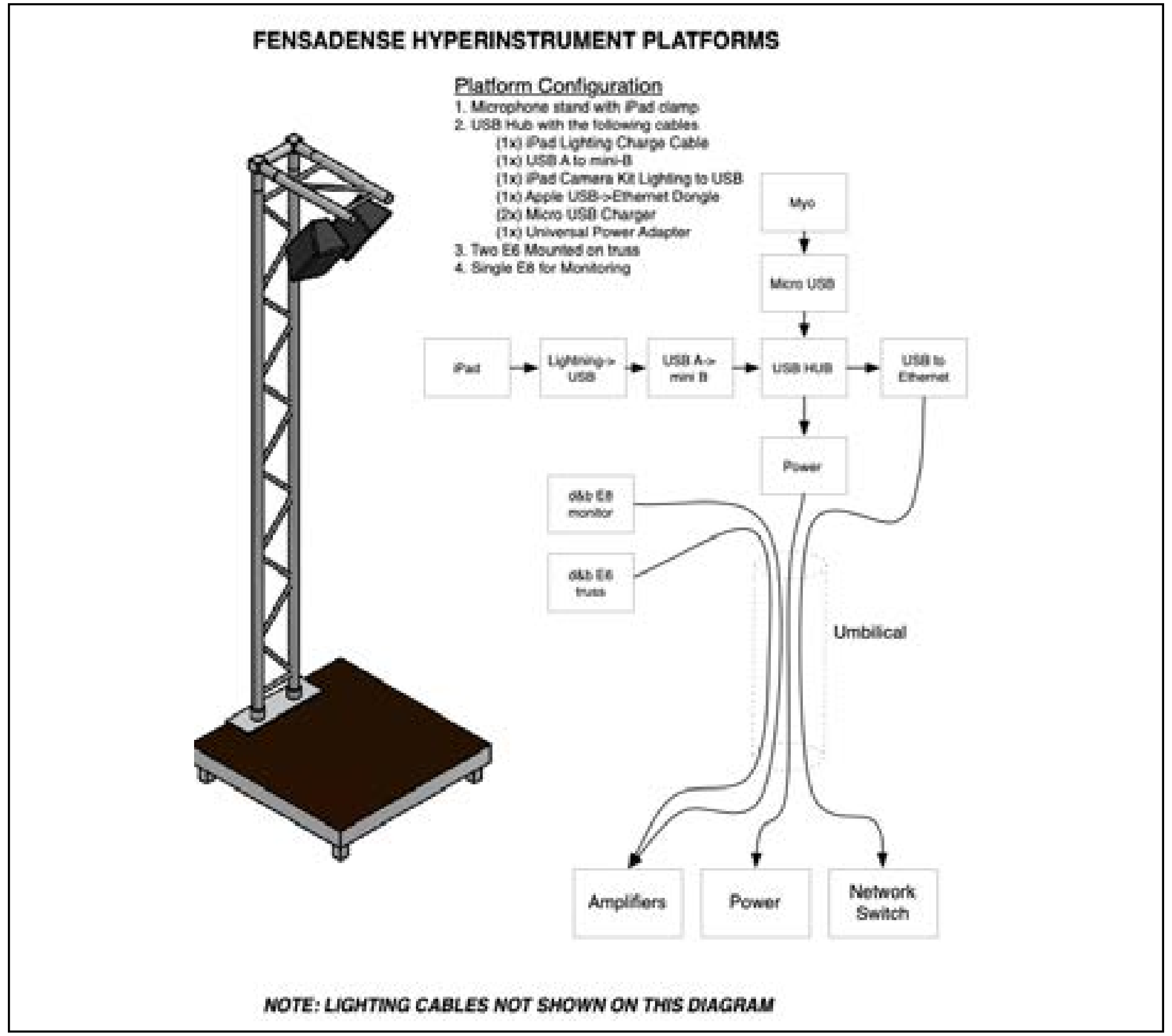


Fig. 3-19: *Fensadense* Hyperinstrument Platforms

had something usable, we would ask Tod to come play the cello and test it on parts of the piece. This ultimately provided much more information about which techniques would work well with repetition and slight variation of performance.

We were able to develop analysis which could reasonably recognize the difference between legato playing and short staccato interjections. This was primarily based on looking at the IMU data for acceleration and velocity with periodicity used to determine legato bowing, which included slow back and forth motion. We linked the staccato movements to trigger sounds from the March recording session that included short abrupt interjections. The legato playing we connected to a grain cloud which would elongate whatever section of playing happened at the start of the legato bowing. This allowed Tod to harmonize with himself by elongating a note and then keeping the same bowing motion while playing additional notes.

We also continued to develop the infrastructure and stage arrangement. The piece was commissioned along with a movement director, Shila Anaraki, and to facilitate some of the planned movements, we designed a series of mini stages with speakers and lights on each stage. Originally the stages were envisioned to be movable. As we began to see how complex the cabling for the platforms would be, this idea was scrapped.

The process of creating the system design, the analysis and working out the basics of the Myo, iOS and Hyperproduction software to enable the interactions (as well as working on the production for several other pieces[4] of Tod's which would be premiered and performed) took five months. Armed with the two interactions we had developed with Tod and the show production designs, we set out for Lucerne for a final period of almost 3 weeks of rehearsal before the premiere of the piece.

3.6.3 Final Rehearsals

In our development, we discovered that Hyperproduction was missing a lot of needed functionality which would make developing the piece and later rehearsal possible. We did not have time to add this functionality while we were developing the piece, so it was added after we arrived in Lucerne. In the span of several weeks with marathon coding sessions in the hotel in Switzerland, Garrett Parrish, Peter Torpey and I added a number of features:

- True nesting of maps to compartmentalize functionality, similar to "sub patches" in MaxMSP. This was developed to allow us to abstract away certain processes that we would reuse over and over again. It was also used to provide organization of the analysis, dividing the main analysis map up by instrument and output.
- Complete referencing system: which allowed sub maps to be saved as files and used on their own or imported into other maps. Imported or duplicated maps could be dereferenced from the saved units and modified separately, or changes could be propagated to all sub maps based on the saved files.
- Variadic nodes which change their behavior based on the number and type of configured ports.
  Numerous basic analysis nodes for doing basic stream manipulation.
- A WebSocket server similar to Powers Live.
- A preset node which added a URI based recall system for all nodes and port values.
- Complete MIDI support.
- Updated tab and pane based UI with built in code editor, graphing tools, interface widgets.
- Data recording and playback nodes for capturing a performance to time code and playing it back with audio and data synchronized.

This condensed development effort resulted in an analysis platform that was usable for rehearsals, the premiere performances and the later tour. Debugging was quite stressful as we would add features in the order they were needed for rehearsal and take rehearsal time to test them, then go back to the hotel to debug any issues. The rehearsals were scheduled for the entire day, so the debugging and implementing happened late at night or early in the morning.

The performers would update their parts as the piece slowly settled. Garrett would put these parts into the iOS app, scanned with handwritten notes. The rehearsals were also quite stressful, because the piece was not finished (which was supposed to be done with the collaboration of the players) and the technology was not fully functional. The final structure and flow was determined days before the first performance:

[4] Dizikes, Peter, and MIT News Office. "The Sound of (New) Music." MIT News, 4 Sept. 2015, news.mit.edu/2015/tod-machover-lucerne-festival-0904.

Fensadense Final Structure:

Part 1: A section of playthrough music with notated improvisation instructions. Some examples are shown in the following figures.

Fig. 3-20: Score Excerpt from *Fensadense* (mm. 1-4)

In certain cases the players would repeat phrases changing each iteration slowly.

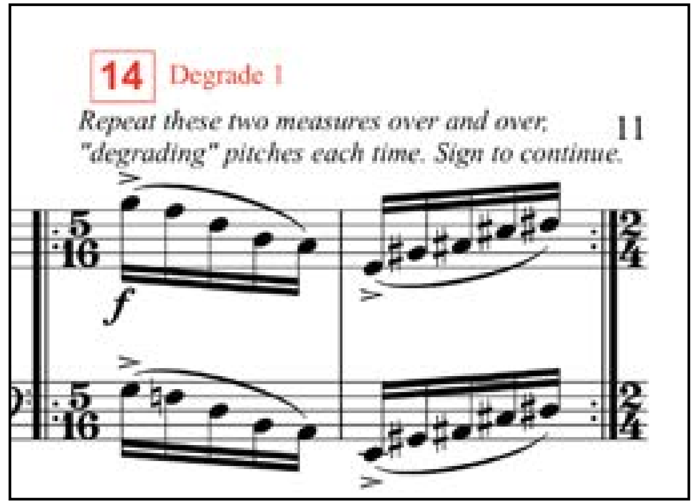


Fig. 3-21: Score Excerpt from *Fensadense* (mm. 9-10)

In certain sections, pitch or rhythm would be left to the players to improvise.

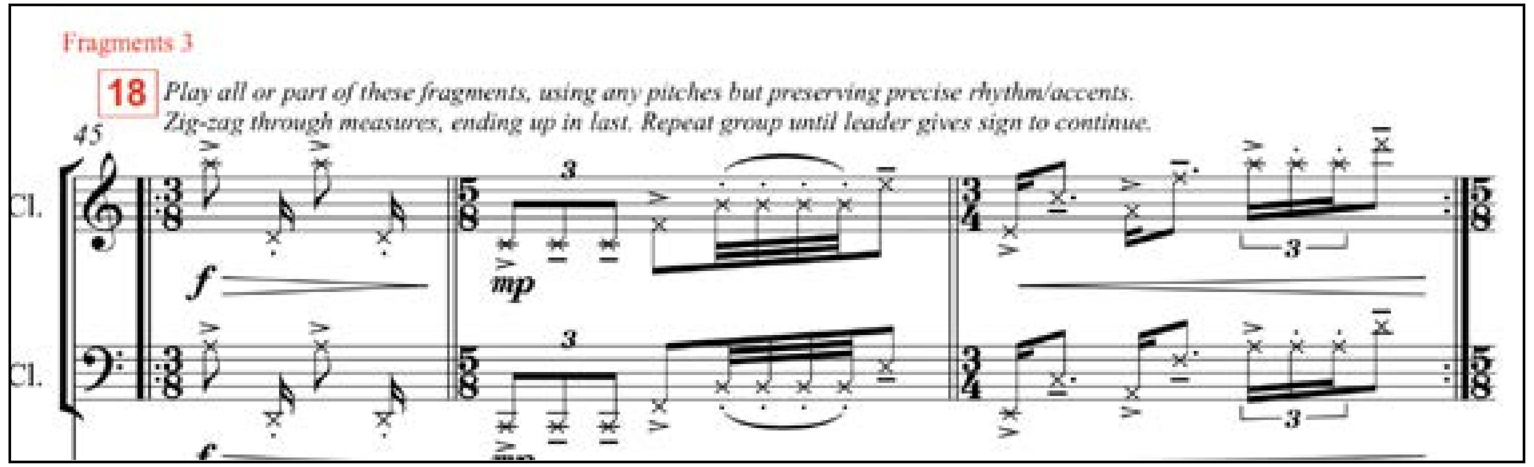


Fig. 3-22: Score Excerpt from *Fensadense* (mm. 45-47)

Particular sections used notated lines but allowed players to deviate the timing of the melodic material by "scrolling."

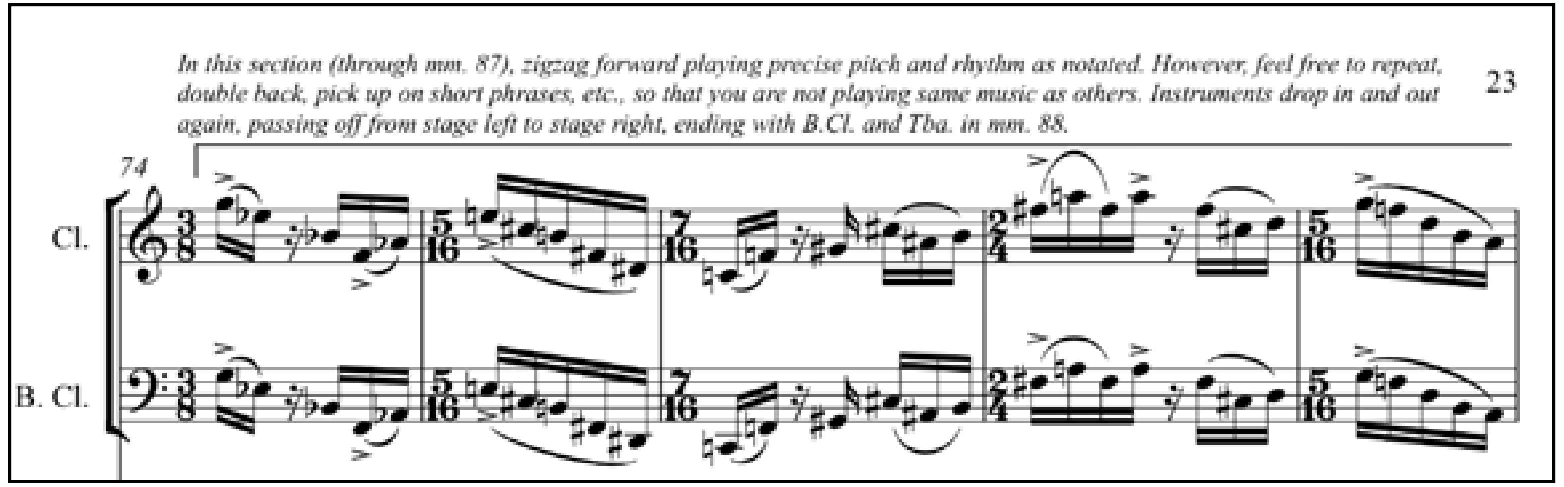


Fig. 3-23: Score Excerpt from *Fensadense* (mm. 74-78)

Part 2:
The piece would come back together with everyone playing together in perfect synchronization

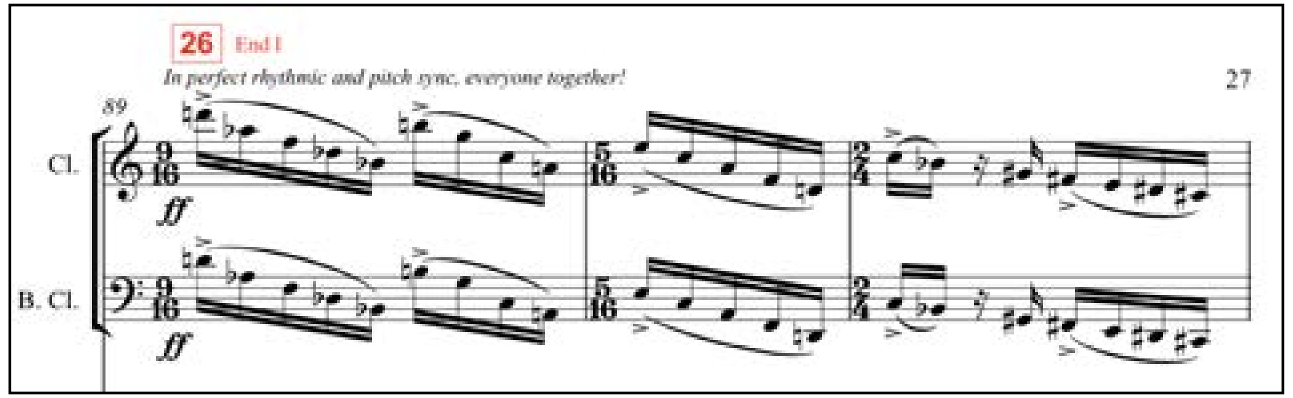


Fig. 3-24: Score Excerpt from *Fensadense* (mm. 89-91)

Part 3:
Solos with the sensor system, using a combination of staccato triggers and legato grain manipulation. Special movement and choreography was given to the soloists for the solos.

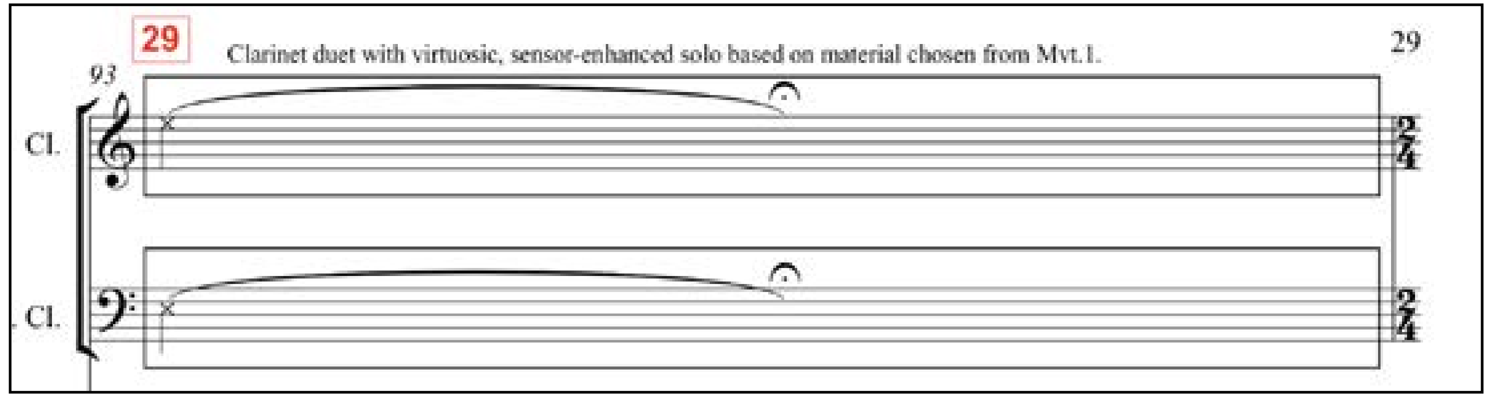


Fig. 3-25: Score Excerpt from *Fensadense* (m. 93)

Part4:
Play through music and improvisations including a drone section.

Fig. 3-26: Score Excerpt from *Fensadense* (mm. 214-215)

Part 5:
Strings solo section, with staccato interactions initially disabled involving spatialization across all speakers simultaneously. Eventually staccato interactions are re-enabled and the strings solo all together.
Part 6:
Percussion and piano sensor solos.

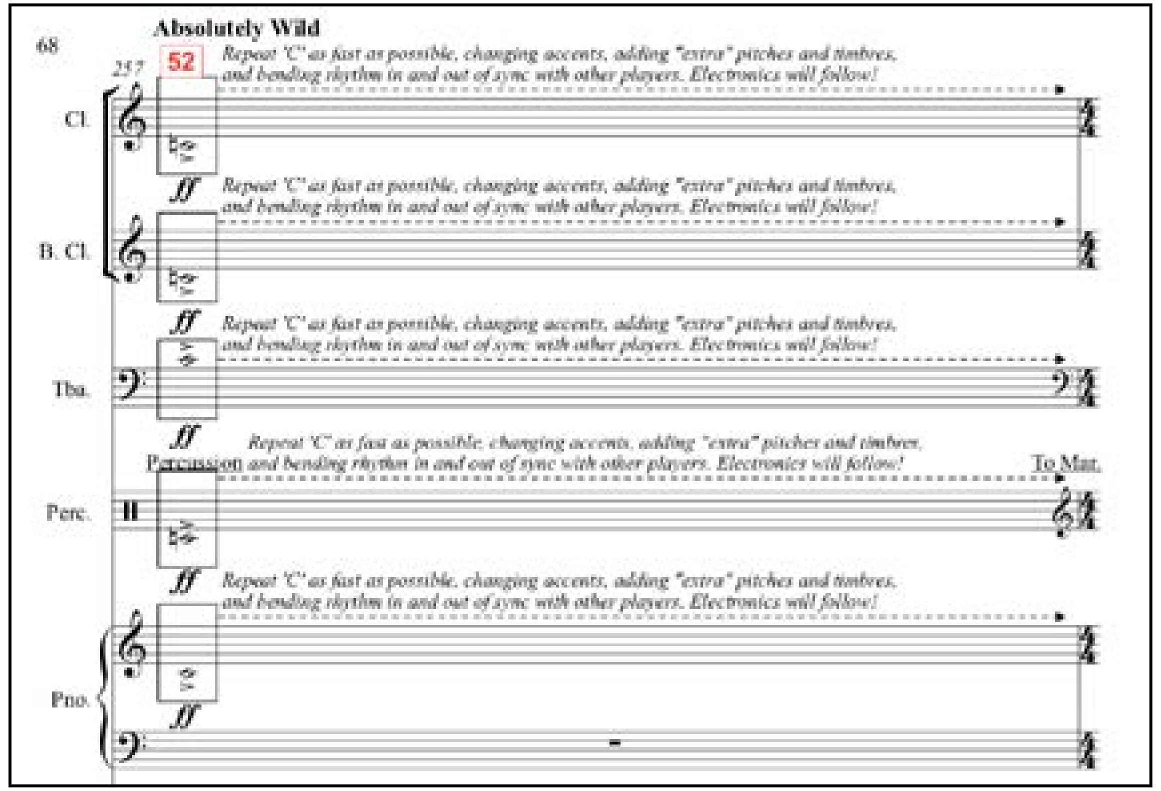


Fig. 3-27: Score Excerpt from *Fensadense* (m. 257)

Part 7:
Build up to "Absolutely Wild" C section shown below.

Part 8:
Playthrough finale, incredibly fast and synchronized with triggers to accentuate big moments.

3.6.4 Performance and Touring

Fensadense was performed 11 times over a period of one week in September 2015. The piece included both interactions as well as audio analysis that was used to generate all accompaniment and control the LED lighting strips which were mounted on the stages. There were no sounds in the piece that were triggered manually, with the exception of one trigger to finish the piece.

The performances were a chance for the piece to settle as changes were being made up to the list minute. It also provided time to create documentation for the spring tour. The piece was being taken to several theaters around Europe and this required us to make the show possible to set up by a standard sound and lighting crew. We created a 14 page User Manual that describes the systems and load-in, calibration, show and load out procedures. The performances and tours were ultimately successful although Peter Torpey accompanied the production as the visuals operator.

As part of the hand off, I had to communicate the mix to Maxime Le Saux, who operated the mixing console for this. There were many challenges to this mix, but the most complex was the balance between the live dry signal, the reverb in the space and the electronics created by the legato interaction, which had many similarities to a "shadow line" discussed in 3.2.1 for Skellig. The balance of this was continuously altered to fit with the soloist in a dynamic way, rather than a static ratio. The dynamic was based on the musical intent of the soloist, with the electronics swelling along with the arc of their phrasing and the solo as a whole. This type of swell was entirely based on emotional interpretation and could not be programmed or built into a mapping. To train Maxime to do this I had him listen as I did it several times. I started the rehearsals mixing and then would slowly hand off certain aspects to him to look after. By the end of the first few performances, I did not need to touch the mixing desk, and he was handling the entire mix and taking comments and changes from Tod. I used the same technique to later train Jacob's touring crew.

Fig. 3-28: Stage Arrangement for *Fensadense*

3.6.5 Reflections

The largest challenge with *Fensadense* was ultimately that the data from the Myo arm-bands was not detailed enough to abstract the performance quality away from the instrument and performer. In the end we found two reliable interactions based on abrupt motion and slow swaying. We were able to use these to get some sense of how frenetic the playing was regardless of instrument. The players also discovered that they could intentionally trigger both types of detection, so we attached the slow swaying to realtime transformations (grain resynthesis, delay, pitch) and the abrupt movements to samples we had recorded of extended technique for each respective player and instrument during the rehearsal period. These were spatialized according to the location of each player on an arrangement of small, detached stages. Sensing was used to drive Chauvet Epix led bars mounted on each stage as well.

Tod composed the music so it would shift from smooth and connected to quite fragmented, and this was notated in the parts for each player that were displayed on the screens of the iPads. Directions were included along with the score giving instructions for each section. The system was able to magnify the differences and similarities between these styles of playing, differentiating especially well between legato connected movement and frenetic disconnected lines. The performance was able to tour successfully and the mix, while quite complex because of the addition of live transformations, was able to be operated by an engineer rather than Tod or myself.

3.7 City Symphonies and Orchestra Imaging

The City Symphony projects started in 2012 when the Toronto Symphony Orchestra, then under conductor Peter Oundjian, commissioned Tod to create a piece for their 2013 New Creations Festival, which he was also invited to curate. Tod accepted on the condition that the entire city of Toronto be allowed to collaborate on the piece with him. At the time he made the proposal, we didn't quite understand exactly how he would collaborate with a city to make a piece of music. When the orchestra accepted his proposal we had to determine the best way to do that. This resulted in a fruitful one-year-long collaboration and the creation of *A Toronto Symphony*.[5] We initially imagined people using online apps to collect material, mash it up and remix it. Akito Van Troyer, Peter Torpey and Garrett Parrish built many such apps. One would allow users to record sounds and geotag and submit them. One would allow users to explore the sounds and create compositions by mixing

[5] "Toronto." CITY SYMPHONIES, citysymphonies.media.mit.edu/toronto.html.

them together. Another would take music Tod had written and allow the public to alter its personality and performance style. These were quite groundbreaking at the time because they all ran in the web browser, using the newly released Web Audio API, were sophisticated musically and yet were both easy to use and understand.

Unfortunately, it proved quite difficult to get people to use the apps. Although we did receive many submissions, it was from a very specific population. We feared we were missing out on contributions from people who were not tech savvy. To help address this, the orchestra hired a community organizer, Jennie Green, who set up workshops and interventions all over the city with as many different types of people as we could find. We tried for a big mix of ages, backgrounds, interests, and geographic areas. Additional collaborations were set up with the CBC and in the school systems,[6] where students were given the opportunity to create pieces about the city using composition software, developed in the Opera of the Future group, called Hyperscore.[7]

We developed a complex visuals system to go along with the piece that controlled triggered live projections in the hall and LED animations on the CN Tower, all based on the live playing of the orchestra. The concert was streamed via the CBC and we attempted to match the latency of the Tower lighting to the latency of the stream, so that listening to the concert outside of the hall resulted in a synchronized video and light show. The core of the system was similar to Powers, where MIDI triggers for the piece also triggered visual fragments. We attempted to use QLab[8] to achieve this rather than using Peter's custom engine. QLab served as the master cue list and playback of video in the hall. These cues also triggered the remote Light System Engine for the CN Tower and a Blackmagic Aten video switcher for the projection and live stream program output, at times keying titles or selecting Quartz Composer patches and a live video feed of the CN Tower as other sources. This arrangement proved to be bug ridden, hard to work with, and unstable. It made the production process quite stressful. QLab did not have OSC[9] support at the time, so we were using Apple Script to call python to send OSC. We also ran into RAM issue on the Philips Light System Engine[10] generating the control signals for the Tower infrastructure. Our content crashed the server up until the day of the performance, and we had to take

[6] Lumaga, M. B., "A Toronto Symphony, Tod Machover's participatory orchestral opera." LOFT Magazine, August 2013

[7] Farbood, M., Pasztor, E., Jennings., K. "Hyperscore: A Graphical Sketchpad for Novice Composers." IEEE Computer Graphics and Applications, January–March 2004.

[8] "QLab Overview." QLab Icon, qlab.app/overview/.

[9] Huntington, John. Show Networks and Control Systems: Second Edition. Zircon Designs Press, 2017.

[10] "Light System Manager Gen5." https://www.colorkinetics.com/b-dam/color-kinetics/products/light-system-composer/downloads/lsm_userguide.pdf

the LSE unit to debug in our AirBNB through the night, disconnecting it from the Tower. The system worked completely for the performance, but it was a very close call.

For the music itself, Tod took all the material and wove it together into a piece which was then orchestrated for symphony and electronics. The symphony would sometimes mimic submitted field recordings, other times playing texturally, and other times playing virtuosically— or most commonly, some combination of all three. The electronics consisted recorded of sounds from all over the city and synth textures created by Tod in his studio.

The original PA system in the room was mono and mounted very high up in the room, which had several balconies. My challenge was to try to find a way to manage the playback of the electronics with two goals: to keep the electronics feeling somehow connected to the orchestra despite the speakers being very far away and up high, and to keep the balance of the sounds from the orchestra and the electronics from covering one another or competing for volume and detail. I was not well versed in the mechanics of speaker design at the time, but there were a few qualities of sound systems that I felt would be good to help with these goals:

1. With *Powers*, I had gotten a good sense of the ***throw*** of a speaker. This was the feeling of having a speaker close to your ears when it in fact might be located across a room. I had had the chance to sit close to high-end studio monitors in a perfect triangle and became fascinated by the fact that certain speakers were so capable of beaming the sound right into one's head. On *Powers*, we had several Duran Audio T2115 speakers, and they initially came with 60 degree horns installed. I always felt these speakers were the gold standard for good ***throw***. Even hundreds of feet away in a theater, they felt close and present, like studio monitors. This meant the high frequency pattern of the horn would fall off more than 30 degrees from the center of the driver. Half way through the *Powers* tour, we switched the horns to 100 degrees, just by physically swapping the plastic waveguide mounted on the tweeter's compression driver. I noticed the speakers no longer had the same ***throw***, i.e. they did not have that "in-your-face" feel of studio monitors despite being far away. I later learned from John Taylor at d&b audiotechnik[11] that what I was perceiving was a ratio of direct to diffuse sound.[12] Speakers with narrower dispersion patterns sound "closer" because one hears less reflected, diffuse sound coming from the room in comparison to the sound coming from the speaker. When it came to field recordings of the city, a speaker with narrow dispersion was capable of getting more detail to further seats without that detail being muddled by the reflections in the room.

[11] "About d&b Audiotechnik." About d&b | d&b Audiotechnik, 10 Jan. 2020, www.dbaudio.com/global/en/about-db/.

[12] Davis, Gary, and Ralph Jones. Sound Reinforcement Handbook: Written for Yamaha. Hal Leonard Corporation, 1989.

2. Another important concept was ***time alignment*** of multiple loudspeakers. I found when installing the audio system on *Powers* that we could change the delay by a millisecond and have the perceived audio image shift dramatically. This is because of the precedenvce effect— audio close enough together in time appears to be the same sound, even coming from multiple different locations.[13] The perceived location of the audio is primarily based on whichever source arrives first at one's ears, but it can be shifted between multiple locations by carefully controlling delay. I typically used solo vocal recordings to explore this. There is a very clear phenomenon: out of time, such recordings sound pinned to various speakers around the hall. When perfectly aligned, the voice will tend to float in mid-air in the middle of the stage. The image is pulled up by the speakers above and down by the speakers at stage level. Of course, this also depends on the relative distance of the listener from the speakers. In the balcony seats, there is a completely different time relationship between upper and lower speakers compared with orchestra level seats. It's the job of the sound designer to navigate this timing relationship across the entire venue such that the sonic image is "believable" and coherent. For the symphonies I would typically put speakers in and around the orchestra and time align hanging speakers so their sound arrived after the speakers in the orchestra. This would give the apparent sense of electronic sounds coming from within the orchestra. Once the time alignment was in place, I could increase the volume of the hanging speakers to allow them to provide the horsepower (generally the rigged speakers were much larger) while using the speakers on stage for localization. This presented a problem if there needed to be presenters speaking in the performance. Having speakers behind a microphone is difficult for feedback. In other scenarios, the orchestra would complain about hearing the electronics too much. In both of these cases, we could add another set of speakers downstage in front of the orchestra. The sources to these speakers would use a delay matrix, with live sources delayed to their location on stage (approximately 1ms per foot upstage), and electronic sources delayed to the location of the rear speakers. Many mixing desks are not set up to allow multiple differing delays to different output speakers.

[13] McCarthy, Bob. Sound Systems - Design and Optimization: Modern Techniques and Tools for Sound System Design and Alignment. Focal Press, 2016. (pg. 161)

3. I also had a sense that dynamic range across a wide frequency spectrum would be incredibly important for helping the electronics and orchestra to blend well. An acoustic orchestra is able to be incredibly quiet and incredibly loud at very high and very low pitches. When mixing electronics with the orchestra, it was important to be able to match or slightly exceed the dynamic of the orchestra with a full range of frequencies. On *Powers,* this meant that we needed 8 subwoofers. They didn't necessarily get loud, but we needed to feel very low frequency energy in a very large room, and this meant moving a lot of air. It becomes immediately apparently listening to an electronic sound crescendo if the system does not get uniformly louder over all frequencies. Without balanced spectral response, the sound does not feel "real" as it gets louder, but trapped in a small box (the speaker).

4. The final important quality of an audio system was its ability to render spatial resolution effectively. There's an amazing phenomenon with an acoustic orchestra where it's possible to pick out a single player and mentally focus on their line, bringing it out from the ensemble. I learned later from John Taylor that this effect is often called "binaural unmasking"[14] and it works because each instrument in an acoustic orchestra has its own physical placement and this implies slightly different delay and volume relationships when compared with every other instrument in the ensemble. Our brains can untangle each separate source in separate locations, but when all the sources are played through a single loudspeaker or even panned in stereo on two loudspeakers, this effect can fall apart very quickly. For this reason, using multiple speakers and separating sounds to different speakers can give more "resolution" to allow more sounds to exist simultaneously— realistically this only works for electronic sounds. Reinforcing an orchestra naively with microphones and loudspeakers takes away a huge amount from the sonic experience of hearing an orchestra live and completely acoustic.

Given that the majority of the PA system for *A Toronto Symphony* was mono, we made the choice to keep the orchestra completely acoustic, while reinforcing only the electronics, and focus on optimizing the sound system for electronics. This meant adding some subwoofers, speakers in the orchestra and time aligning the systems to make it feel as though the electronics, which were only stereo on the first level, felt a part of the orchestra.

[14] Flanagan, J. L., and B. J. Watson. "Binaural Unmasking of Complex Signals." The Journal of the Acoustical Society of America, vol. 40, no. 2, 1966, pp. 456–468., doi:10.1121/1.1910096.

The project was highly successful partially because the performance went well, and of course because Tod's music was powerful and effectively balanced and showcased everyone's contributions, but also because the piece brought together many communities who then came to hear the final piece together. The simple ability to actually hear all the contributions all woven together from so many people and communities gave those involved a sense of ownership and connection to each other and the piece itself. As a concept it was powerful, and Tod began to receive more requests for commissions for other cities.

3.7.1 City Symphonies Around the World

Our workflow for the next few symphonies remained largely the same: For *Festival City*[15] in Edinburgh, we added a stereo hanging PA system consisting of d&b V series line arrays. Having a stereo system for all levels of the Usher Hall meant that we could rely on the spatial resolution of the stereo image in Tod's triggers to give a sense of width. I time aligned the V arrays to a pair of Q7 speakers on stage. These were relatively narrow speakers with good throw, and provided detail and localization on stage. For *Between the Desert and the Deep Blue Sea*[16] with the Western Australian Symphony Orchestra in Perth, I switched to several Q7 point source speakers on each level. These could be separately time aligned to the speakers down stage. We also took the balcony delay speakers and flipped them to provide surround ambience.

In Lucerne, we used hanging d&b V arrays once again, similar to Edinburgh. The Hall at the KKL in Lucerne also has a large echo chamber surrounding the entire space— large doors can be opened which increase the volume of the hall by one third. There are a number of configurable curtains and hard surfaces inside the chamber. We added effect speakers in the echo chamber as well and configured certain triggers to be sent to separate outputs. Lucerne was the first time the trigger samples were broken up and routed differently depending on their content.

[15] "Edinburgh." CITY SYMPHONIES, citysymphonies.media.mit.edu/edinburgh.html.
[16] "Perth." CITY SYMPHONIES, citysymphonies.media.mit.edu/perth.html.

Fig. 3-29: Youthville Students set up on stage at the Detroit Symphony Hall with laptops and keyboards

In Lucerne, Tod, for another first, decided that some of the communities who had helped make the piece should perform live along with the Symphony as well. There was a children's choir in the choir loft behind the orchestra and a *Fasnacht*[17] brass band who burst into the hall at the right moment. I had to reinforce the voices to help with intelligibility, because their words were specifically written for the piece and did not carry well over the orchestra. It was the first time we had live performers needing microphones as part of a city symphony. Luckily as the microphones and performers were behind the orchestra, the time alignment was still simple to achieve for electronics and live microphones without a delay matrix.

3.7.2 *Symphony in D*

Tod's *Symphony in D*[18] in Detroit became more elaborate, as he decided to expand the involvement of guests performing with the symphony. The hall had never had a large PA system installed either, so we had to investigate the capability of the rigging system to handle the weight of a large number of speakers. The guests for Detroit added a unique challenge. Many were traditional acoustic performances, such as Jeffe, who was an african drummer, and a Iraqi Caldian Choir. We also had amplified instruments: Jonathon Muir-Cotton playing motown style bass and Bryan Pope playing indie electric guitar. Then we had electronic instruments with a group of middle schoolers using FruityLoops[19] to make beats and play samples, and Adult, a duo electronic group playing only analog synthesizers. Finally, we had a group of senior citizens describing sounds that had disappeared, 3rd graders describing their hopes and fears for the future, and two incredible poets, Tonya Matthews and Marsha "Music" Battle Philpot, who dramatically read poems set to music played by the orchestra. For this piece, we had close to 40 microphones on stage along with the orchestra. The PA system needed considerably more horsepower as well.

We used a new technology provided by d&b called Array Processing, which was designed to optimize DSP for each element in the line array to provide consistent coverage and SPL (volume) across the entire venue. Array Processing allows one to configure how the line array behaves as distance increases. Typically laws of physics dictate that SPL must change in level according to the inverse square of the distance. This means that a doubling of distance results in 6dB less SPL. A line array typically enables one to beat this law. We can understand this intuitively by understanding that more speakers cover the farther distances from the array, while being very close to the array the listener is out of the dispersion pattern of all but a few speakers. A typical line array allows one to achieve 3dB loss per doubling of distance. ArrayProcessing[20] allows one to specifically create a near, medium and far zone in the hall where the fall off can be precisely specified to be any value. It does this by shading SPL across the system. It is possible to use ArrayProcessing to increase the SPL per doubling of distance, or to have it remain completely constant, or to have it fall off at any amount. This is a powerful sound

Fig. 3-30: *Symphony in D* audio system rigged at the Detroit Symphony Hall.

[17] Kinoshita, J. "Tod Machover's Fasnacht Adventures." Opera of the Future, 18 Feb. 2015, operaofthefuture.com/2015/02/18/tod-machovers-fasnacht-adventures/.
[18] "Detroit." CITY SYMPHONIES, citysymphonies.media.mit.edu/detroit.html.
[19] FL Studio, www.image-line.com/flstudio/.

[20] "ArrayProcessing." d&b Audiotechnik, 10 Jan. 2020, www.dbaudio.com/global/en/solutions/enabling-technologies/arrayprocessing/.

design tool for matching the dynamic of an acoustic ensemble. We're able to have a very large audio system, capable of extreme dynamics and giving us throw and dispersion exactly characterized to the hall, which is impossible for a point source speaker or set of speakers. Yet, the system can have the same fall off as a typical speaker and as any acoustic source. The system was also optimized for frequency response to maintain similar "voicing" (i.e. balance of all parts of the frequency spectrum) across all types of seats. The closest seats were 25 feet from the speakers and the farthest seats were over 150 feet, but we were able to have a consistent sound and SPL in both places. The last element of the system was necessary for the electronic synthesizers. We had subwoofers rigged in the air with V series Array-Processed speakers, but we also added a pair of very large J-Infra triple 21 inch subwoofers. These subs were capable of reproducing very low frequencies (down to 27Hz) in the entire concert hall. These frequencies are commonly found in electronic dance music— to support Adult and our middle school DJs we added these speakers at floor level.

Fig. 3-31: Youthville Students set up on stage at the Detroit Symphony Hall with laptops and keyboards

The process of creating the piece involved travelling to visit all of the many guests in their communities and environments and getting to know them and their musical proclivities so that they could be woven into the piece and the live performance. Tod did much of this, but I was lucky to attend on a few occasions. I spent a few days in an afterschool program called YouthVille helping middle schoolers, who had signed up to learn DJ software, to make sounds for the symphony. During these visits Tod and I tried to help the students think about composition beyond the basic templates and sounds they worked with. To do this, we showed them Hyperscore[21] which was a departure from the grid-based DJ software they were used to. It was very difficult to communicate the importance of critical listening and making music that might not sound like a radio hit. After some discussion about what would work best, the students decided to collect their favorite sounds that had been recorded from around the city and assign them to the keys on the keyboard. For a section of the symphony, they would take a solo using those sounds. We set up laptops and keyboards on stage for each of them to use.

Because of the scope of the project, compared to previous city symphonies, and because this was the first project completed in the US. Dennis Scholl of the Knight Foundation produced a documentary[22] following the development of the piece and the story of the collaborators. The final piece[23] was recorded by the Symphony's webcast team and released to the public as well.

[21] About Hyperscore, www.hyperscore.com/harmony_line/about_hyperscore.php.
[22] "SYMPHONY IN D." APTonline.org, aptonline.org/catalog/SYMPHONY-IN-D.
[23] "TOD MACHOVER Symphony In D." Open Music Library, openmusiclibrary.org/videos/5720/

Of all the pieces, this one brought together the widest group of communities. Some had never seen an orchestra or been in a symphony hall. At the same time, it was also a challenge for the orchestra to learn to play along with analog drum synthesizers. Everyone was changed by the collaboration and it was just a wild journey to produce the performance. Even the act of getting guests on and off stage became a complex choreography. On the mixing desk, I had to keep track not only of exactly which guests were in what part of the piece, but also how to tie together many different sounds and create a cohesive narrative and musical arc. That involved a lot of 'active' mixing, where my deliberate actions would help to shape and balance the different parts of the piece. I had to memorize a performance of my own, with specific fades at specific times to help the piece have musical coherence and an overall shape. For example, the climax for the finale was as much a function of the mix as it was the musicians playing on stage. This was really a truly special experience because I felt as if I were contributing musically to the piece as well.

3.7.3 *Philadelphia Voices*

Our most recent city symphony, *Philadelphia Voices*,[24] commissioned by the Philadelphia Orchestra and conducted by Yannick Nézet-Séguin also had performing guests present. Since Philadelphia has a very rich vocal tradition, Tod decided to use choirs from all over the city. The piece is a significant choral work with many interwoven vocal parts. In total, we ended up with 6 different choirs of different ages and backgrounds all performing together in the piece. We also had more collected and submitted sounds than we'd ever received before. Tod built the textures for the piece with many, many layers of sounds overlapped. While I had become pretty confident with the traditional system design, for this piece there was a unique challenge in that all the voices would need amplification, along with the electronics. A typical stereo system or even a few effect speakers were not going to provide the necessary spatial resolution to keep everything from stacking up sonically and competing for detail and attention.

[24] "Philadelphia." CITY SYMPHONIES, citysymphonies.media.mit.edu/philadelphia.html.

Several years back, having become close with d&b around the time I started working with Jacob Collier, John Taylor and Ralf Zuleeg kindly invited me to visit their R&D lab and factory in Backnang, Germany. While there, we discussed concepts of binaural unmasking, and they showed me some listening demos of a new system. I showed them some of the DSP and workflows we'd created for *Powers*, and some of the more modern Ambisonic technology, including an experimental 7th Order Ambisonic setup made for me by Richard Furse. We discussed what was important about such systems. By the time Philadelphia Voices arrived, d&b had finished development of a product called SoundScape,[25] which was intended to allow object based positioning

Fig. 3-32: *Philadelphia Voices* audio system rigged in Carnegie Hall

of all inputs to the mixing desk across a large number of speakers. The system required at least 5 separate speakers or line arrays across the front of the stage, and would then position sources based on delay and volume to each of the 5 arrays. The result was that each source had more space to exist, and the system allowed a certain degree of binaural unmasking even on electronic sources or with live microphones. The system worked by taking a 3D model of the performance hall and optimizing the spatialization to work based on the actual distances of the performers from each speaker in the system.

We thought that with so many layers and textures and live voices, Philadelphia Voices would be the perfect chance to put SoundScape through its paces. Working with Shawn Duncan, Matt Bell, and Dave Harris of Specialized Audio Video Inc (SAVI), we designed a system comprised of 40 d&b Y speakers in 5 arrays of eight speakers rigged across the stage. The piece was set to be performed both in Philadelphia at Verizon Hall and at Carnegie Hall in New York. The system had to work well, with the same equipment in both rooms which were wildly different shapes and sizes. Verizon Hall was quite large and cello shaped. It was dark sounding and we needed to help get detail to every seat in the house purely as a result of the seating being so far from the PA system. Carnegie was a much brighter room and more live with a smaller stage. There, we had to watch our high-mid frequencies to keep them from getting out of control and covering the detail in the high end. We devised a rigging and cabling plan for both rooms.

To allow the electronics to be spread out, we bounced out each section of the piece broken down into several layers. This was made possible because I had recently redesigned Tod's home studio, centered around a Digico S21 mixing desk. We came up with a new workflow to export triggers that printed both a stereo mix exactly the way Tod heard it in his studio, and a stems divided so that every individual component of the mix was separated but still at the same levels as Tod's mix. Combining the separated version of the trigger with all outputs at unity gain would result in exactly the stereo mix, but it allowed the elements to be spaced across the 5 arrays rather than limited to stereo. This workflow quickly became the standard method for creating triggers in Tod's studio following the piece. It allowed us to start with a mix we knew, but left flexibility for fine tuning certain parts (sometimes voices in the triggers needed to be louder or quieter) and spreading the sounds out in the room. All together, there were 32 separate layers of exported triggers that were positioned individually.

Fig. 3-33: *Schoenberg in Hollywood* production photo. Credit: Liza Voll Photography

We did the same thing with 16 choir microphones, positioning them virtually in the sound system relative to their location on stage. This was important because there were moments in the piece where soloists from the choir would speak or sing alone. On a typical stereo system, it would be very difficult to tell the difference between recorded samples of voices and live voices, especially telling that one person was speaking in a crowd of 350 rather than a simple recording. The localization of the choir microphones across the five arrays made it much easier to tell not only where the soloists were, but also to tell that they were live and to immediately give the audience the ability to pick them out of the crowd. We were quite amazed at this quality of the system, and it represented a significant advance our ability to binaurally unmask the live sources even though they were reproduced live. Immediately I thought about using the system to reinforce instruments to see if we could now finally have a reinforced experience more like an acoustic experience, with instruments as well as voice.

3.8 *Schoenberg in Hollywood*

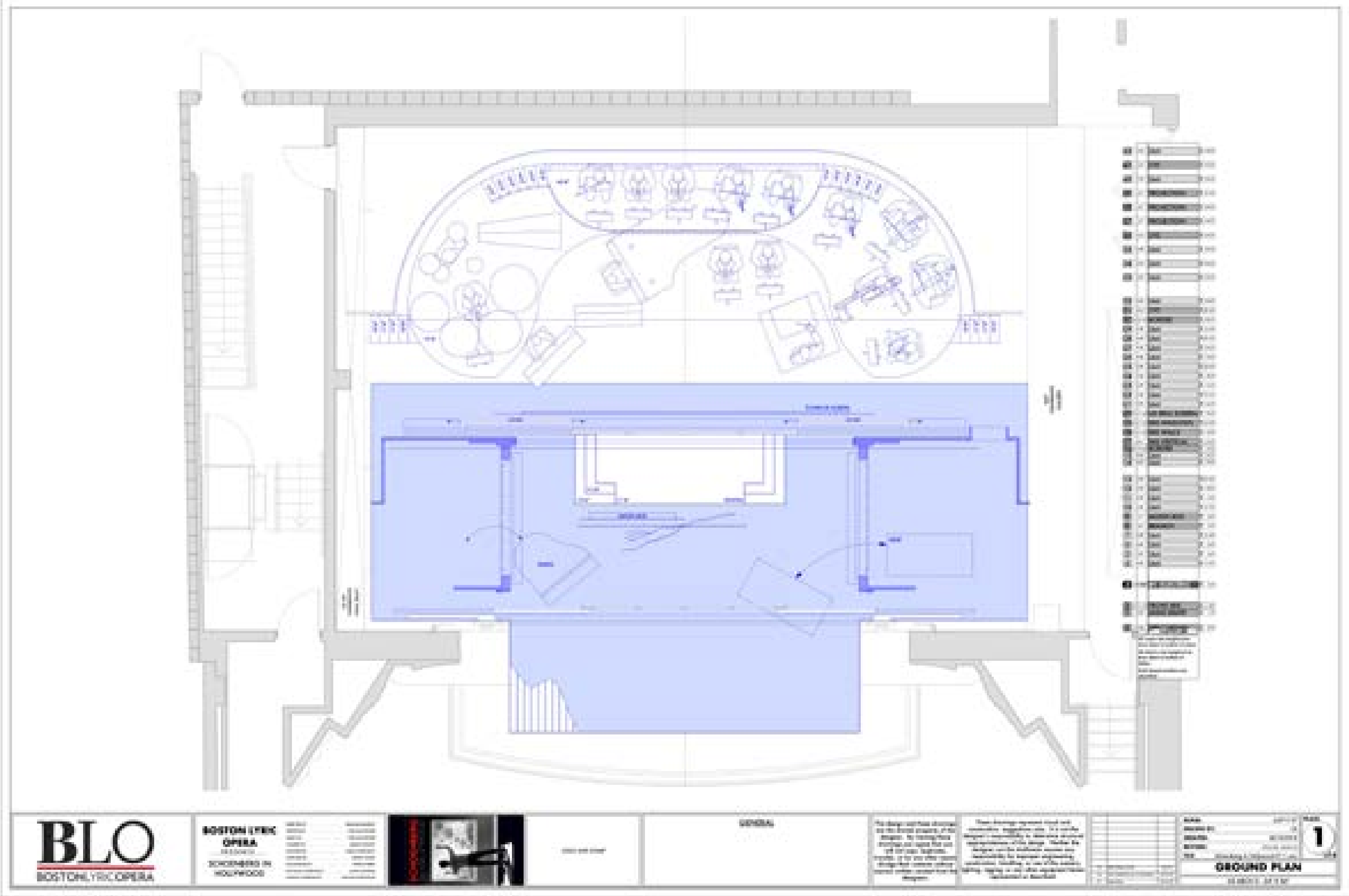
Schoenberg in Hollywood[26] was Tod's first major opera work following *Death and the Powers.* The story of the piece was based around an early interaction that Schoenberg had with the head of the MGM film studios, Irving Thalberg, upon arriving in the US after fleeing Nazi Germany. Thalberg asked Schoenberg to score the film The Good Earth, and Schoenberg requested a very high fee and complete control of the sound and music of the film, including dialog. He was subsequently outraged to learn that film scores were often edited and changed to match the image. Although he was turned down by Thalberg, the opera explores his music in connection with many film genres of the time. Vignettes from many of his pieces and moments of his life are included in the libretto. The piece explores the question of creating artistic work for mass consumption versus following one's own vision to ultimate completion. Schoenberg's music was quite unpopular with both critics and the public, which is what made him such an odd choice to score a Hollywood film. This dichotomy is central to the aesthetic of the piece.

Where the *Powers* story was centered around the role of technology, this piece was about Schoenberg's vision and music. We aimed to keep the production infrastructure simple, while encompassing many styles of film and representations of his work. We were able to use some of the systems and workflows we had developed for *Powers* and the City Symphonies, and refine both to keep our process relatively constrained and streamlined.

[26] "Schoenberg in Hollywood." Schoenberg in Hollywood, schoenberg.media.mit.edu/.

The production was premiered at the Paramount Theater in Boston, which had quite a small and deep orchestra pit. The creative team made the choice to place the orchestra at the back of the stage, where there was much more room. Several projector screens and set elements were positioned in front of the orchestra on a pulley system. The challenge with this physical arrangement was that many of the projection surfaces were solid. This meant that for much of the performance the orchestra would be blocked audibly and visually.

Fig. 3-34: *Schoenberg in Hollywood* Orchestra plan



3.8.1 Orchestra as Character

Because of this, I hoped to test some of the more advanced imaging capabilities of the d&b soundscape platform that we used for *Philadelphia Voices*. This was something I would typically not try because we had previously had significant trouble reinforcing acoustic instruments, but the advances in speaker and imaging technologies and the performance of our sound system for *Philadelphia Voices* encouraged me to revisit the idea. The concept of recreating the sound of the orchestra almost completely with speakers allowed us to consider additional manipulation of the sound. Since the production's technology was otherwise limited in scope, we considered this an appropriate opportunity to experiment with the mechanics of the reproduction and reinforcement. In this vein, Tod wished to treat the sound of the orchestra and electronics as a character, moving and changing throughout the piece with its own blocking. We hoped that reinforcement of the instruments through speakers exclusively, the mixing of the electronics and instruments, and the manipulation of placement and apparent size of the orchestra, would support the many varieties of film style and the narrative quite well.

To achieve this, we needed to create a system to support the choreography of all the sound elements in the piece. Traditionally, productions use cues built into the mixing console to modify processing of the live sound. The operator reads a score with notations for fader positions and mix notes, and triggers cues at the right moment. Since we already had triggers notated in the score, we hoped to connect the choreography elements to these triggers rather than adding another set of responsibilities to the mixing desk operator. We also needed a way to design, rehearse and modify the choreographies easily. Both the Soundscape processor and Digico mixing desk had published Open Sound Control (OSC) specifications that allowed for control by external devices. The Digico had rudimentary OSC control abilities, but fine-grained cueing and show control abilities. The soundscape processor had relatively advanced OSC control but very basic snapshotting and no cueing. It also had a good interface for controlling the movement of many sounds in groups, which the Digico lacked.

Given this, we wished to be able to edit sound locations using the native soundscape interface, and to store and manage cues with the flexibility of the mixing desk and recall the cues based on triggers from the orchestra. The mixing desk was able to save cues with OSC information but the data had to be input using three knobs for each source. We had 64 sources and setting their location with knobs would have been much more time consuming than clicking and dragging objects on the screen. What was needed was a system that would listen to location data from Soundscape continuously and allow the storage and recall of snapshots based on MIDI. With Ethan Nevidomsky and Nikhil Singh, we created a Hyperproduction node and patch to listen to positional information from soundscape, store the coordinates when needed using the Hyperproduction's preset module, and then allow interpolated transitions between presets. The soundscape processor would continuously send and receive data so the Hyperproduction node had to be designed as if the protocol were bipolar, where it would stop listening when sending and vice versa. We spent some time tweaking timeouts to make it possible to recall a preset, then quickly edit it and store the data back to that preset without having to wait for the Hyperproduction system to start sending data again.

The inputs to the soundscape processor were multichannel layers of the triggers exported in Tod's studio with the same workflow that was used for Philadelphia voices. The combination of the MIDI triggered preset recall from Hyperproduction to Soundscape resulted in flexible positioning of the triggers throughout the piece with an easy workflow for editing and refining positioning for each layer of the triggers.

The orchestra musicians were panned in relation to their actual locations on stage. We ultimately decided to do this to keep the majority of the piece sounding naturally acoustic. The soundscape processor did a fantastic job of reproducing the orchestra in this way. This acoustic was combined with reverbs built into the processor and additional quad reverbs from Altiverb to allow us to change the acoustic nature of the space. There were certain elements of the production where instruments were moved around for solos, such as the cello solo in the section of the piece "Man's First Song." The keyboard sounds were also moved around between scenes. Rather than carefully document the location of the sounds and reproduce them, we set aside some time in the hall and listened to the electronics in the space, setting each position according to a few properties:

1. The spatial size— should the music feel very large in the space or very small?
2. The imaging and width of the various layers of acoustic and electronic sounds— should the sounds separated and wide or floating together?
3. The location— where should the music be? Should it be completely enveloping? Positioned on stage, off stage in the distance, or in and around the audience.

We also had a recording system set up to use as a virtual sound check, allowing us to work on the orchestra reinforcement even when they were not present. For the orchestra we kept the arrangement relatively consistent and changed the size and imaging by varying the amount and type of reverb. We used Soundscape for a basic short reverberance, taking into account the dimensions of the room. We used Altiverb for effects reverbs to grow the size of the space significantly. This was routed to faders on the desk and mixed throughout the performance according to the part of the piece and the live performance of the singers and the ensemble.

Riding reverb is a huge part of the cohesiveness of the mix. Especially when close micing singers and instruments, it's important to have the tail of the reverb feel natural given the volume of the space. Often this means adding a bit more reverb as the performance grows in intensity, which is a good way of simulating a room that reacts a bit more as more energy is added. This is the case in the finale of the piece. There are also moments when a very quiet performance might have a lot of reverb, for example during the lyrical sections of *Verklarte Nacht* in the scene titled "Friendship".

3.8.2 Sound Design

The audio system used to support the piece was based around point source speakers, since the hall was small enough to be served by these rather than line arrays. Shawn Duncan at SAVI provided help with the design and installation of the system and together we specified 10 V series speakers, half V7P and half V10P on a main truss across the top of the proscenium. The choice of these speakers was also informed by the weight limitations on the truss and lighting instruments placed along with the speakers. The wide dispersion speakers were pointed at the lower levels of the audience, just reaching the back of the orchestra level seating under the balcony, but with coverage to hit the closest seats as well. The upper speakers had narrow dispersion to avoid reflections on the architecture, pointed at the balcony which was much further away.

A selection of speakers, Q10 and E8, were located closer to stage level to allow us to push the image downward and allow fill for the front rows of seating. A set of four V 18 inch cardioid subwoofers were located to the left and right of the stage, two per side. These provided music support going down to about 40Hz. A second set of J Infra triple 21 inch subwoofers was located under the orchestra pit for effects. A part of the piece simulated a World War I bombing and we needed the extra low frequency coverage down to 27Hz for these effects to be realistic.

A set of E and Y series speakers were used for surround sound. There were minimal patch points in the house, so we located an amplifier at the rear of the lower level and had 3 rear channels and 2 side channels for the orchestra seating. There were 4 rear channels and two side channels for the balcony. We were not able to rely on this number of speakers for detailed surround textures as in *Powers*, but it gave room to control the sense of

space in the hall and spread out orchestra and electronic elements subtly. This went a long way to help the music feel less confined to the stage and less exclusively frontal.

Fig. 3-35: *Schoenberg in Hollywood* mixing desk and computer configuration

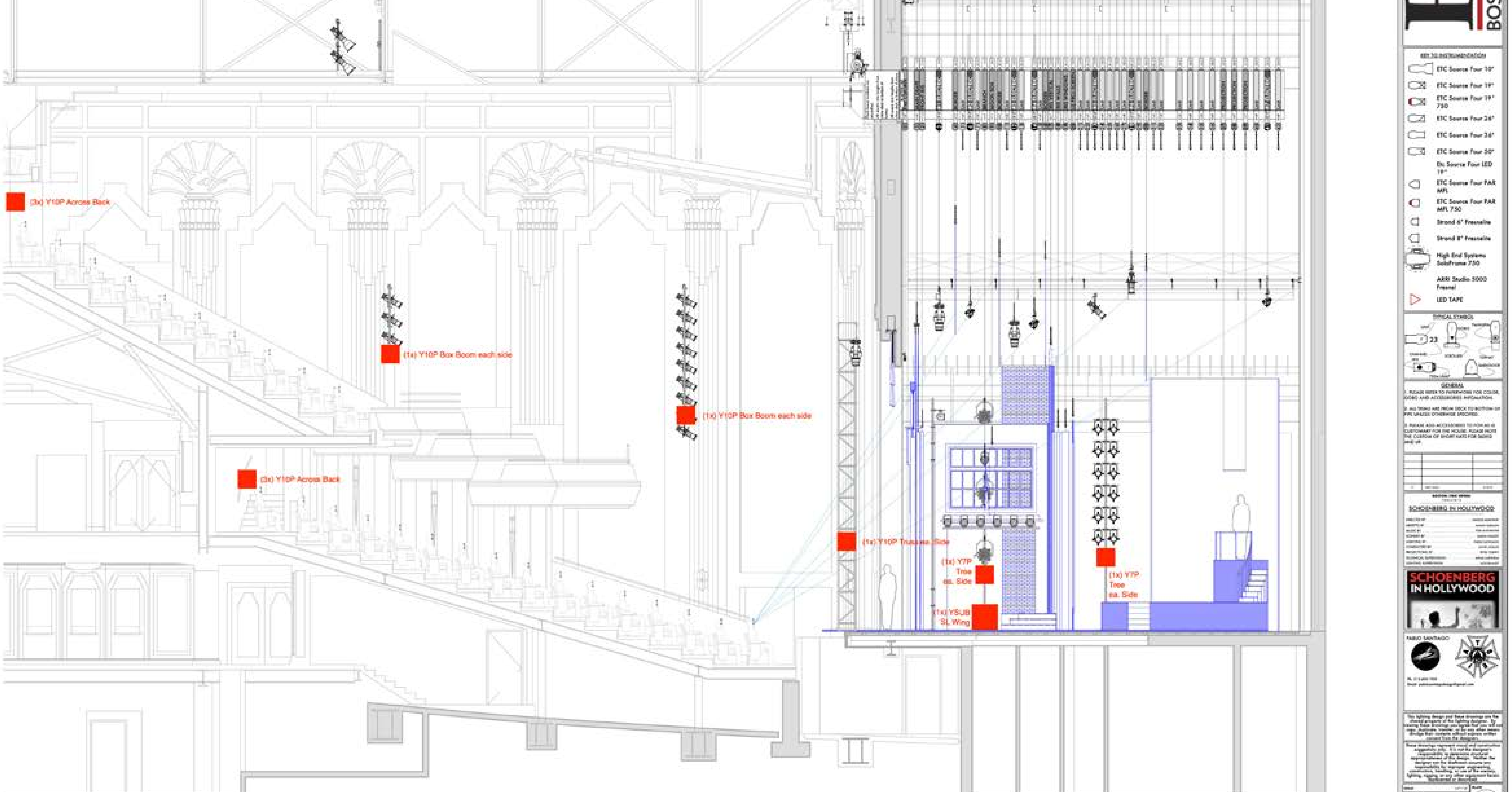


Fig. 3-36: *Schoenberg in Hollywood* surround sound peaker locations

A final set of 6 speakers were used on stage, Y series, for monitoring for the cast. These were mounted left and right at three points in the wings. The front pairs of speakers were primarily for the singers, and used so they could hear the orchestra even when the set was blocking them. The rear pair served as monitoring for the orchestra and was supplemented with small power speakers with local volume controls for the keyboard players and the conductor. The conductor had two of these small monitors, one with exclusively electronics and one with exclusively vocals. He would receive a mix of these from the mixing desk but could vary his own balance as needed throughout the piece.

The system was powered by d&b amplifiers fed via AES/EBU with DS10 Dante bridges. The bridges were fed from the Soundscape processor. Our mixing desk was a Digico SD10 fed via an SD Rack on an Optocore loop. An Auvitran AudioBox with Dante and MADI cards converted the SD10 MADI outputs to Dante for connection to the Soundscape processor. Two more computers were used for a redundant record systems via a MOTU 112D splitting the desk's second MADI output via Thunderbolt and USB 2.0. The returns for the MOTU 112D were used for virtual soundcheck via the SD10's second MADI input.

Three laptops connected via Dante supplied effects, keyboard sounds and multi channel triggers. These roughly mirrored the systems for *Skellig*, but with all keyboard sounds provided by Omnisphere and Keyscape plugins rather than hardware synths. We used MainStage to automate patch changing. It was able to successfully run 23 instances of Omnisphere which became our benchmark for performance of a keyboard automation system. Nikhil Singh tested Digital Performer, Reaper and MainStage and we found MainStage to be the most reliable.

Shawn Duncan and I conducted several listening tests with the soundscape processor and a Smaart measurement system to try and understand what the delay and volume relationship was with the system operating in different modes. We discovered there were different behaviors between sources placed inside a set of speakers and beyond them, for both front and surround systems. In general, we preferred front sources to be located behind the speakers and side sources to be located within the audience listening area in front of the speakers. Particular care had to be taken when moving a source from the front system to the surround system. In this case, the trajectory mattered and a path that moved in front of the PA speakers could cause an abrupt change in volume if the source was not in the correct placement mode.

We also spent some time measuring the time relationship between virtual sources to understand how fully time-aligned sources in the system would compare to acoustic sources on stage. For this we used a non-Soundscape output to feed a single V7P speaker on stage. We measured this signal compared to a virtual source in Soundscape which we panned upstage and downstage of the physical speaker location. We discovered that it was quite important to keep virtual sources confined roughly to their location or slightly upstage of that location in order for the proper timing relationship to hold. The system was quite sensitive to timing. For this piece, we were actually lucky that the stage playing area was relatively small, otherwise it would have been very important to carefully track the performer's movement and send this to Soundscape.

3.8.3 Evaluation

Schoenberg in Hollywood is a particularly interesting project because it represents a natural conclusion (to date) of many of the technologies that I began to experiment with originally when I started in the Opera of the Future group. Each item below traces the development and evolution of technologies from my first use of them to their current form on this project:

- After testing and experimenting with ambisonics and wave field synthesis in various forms, including developing our own DSP in hardware and software, the best ideas were distilled into a commercial product, SoundScape, which we were able to develop custom control for and was used in *Schoenberg in Hollywood* for panning both electronics and acoustic sources.
- Hyperproduction, having grown out of the *Death and the Powers* mapping system and used on *Fensadense* was able to be integrated with SoundScape very quickly and effectively to allow a system of triggers, a concept originating from Tod's early performances, to control choreography of technical elements and spatialization, as well as content.
- The triggers themselves had evolved from simple stereo files, to multi channel ambisonic-encoded samples with fixed choreography, to object based multi-layer exports that could be choreographed completely flexibly during the rehearsal process in the live space to maximize the ability to tailor the performance to the space.

- The type and quality of sound reinforcement transitioned from simple electronics-only stereo systems:
    - ◊ to multichannel electronics with stereo orchestral reinforcement in *Death and the Powers*,
    - ◊ to multi-layer spatialized object-based placement of electronics and vocals with acoustic orchestra for *Philadelphia Voices*
    - ◊ to completely object based full orchestral and electronic reinforcement for *Schoenberg in Hollywood*.

The combination of all of these elements together allows the technologies used to be integrated with the production in a way that is more seamless than other types of large-scale productions. Some examples of existing work using one of these strategies:

- Cirque du Soleil[27] uses very complex interlinked production systems, object based panning with surround sound, live music performance and flexible timing based around vamping. However, the systems are still dependent on operators to cue almost all aspects of performance, so there is no way for the elements to flow musically in the way that triggers and modes make possible for the projects documented here. The same is true for Broadway musicals, and this limits the ability of the performers to improvise or even to perform a fixed score with truly musical timing.
- Large stadium pop tours use many types of incredibly complex, synchronized production systems, but their choreography is all based around timecode, so there is no ability to be flexible with timing at all, despite the high level and cost of production.
- Professional DJ acts, touring orchestras and large jam bands may improvise or put on a very musically timed, natural feeling performance with significant audience interaction and flexibility. However, these acts are limited in their ability to choreograph

[27] Huntingon, J., "Tangerine Trees: Cirque du Soleil meets The Beatles in the psychedelic new spectacle called LOVE." Lighting and Sound American, p. 56, September 2008

complex elements to closely follow the structure and nuance of the performance on stage.

In contrast, *Schoenberg in Hollywood* integrates elements that would typically be triggered via technician or time code and attaches these elements directly to the musical performance of the musicians through simple triggering and basic audio analysis. The technology supports the feel of the piece. It was developed in the studio based on an emotional process, and it was rehearsed and choreographed at full scale, with the ability to make changes on the fly. To me this is an example of a result that in reality will seem quite natural and unremarkable to audiences, while in fact it was based on almost 10 years of work and experimentation.

It enables more emotional, human performance through its ability to support complex production choreography and tying of musical and technical elements intimately together. It is quite exciting to be able to take these ideas and begin to apply them with mainstream hardware and refined (not quite mainstream yet) software. The instrument becomes the full environment, and this begs the questions: what other types of intertwined technology/performance production-instruments can be imagined, and how completely can we articulate new and different types of production technologies with sensibilities of musical performance? Furthermore, can we design these live experiences to encompass more than just performance venues with colocated seating? Could they spill into everyday life?

3.9 The Rhythm of Production: One-offs vs. Mainstream Touring

The performances detailed in this chapter for Tod Machover and developed in the Opera of the Future group at the Media Lab have a very specific type of development cycle. Often the performances take months or even years to create. This is due to the large amount of customized technology present and even more critically, the large amount of novel approaches attempted both for the production and the most basic creation necessary to write the pieces (ex. collaborative symphony). Tod does not like to do the same thing twice, so almost every project involves significantly pushing the boundaries of previous projects.

At the same time, it is very rare to perform an Opera of the Future piece more than a few times once it is developed. Often, the final rehearsals and performance may take place over a week or less, and then the piece is never performed again, or if it is, new elements are added or adapted. This approach and cycle of development keeps innovations happening very frequently. Although many techniques are recycled, such as triggers and mapping systems, Tod has never "gotten comfortable" with a workflow or system of creation. Every single project is used as an excuse to advance some concept, technique or technology, and pieces must be developed to have their maximum impact immediately, since there are so few opportunities for performances.

As I became familiar with the cycle of development for Opera of the Future productions, I became intrigued by the mainstream production world. These types of productions were developed much more quickly but then toured for months, and sometimes even years. Because of this, the style of production process was significantly different, with a very large focus on redundancy and simplicity of implementation. It was very rare to find artists taking risks in this world (one notable exception is Imogen Heap with her MI.MU. Glove[28]). Over time I began to wonder if it would be possible to practice some of the innovation techniques and processes from Opera of the Future for productions that followed this very different life cycle. As chance would have it, I serendipitously ended up with an opportunity to do just that for Jacob Collier, and I have documented my findings in the next chapter. The chapter following this investigates an entirely different type of creative process that is more common to mainstream touring artists. I am excited that many of our techniques do, in fact, apply even where the priorities of mainstream production differ. This means that we can begin to bring the special parts of our research to the broader world.

[28] Morgan, Hannah. "MI·MU GLOVES: The world's most advanced wearable musical instrument, for expressive creation, composition and performance." MI·MU, mimugloves.com/.

4 Jacob Collier and the *Djesse* World Tour

I first saw Jacob Collier's work in late 2014 when one of his YouTube videos was reposted by Michael League of Snarky Puppy. Michael was working on rearranging some of Collier's songs for his band's *Family Dinner*[1] project. I became obsessed with the videos and began showing them to my friends and family. Jacob had a sense of playfulness, but also very finely tuned production aesthetics and an unbelievable command of harmony and rhythm. In the videos, he sings and plays hundreds of musical lines simultaneously and I was mesmerized not only by the way they interlocked musically, but also by the way he mixed the sounds and textures, and edited perfectly synchronized snippets of video to go along with the music. On a whim, I sent him a Facebook message to see if he was curious about live performances or music technology.

About two weeks after I sent the message, I received a reply from Jacob asking for some help and advice on an upcoming performance. He had been invited to play at the Montreux Jazz Festival,[2] opening for Herbie Hancock and Chick Corea, and introduced by Quincy Jones. Previously, he had only performed in jazz trios in very small venues, and he was interested in coming up with a way to perform live in front of an audience that could reflect the process and aesthetics of his YouTube videos. I eagerly agreed to have a Skype conversation with him so he could ask some questions he had about performing music with technology.

In our first conversation, we discussed some big ideas surrounding what he wished to do for his performance, and about the best strategies to end up with reliable technology on stage. Jacob had a lot of ideas about ***looping*** and many of his thoughts involved creating new hardware or software, specifically for the performance. Having just completed some stressful and intense productions, I tried my best to convey the scope of effort required, and some of the drawbacks, to designing entirely new technology and using it for live performance. There were about 6 months until the performance and there would not be a lot of opportunities for testing the show beforehand.

Fig. 4-1: The Jacob Collier youtube video that I found in 2014.

Looping: The act of performing by recording over oneself again and again to create layers of musical texture. One of first modern looping performances was KT Tunstall's *Black Horse*

Feeling is more important than any specific technology, idea or mechanism.

1 Fordham, John. "Snarky Puppy: Family Dinner Vol Two Review – Uninhibited, High-Flying Jazz-Fusion." The Guardian, Guardian News and Media, 11 Feb. 2016, www.theguardian.com/music/2016/feb/11/snarky-puppy-family-dinner-vol-two-review-jazz-fusion-manchester.

2 "The Universe of the Festival: Montreux Jazz Festival." The Universe of the Festival | Montreux Jazz Festival, www.montreuxjazzfestival.com/en/universe-festival.

It is incredibly difficult to become comfortable with new technology except under performance conditions.

When building any new software or hardware, it is rare for systems to work perfectly the first time. Typically, a long period of time must be spent finding and fixing bugs. Most bugs do not reveal themselves until the system is used in exactly the way it will be for performance, and it can be difficult to replicate those circumstances without a dress rehearsal. There is also a large difference, in practice, between building new software or hardware from scratch versus using pre-existing elements that are already trusted and assembling them into something new. It is easy to take for granted how complex even the most simple performance components can be; building from scratch opens a minefield of difficult testing and requires taking countless edge cases into consideration. Most importantly, it can take a significant length of time to reach a point where it is possible to even test the viability of the underlying creative idea. One might spend weeks or months in development, just to find that the system does not satisfy the creative needs of the piece.

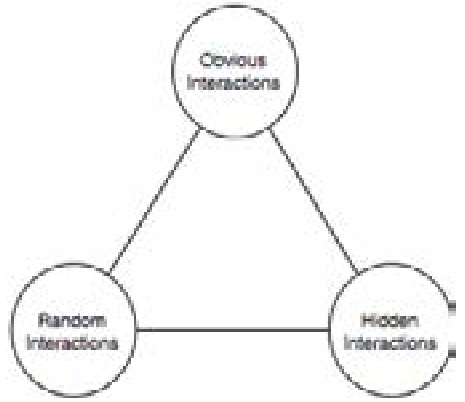


Fig. 4-2: The trinity of bad experiences

Given all of this, I gave Jacob this series of recommendations:

- He should find a set of production elements, sounds, performance tools and effects that he finds compelling and that individually work well on their own, and use those together to make something larger, letting the "feel" guide him more than any specific technology, idea or mechanism. For example, if he enjoys looping, he should watch all his favorite looping performances on YouTube and write down what aspects do and don't appeal to him, and try to understand the reasons why.

Technology should strike a balance between feeling connected to the music and the actions of the performers, while maintaining enough nuance that the relationship between technology and performance isn't immediately obvious.

- Many of the best live performance technologies strike a delicate balance between feeling connected to the music and the actions of the performers, and maintaining enough nuance that the relationship between technology and performance isn't immediately obvious to the audience. For example, if a lot of effort is spent trying to engineer a system to recognize a certain action or musical moment, that effort is wasted if the performer could instead simply push a button or foot pedal, or if a timer could be used to achieve the same result. There isn't a need to make complex systems that are invisible to the audience unless they truly add to the experience, creatively. We have all experienced performances where we watch a DJ or producer in front of a laptop and it is impossible to tell if they have just pushed "play" on a recording or if they have any real time control of what we hear. We have also experienced performances where there is no discernable connection between the physical actions of the performer, the technology and the music (ex. many brain music performances fall into this category). We have also been to performances where the entire set of capabilities and interactions is obvious immediately; the blue button makes sound A, the red button makes sound B— in this situation, the audience loses interest quickly. Meaningful experiences are found in between these extremes, where the audience is kept interested, but there is a clear relationship among the performers, the technology and the music.

- If he created new technology, he should set up "show situations," with a real audience, to put himself and any systems through their paces well ahead of time. Even with extensive development, testing and rehearsal, it is incredibly difficult to become comfortable with new technology except under performance conditions. He might discover, after performing, that the technology needs drastic revision; the earlier this can be understood, the better. It is important to always test in real conditions, and this not just for the technology. For true success, the performer must get to know how it feels to perform *with the technology.* It is unfortunate, but common, to uncover awkward interactions only in these conditions.

It is essential to understand each and every failure mode of a system.

- Anything that can go wrong, will go wrong. It is essential to understand each and every failure mode of a system. Live performance conditions are unforgiving and things often do not go as planned. Any device that is intended to be used in this environment must be hardened against basic stresses and be very simple to operate. Computers will crash at the worst possible moment, cables will fail, and software will behave erratically and break everything. We should design technology to be simple and durable to allow us to focus on performing and creating rather than troubleshooting. To truly test new systems, risky technology, and experimental functionality, the fundamental elements of the hardware and software must not fail in basic ways. Despite this, we should recognize that our process will not allow for perfectly refined designs. We must be efficient and smart about our development so that our systems stand up to the abuses of performance without requiring a massive scale of resources to make them robust.

Following this advice, Jacob decided to focus on two ideas, each with several pre-existing products for support: vocal harmonizers and looping.

4.1 The Beginning of the Harmonizer

As it happened, Jacob was visiting the East Coast of the U.S. to record with Michael League and Snarky Puppy, so we made plans for him to stop by M.I.T. on the way to that session. He brought along a TC Helicon Voice-Live Touch 2 multi effects processor which had a vocoder mode, taking in MIDI pitches and making a virtual choir of all the played MIDI notes with the words and timbre taken from a microphone.

We began by discussing the things we liked about the processor and the things that we didn't: it had low processing latency, so high voices and harmonies sounded natural. However, low voices and harmonies did not sound particularly good. It only had four voices; and its user interface was cumbersome. It did many other things as well, but we weren't particularly impressed with the other features. We tried other harmonizers with similar functionality by Antares, iZotope, and others. With software plugins, the latency was much higher, but the sound quality was better. Antares was by far our favorite sounding plugin, and it especially had great low end. By "great low end," we mean that even when singing high notes into the system, the playing low notes resulted in full, loud bass that hit hard with significant volume at the fundamental frequency of the note and a realistic formant (meaning upper harmonics) given the high qualities of the input voice. In the audio example appendix, I have included examples of the TC Helicon low end, and the harmonizer low end, to hear a comparison of the product on its own and what we became excited about when testing Antares.

In addition, most existing products were limited to a small number of voices, which proved to be the biggest challenge— Jacob wanted to play huge chords with more than four notes. Many of the plugins we tested operated in different modes - some were fixed chords and intervals, others took notes via MIDI on a single channel - but a handful had a mode which used a separate MIDI channel for each voice, and on that channel, the voice would behave like a monophonic synthesizer. Antares was one of these plugins. Working in Reaper, I

found a script by Final Audio[3] to split polyphonic MIDI into monophonic MIDI on each channel. For example, playing a C major triad, the C would be sent on channel 1, E on channel 2 and G on channel 3. Using that script, we were able to stack 4 plugins and get 16 voices by routing channels 5-8, 9-12 and 13-16 to channels 1-4 on each additional plugin. Just that simple setup, with 16 voices allocated to monophonic MIDI channels, was very enjoyable. Jacob spent a good amount of time experimenting with this setup, and we slowly realized that he was more agile with his playing than the instrument could be, since each voice had a long release time in the hundreds of milliseconds. There didn't seem to be an easy way to change that, but we could route each voice separately out the plugin to its own channel strip. In Reaper, tracks can contain both MIDI and audio, so with the audio from the plugin and the MIDI from the channel strip both routed to the same place, we could apply a MIDI-triggered gate to the channel to set a custom attack and release for each voice. At this point, I added some custom EQ and Mid-Side processing to the output to make it so that low voices were mono and higher voices were panned left and right. Jacob also became enamored with the "Glide" feature, which was essentially portamento for the voices. We wanted to find a way to incorporate it into the system, but because of the polyphonic to monophonic transformation, this was difficult to do. The solution was to add another instance of Antares running in its normal mode and only on the last four voices, so the glide would always happen between the most recent notes.

At this point, we had a thrilling demo after only one afternoon of experimentation. Our improved harmonizer ran in Reaper on Jacob's laptop, and we were both impressed with the "feel" and nuance of it, especially combined with his voice. We had something that felt like it could be explored with a seemingly infinite variety of sounds just by using the voice as input. The system did a good job of letting interesting qualities of Jacob's voice become part of the overall sonic texture and he kept discovering new ways of using it— for beatboxing, for whispering, for belting, high singing and low notes or vice versa. We both wanted to keep exploring to see what else it could do, and it seemed to be capable of a vast number of different behaviors that we couldn't have expected or predicted. It was a system made of existing software, nothing particularly special, but with the right small tweaks, it came to life.

[3] "MIDI Chord Splitter - Cockos Incorporated Forums." Cockos Incorporated Forums RSS, forum.cockos.com/showthread.php?t=26559.

Even given the many positive attributes of this setup, the latency was quite high and Jacob was interested to have a device that he could easily use in any performance situation. I have to admit that, by this point, I was quite intrigued and agreed to think about how to take the device further in my free time. In the meantime, Jacob began to think of what it would mean to use this device to recreate some of his YouTube arrangements.

4.2 Live Looping and Mixing

Next, we began to think about what a stage setup might look like. There are many classic looping artists and we had spent a good amount of time watching a lot of their YouTube videos. Bernhoft[4], Imogen Heap[5], KT Tunstall[6] and others had put together some remarkable performances using Boss stomp pedals. We liked the idea of using a pedal to do the looping because it was simple and reliable; there was no computer to crash nor audio interfaces with complicated cabling to connect to the mixing desk.

The challenge was that we were unable to find examples of people making music on these pedals that was anything like Jacob's. Subsequent research revealed that it was extremely complicated to change meters or tempi; it was even difficult to find pedals that could loop one instrument, store that loop and bring it back later in the song after other loops had been recorded, as well. Most looping performances required the audience to wait while the performer built up a lot of lines, and we didn't like that kind of experience. It was too easy for the audience to become bored because of the obvious and predictable nature of the loops— after making one or two loops, the audience could immediately tell what would happen for the next several minutes. Our ideal looping system would be flexible enough to record snippets of anything played, so all the material in the song would be captured naturally as part of the arrangement. We did not want Jacob to play similar lines one after another, or the audience waiting for him to stack up harmonies or bass lines. We hoped avoiding this would give the audience a sense of surprise and wonder by keeping them guessing as to what would happen next. We also wanted the system to be able to change keys, tempi, and meter and have the loops follow along, as well.

[4] Bernhoft - C'mon Talk (Official Video), YouTube, www.youtube.com/watch?v=rxoiZZ8UBEY.
[5] Imogen Heap - Just For Now, YouTube, www.youtube.com/watch?v=25VGdNU3nrU.
[6] KT Tunstall - Black Horse & The Cherry Tree Live, YouTube, www.youtube.com/watch?v=T7oIa0L7j0M.

We ended up selecting Ableton Live, despite the challenges of using a laptop in a performance, and this required a way of interfacing with a PA system, quickly. It is one thing for a system to work well once while experimenting or testing, but the goal for this system was to be able to use it at Montreux with limited setup time and no flexibility to troubleshoot or fix bugs. We needed a completely foolproof system. Another difficulty in making a good computer based looping setup was to maintain the correct gain structure. Overdubs add up small discrepancies so that any minute change in gain, added latency, or noise floor can become massive after 5 or 6 passes. This necessitated a digital interface between the Ableton computer and the mixing desk. Our loops shouldn't be distinguishable from the live performance, so we had to loop exactly what was sent to the PA system.

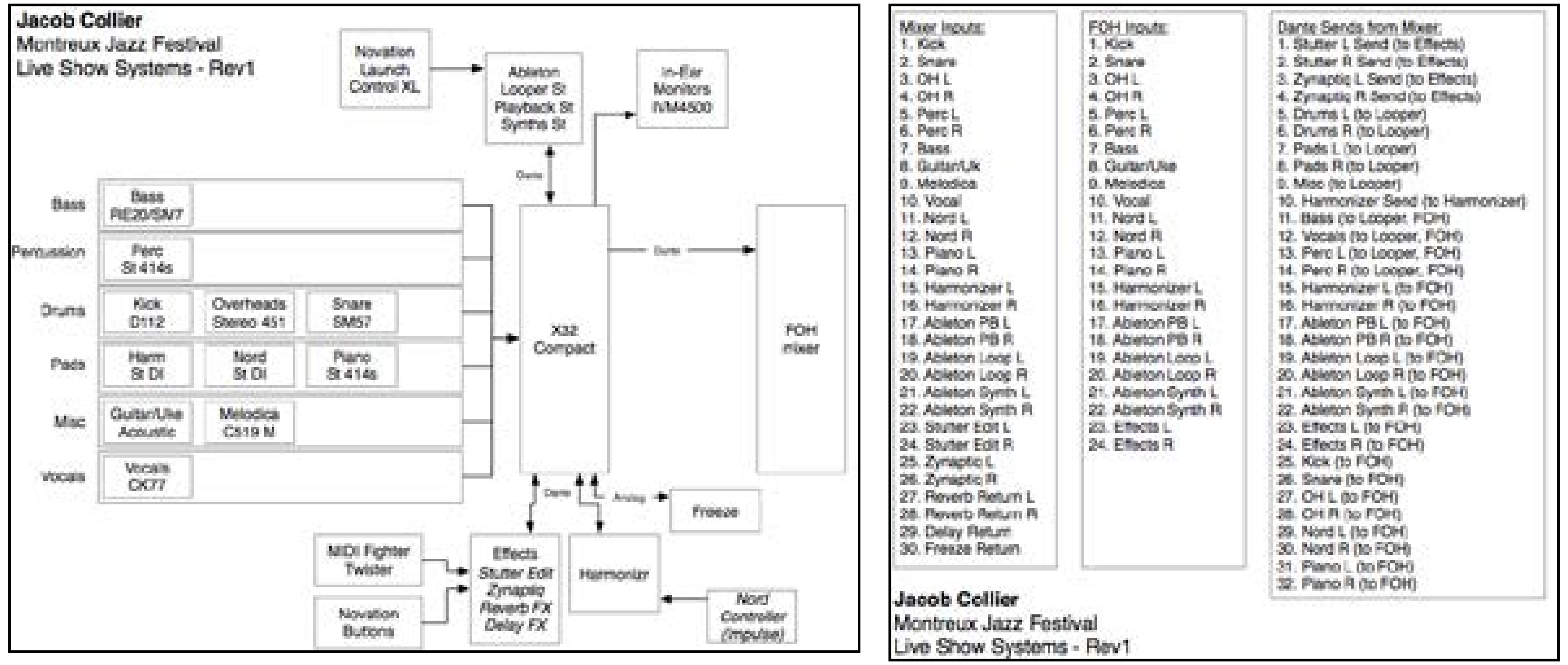


Fig. 4-3: Initial Montreux Jazz System Diagrams

tem. To handle memory situations, we used multiple loopers. Ableton allows an unlimited number of loopers, so it was easy to handle situations where multiple musical phrases needed to be stored simultaneously. Floor monitors would leak into the mics if we were looping, so we made the decision for Jacob to wear headphones. He also needed to run around the stage and wanted to be able to sing from any location. At first, we thought of placing 5 vocal mics on stands, but we also considered that a headset might perform even better.

Fig. 4-4: Initial looping performance tests at MIT

Jacob wanted to play a large number of instruments, so we started to think about how those could be grouped together into loopers. At the end of the night, we arrived at the drawing in Fig. 4-3. Along with the harmonizer prototype, this was what we were able to achieve in one day of collaboration, but both would require a lot of refinement before becoming truly usable on a festival stage. The next time Jacob came to Boston, he stayed for a week and we did a deep dive on looping and the harmonizer. We took over a small, unused room at the MIT Media Lab and set it up with all the instruments he thought he might need on stage. We set up buttons and faders for each of the 6 loopers and tried to make an arrangement of one song, Stevie Wonder's *Don't You Worry 'Bout A Thing*. This involved a lot of experimentation and looping, but by the end of a few days, Jacob had almost gotten it down. After all the experimentation, we knew we would need to meticulously plan out each bar of the song. We also determined that rather than have Jacob manually trigger all the loopers, it would be best to have a system in place that worked automatically, based on the number of bars that made up each section. We wanted Jacob to be focused on playing instruments, rather than triggering the loopers at the right time. Due to the complexity of arrangement, the triggering of loops was taking a major portion of his attention, and we wanted him to spend his time being musical rather than remembering when to push buttons or foot pedals.

4.3 Refining the Harmonizer for Touring

Simultaneously, we were able to improve the harmonizer. I came up with a plan of using a small industrial PC, made by Portwell, with no moving parts or fans, to run the Reaper session. The computer would sit in a Pelican case with an audio interface, a small mixer and the TC Helicon, which we chose to use, in parallel, to give the impression of lower latency and because its high voices were quite appealing. This was all squeezed in and wired up so that using the harmonizer was as simple as plugging in a mic and power, connecting the outputs and turning on the computer. Internally, the computer ran a 'portable' version of Reaper that we configured with all of our custom scripts and plugins. It also ran a utility called "Deep Freeze," which is a block level hard disk restore tool. Deep Freeze works by restoring the contents of the hard drive to a known point at every boot. Nothing done on the machine is saved, so every boot is completely deterministic. This helps the machine to "behave like a toaster"— every time it's turned on, it does exactly the same thing it did the last time.

All of these components fit inside a Pelican 1510 hard case which fits in the overhead bin of an airplane. This was important, since we wanted to take it with us to Montreux.

For the transition between *Don't You Worry* and the next song in the set, *Close to You*, Jacob asked for some additional features. First, he wanted to be able to take anything he was doing on the instrument and freeze it, like the sustain pedal on a piano.

I found an open source reverb called Ambience[7] that had a hold function to create an infinite reverb. We also wanted to script this so that it behaved like a sustain pedal. This required latching the hold button when the sustain was pushed down, but then tweaking some parameters when the pedal was released to reset the reverb algorithm. Otherwise, sustains too close together would bleed into one another. It also involved another

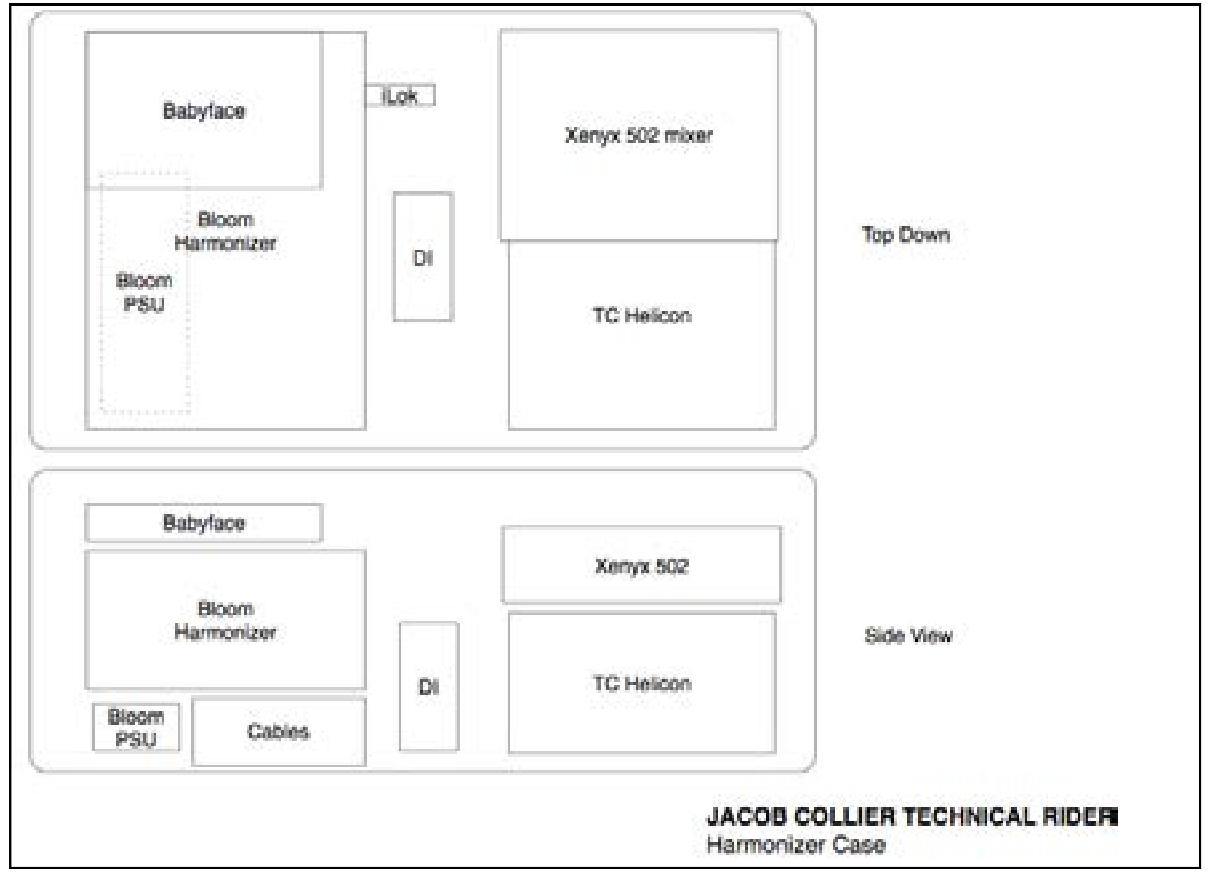


Fig. 4-5: First drawing of harmonizer components

[7] "Magnus' Plugins." Magnus' Plugins, magnus.smartelectronix.com/#Ambience.

specially configured gate to simultaneously mute the input to the reverb and open the output without causing a gap in the audio. Jacob was able to change the volume of this freeze and ended up requesting two more, so that eventually, he was able to layer three copies of 12 frozen voices on top of one another and then add another set of 12 live voices on top of that. Each freeze had two modes, a latch and release and a momentary trigger. To control them, we used a set of sustain pedals, but we also used a set of midi footswitch pedals (Logidy UMI3.) Since these were quite hard to use in bare feet and Jacob wanted to perform without shoes, we taped a set of spoons to the pedals. He had a set of spoons he was using for percussion and they proved to be the perfect pedal covers to make freezes easier to hit. These spoons became an iconic part of the harmonizer setup.

Once we had the sustain working, Jacob asked to be able to transpose some of the frozen voices up 156 cents—the difference in tuning between the end of *Don't You Worry* and the beginning of *Close to You*. I did this by modifying one of the superpitch JS DSP plugins[8] in Reaper to use a custom maximum tuning value of 156 cents. We assigned this to a fader on the MIDI keyboard controller for easy access.

With these elements in place, the Harmonizer was what we called version 1.0. It sounded good, felt good, was easy to transport, and was readily usable on stage.

4.4 Show Control with Ableton Live

We set up a second week of preparation work right before a trial performance at Ronnie Scott's in London. The goal of the week was to refine a looping setup that did not require Jacob to trigger every looper. We were inspired by artists like BINKBEATS[9] who had created interesting and seamless arrangements yet did not spend a lot of time pushing buttons.

That said, much of the existing examples of live looping shows either had people triggering all loopers manually or following a timeline. I wanted to give Jacob the flexibility to trigger moments of the show manually when that was the best way to hit the desired timing, but also to keep other parts of the set on a timeline where that was needed, as well. After doing some research, it became evident that although many people were using Ableton for timeline or manual clip triggering, not many people were doing both at the same time. There were other

[8] Samelot. "Samelot/Reaper." GitHub, github.com/Samelot/Reaper/blob/master/Effects/Pitch/superpitch.
[9] BINKBEATS Beats Unraveled #2: Getting There by Flying Lotus, YouTube, www.youtube.com/watch?v=X-aeDh65mtzU.

reasons for avoiding the typical timeline setup, the most important being that a large show on a timeline can be very difficult to edit during rehearsal. Making a small change can ripple through the rest of the set, and it is not easy to jump around, or run the show out of order. I had learned, by this point, that Jacob worked by spontaneously testing musical ideas and changing them if necessary, and it was critical to be able to easily modify timings and other elements of the performance without needing to go back and verify everything that followed in the session.

Since Jacob wasn't able to hire a dedicated operator for the Ableton system, I imagined that the system would essentially "run itself," with no real interaction from anyone during the performance. That meant putting the entire show in a single session and allowing Jacob to navigate the session from the stage, with simplicity.

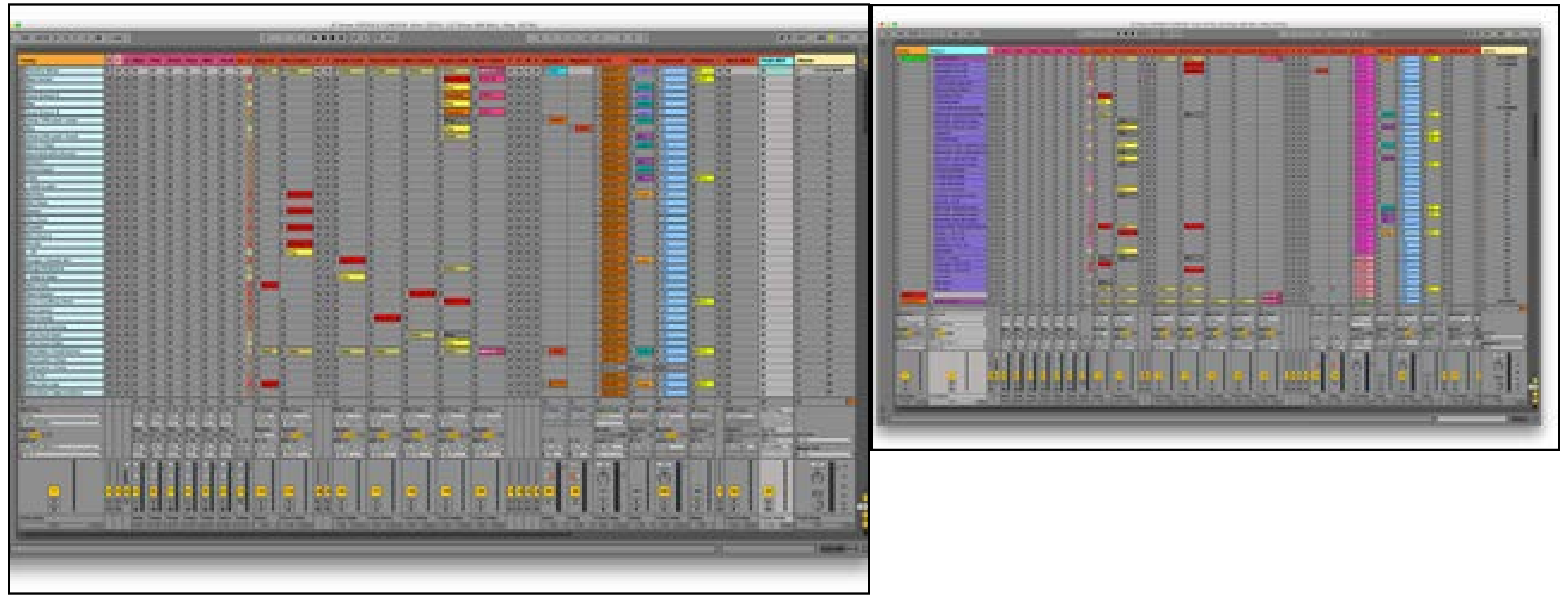


Fig. 4-6: One-Man Show Ableton session screenshot

A typical strategy for automating Ableton live is by using dummy MIDI clips with a MIDI loopback interface. This works because it is possible to attach a MIDI message to any function in Ableton. Using the IAC Bus on OS X, it is possible to have Ableton send messages to itself and trigger functions of the program automatically. This is done to automate the control of loopers within Ableton and is often used for other functions, as well. MIDI notes are ideal for triggers, but continuous data can also be used for faders, panning and other controls.

Each looper was configured to listen on a different channel, but the notes were the same. In this way, the MIDI clips for controlling each looper could be cut and pasted between the tracks for each looper. The tracks determined the output channel and therefore made loopers selectable with the same set of MIDI notes.

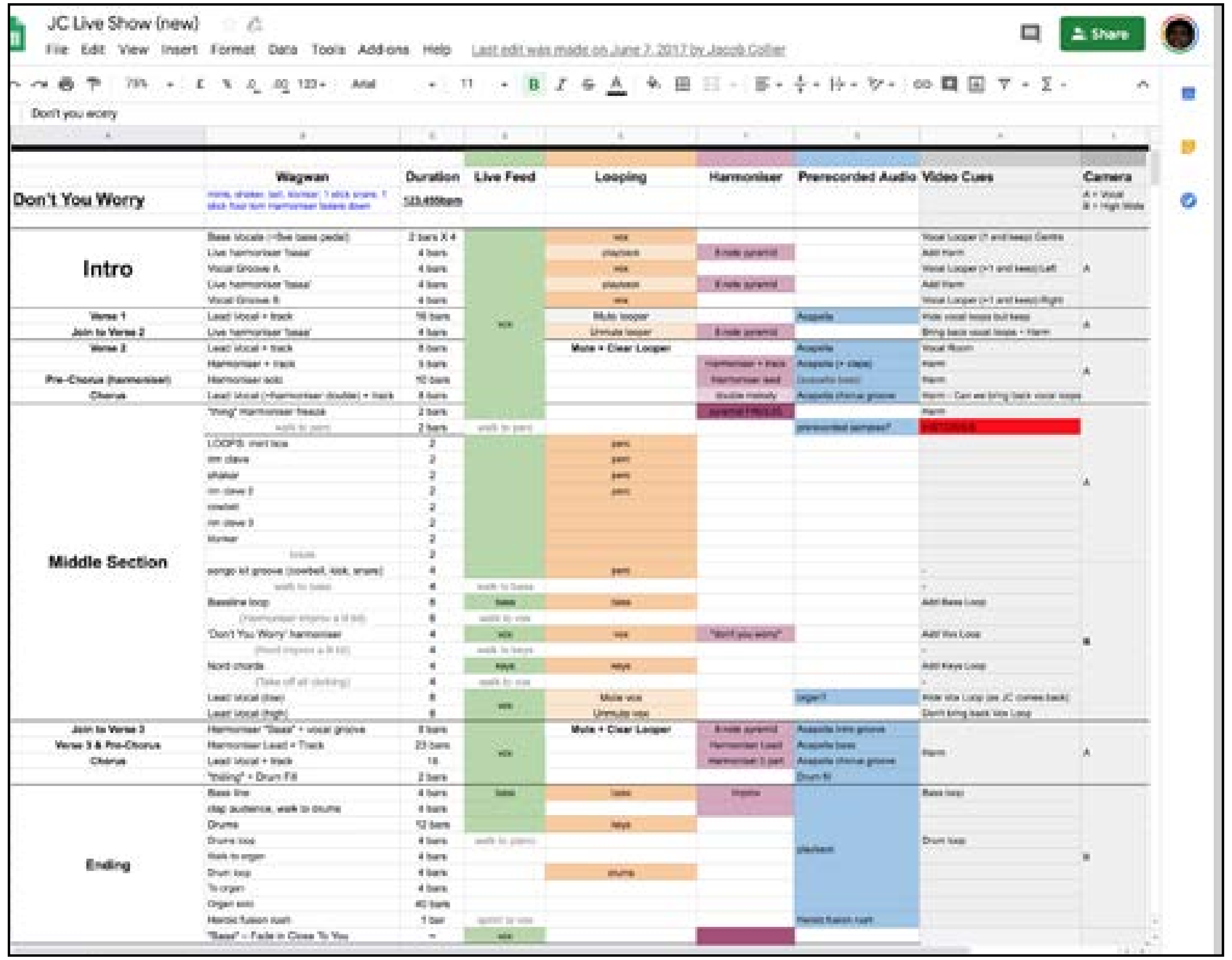
To control timing, I grouped all the looper control MIDI tracks into a track group and triggered each row of that group, based on another set of dummy clips. These clips simply triggered their respective control group slot, but used follow timings to ensure proper musical spacing between each firing. Because we needed so many of these clips, I created a small Python script to generate hundreds of small MIDI files. Since each of the timing clips had to trigger a looper and run via the OS level MIDI routing, I needed a way to ensure the loopers would fire on the bar line despite the delay in that timing. In order to achieve this, the loopers were quantized to the bar (in certain cases the half-bar) and the timing dummy clips set to trigger the control groups one beat before the bar. This leads to some complex behavior, and can make it difficult to rehearse when jumping into the middle of a musical section, but it is possible to jump to a specific musical section by modifying that start clip to be shorter by a beat. The initial count-in is shortened by a beat so that all following clips can be their real lengths.

With this system, section lengths are defined by their follow times and a single set of notes controls the loopers flexibly. This makes it quite fast to add and program a new song, or modify an existing one, by simply typing in follow times rather than having to drag elements on a timeline. It also means that by altering follow behaviors, it is possible to have certain parts of the session be triggered manually even while others are carefully timed.

To achieve a tempo or meter change, there is a slightly different mechanism used. In this case, the timing clip is modified to trigger both its control group and the next scene's master clip after the proper amount of time. This master clip then triggers the control group of the next scene, so that only the follow time is used on that timing clip. By writing time signatures and tempos into the name of the master clip, the session will change as the master clips are triggered.

Using this system, we were able to enter arrangements for all the songs in the set that Jacob hoped to perform. We eventually created a similarly formatted spreadsheet shown in Fig. 4-7, so that Jacob could notate his arrangements and timings precisely for me to add to the session.

Fig. 4-7: One-Man Show spreadsheet represntation



system design, I tried to make loops and playback indistinguishable. In many cases, the playback is made obvious because it sounds "polished," compared to the live loops. In our case, we tried very hard to ensure the loops would feel polished in the same way with careful mic choice and placement. On specific occasions, we made the playback feel a bit less polished on purpose. Our initial system design used a small digital mixer to submix the loops and live microphones, but I was worried about having enough control over the individual elements to keep the illusion of loops and playback being interchangeable.

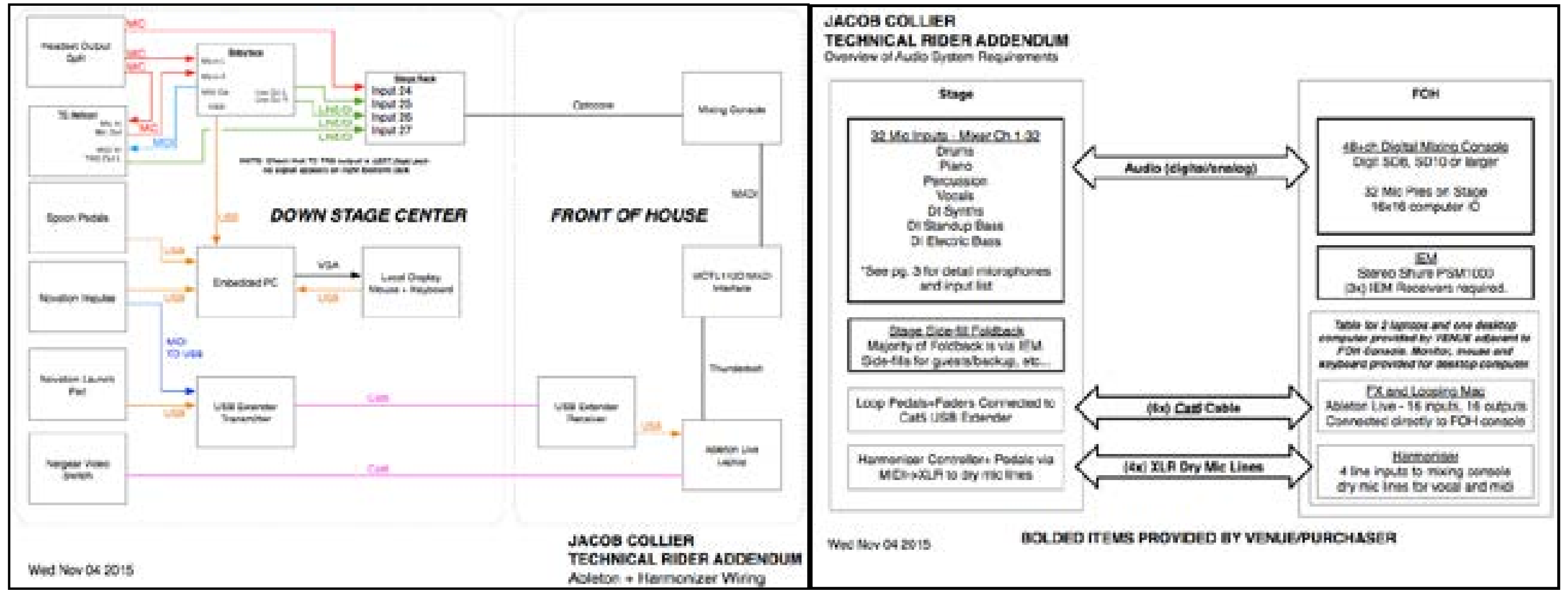


Fig. 4-8: One-Man Show Montreux system diagrams

Instead of using a separate mixer, we chose to interface the Ableton system directly to the festival's Digico mixing desk via MADI. This would give the mixing engineer (me) access to all the raw inputs, loopers and everything directly on the mixing desk.

With this setup relatively stable and agreed upon, we began rehearsing first in Jacob's home to try to develop and refine all of the arrangements.

Fig. 4-9: One-Man Show Montreux stage layout drawing

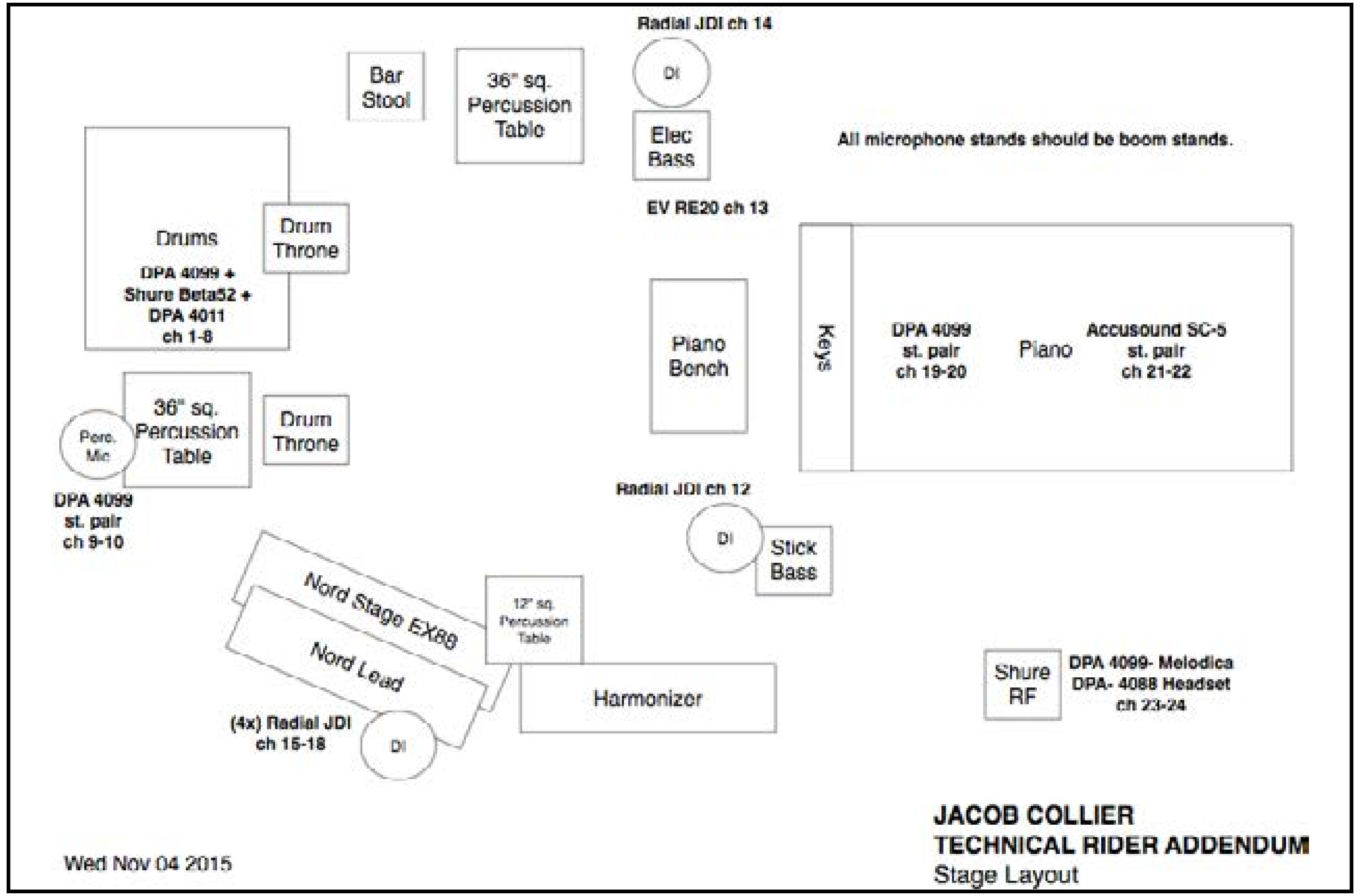


Fig. 4-10: One-Man Show final testing before Ronnie Scott's premiere

Once we felt comfortable with the arrangement, we moved the setup to a small dance club in Hackney Wick, and we spent two days working with a PA system, and mixing console, to refine and practice the arrangements. During this time, I had also been working with my friends, Louis Mustill, Will Young and Arron Smith, at Artists and Engineers in London to make sure our looping system would properly send commands to a VVVV based video looping system they had created. This system utilized two kinect cameras to extract Jacob from the background and loop video of his playing along with the audio. This worked well, as Jacob typically jumped around the stage from instrument to instrument throughout the entirety of his performances.

The combined video and audio looping resulted in a stylized live version of Jacob's YouTube videos. The audience could watch along as multiple "Jacobs" played a variety of instruments all over the stage.

This was rehearsed for 3 days at the dance club, and then performed at Ronnie Scott's Jazz Club in London. We then had a month to make more changes to the setup and the show - mainly tweaking arrangements and changing the video looks, and finally, we performed the set for 4,000 people on the main stage the Montreux Jazz Festival on July 16, 2015.

The set was received very positively by the press and reviewers. The Guardian proclaimed that Jacob was "Jazz's new Messiah," which seemed flattering, if a bit overblown.

4.5 Shrinking the One-Man Show

Jacob went on to release his first album, *In My Room*[1], in July of 2015, which I helped to mix at Remote Con trol Productions in Los Angeles, California. We were excited to try an experimental workflow in the mixing process, involving several computers connected to a Euphonix System 5 Mixing Console. Reflecting on this experience, I am proud of the sound we achieved on the album, given some difficult circumstances. Once the album was released, we added several of its songs to the live show. The show was gaining popularity, with many promoters asking to book Jacob all over the world, but its unique technical requirements (i.e. a digital interface from the computer to the mixing desk) made it difficult to accept some performances where there was not enough time or equipment to integrate the live looping system.

For the first US Tour, we were extremely strapped for cash, and the only way to make the shows economical was to do the tour with just two people on the road: Jacob and me. We could not afford to stay in hotels, so I called my friends from high school and college, seeking free lodging. We slept on friends' couches for the tour, which was 18 performances over 27 days. The schedule was grueling, and apart from that, I had to come up with a way to bring our setup on planes in a package that could be wrangled by only me, carried up stairs to walk-up apartments, checked as luggage by airlines, etc. I ended up putting together a small 4u rack with a similar set of equipment to the original drawing— a Midas M32 with the computer connected as a sidecar. I ran a laptop, MIDI faders and tablet at the mix position and we connected the mixer to whatever PA system was available in each venue. The 4u rack contained wireless in-ear monitoring, Jacob's wireless headset, a stage box and the mixer. We brought the harmonizer, an upright bass, a MIDI controller, and a case of smaller percussion instruments and cables. With this minimal setup I could have the stage set up within an hour or two after arriving at the venue. We would then spend an hour on sound check, do the performance, strike the setup (which usually took about an hour), sleep for a few hours, and finally head to the airport for the next city and performance.

After that tour, we vowed to never again take the show on the road with only two people. It was simply too intense and we both became quite ill afterwards. For the winter and spring headline performances, we added two people to our team, Jose Ortega and Claudio Somigli. Jose had helped in Montreux and was familiar with the big setup based around a Digico mixer. We thought that since we had more people on the team, we could expand our mixing setup with more inputs. I developed a version of the show using the Midas with a 32

Fig. 4-11: Rehearsals at Bloc in Hackney Wick

[1] Moon, Tom. "Review: Jacob Collier, 'In My Room'." NPR, NPR, 23 June 2016, www.npr.org/2016/06/23/482809010/first-listen-jacob-collier-in-my-room.

channel stage box, utilizing about 28 of those channels for mics and the remaining channels and aux inputs for computer IO via USB. This allowed us to add dedicated tom mics, percussion mics, and a pickup to the piano which improved gain before feedback. This required more connections and time to build and check, as the system needed a separate stage box and more mic lines.

At the same time, we were starting to get festival offers which had huge audiences (tens of thousands), but we needed to be able to fit onto a shared stage without a lot of time for setup or sound check. The table provides representative numbers to compare a typical headline and festival timeline.

Table 4-1: Differences in timing between a festival and headline show

Headline Show	Festival
2 hour build	Build off stage on risers
1 hour sound check	No sound check, only a 10 minute line check
30 minute change over	10-20 minute change over

A particularly harrowing experience at the TED Conference[2] was the last straw. Because of the complexity of the setup (and an overeager stage hand) we had a major failure of a component, which overheated as a result of being covered with fabric. Since the setup included a number of extra cables, we misidentified the problem as a failed cable. There was a huge number of cables to test and all were poorly labeled. We ended up frantically scrambling 10 minutes before the start of the talk, as the audience was entering the hall. Ultimately Jacob did the talk with one of the cameras on stage not working. I have never been so stressed. After that incredibly intense experience, we pushed to minimize every possible connection and cable to the most important subset of inputs and equipment. I made a 3rd revision of the system with a re-engineered rack to minimize all cable connections, going back to the 16 input version of the show with a custom patch panel. Tom mics were not worth the extra cases and cables we had to carry for the second stage rack, however, we wanted to keep the piano pickup and percussion mics. In order to have the required number of mics (19), we used a Y-Cable to put the drum overheads and the percussion mics on the same set of inputs. We used the distance of the mics (and attenuation pads) to achieve a fixed level offset between the two sets of microphones. Certain instruments (Melodica, Guitar) were used on the same mixer input and a stage tech switched the mic cable between songs. Several loopers and synths were changed to mono (bass looper, organ pedals, etc.). We also found an excellent

[2] Collier, Jacob. "A One-Man Musical Phenomenon." TED, www.ted.com/talks/jacob_collier_a_one_man_musical_phenomenon.

project MixingStation[3], which was an android reimplementation of the M32 OSC control protocol that added a lot of professional features, such as DCA spilling, custom layers, and macros attached to custom knobs and buttons. We programmed the show on customized pages and layouts so that everything was easy to access, and needed minimal flipping of pages. We added an ethernet based control surface which had scribble strips. This made it easy to see which channels were selected and the setup at FOH was simplified, with both the tablet and surface connected independently by ethernet (for the first US tour, a set of HUI faders[4] were connected to my laptop with USB which connected via ethernet to the mixer, and the tablets were on WiFi).

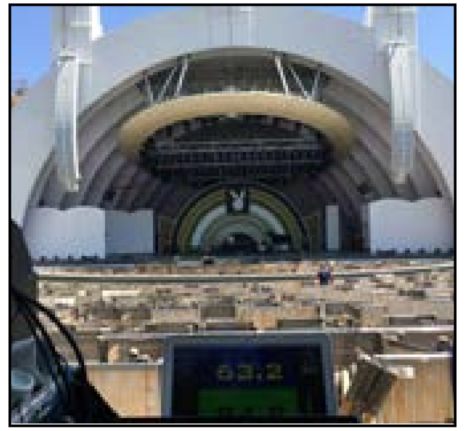


Fig. 4-12: Sound check at the Hollywood Bowl

With this system, the crew started to become very efficient at building the show. We also found that we could improve the mix every night instead of starting from scratch. Our shortest build was for the Bonnaroo music festival in Manchester, Tennessee. We had only 15 minutes to build and check the entire show while the audience was watching. After performing at this festival, we felt the show was quite robust and at the same time, the tour gained a lot of popularity. In the subsequent two years, we performed over 200 times. The system worked well for tiny venues, festivals and even exceptionally large venues. At the Playboy Jazz Festival, we had to install the system on a 50 ft. diameter turntable in 45 minutes, connected to the largest PA we had ever used for 17,000 audience members. The simplification of cabling was key because all the cables FOH to Stage had to be routed up and over the stage so the turntable could rotate properly. The rider documentation and show advance process was improved while we fine-tuned the technology as well. In the end the workflow became streamlined to a point where we could very quickly generate an interactive rider for the venue promoter to fill out with all required information.

The production advance became very quick in this case:

1. We would request all venue specs and send the interactive rider.
2. The venue team would fill in the rider and send back requested schedule
3. We would confirm any additional costs, instruments or equipment rentals needed.

[3] "Mixing Station." Dev, dev-core.org/mixing-station/.
[4] HUI - Human User Interface for Digital Audio Workstations - Reference Guide (PDF). USA: Mackie Designs Inc. 1998., http://www.synthmanuals.com/manuals/mackie/hui/owners_manual/hui_om.pdf .

We found that it was helpful to start our exchanges with the venues by immediately stating, "This is not a typical performance; please pay special attention to the technical requirements!" Otherwise, many smaller venues and clubs would tend to skim our documentation and miss big requirements, such as stage to FOH cable runs, or the fact that we would need a space to install our own mixing console. As time went on, we learned how to efficiently advance the performance to avoid surprises.

Fig. 4-13: Soldering the harmonizer in a hotel room in Monterey, CA

4.6 Harmonizer Hardware 2.0

The harmonizer, while doing well given the amount of travel required for touring, was getting extremely beat up. The Portwell computer was designed as an 'industrial' machine with shock and vibration ratings, but we found the power supply board inside was vulnerable to shock on its ATX connector. Several times the harmonizer was dropped, including off the back of a van and when being checked in on a plane. Each time, I would have to dismantle the PC and reflow the solder on the power supply. Eventually other parts started to fail. The week before TED, the machine stopped booting for good and corrupted its disk.

Even before that we had spent some time making the system more robust. The wear and tear on the cables and connectors was too much and it started to become quite temperamental. Like the One-Man Show mixer rack, we hoped to design a case requiring minimal connections and with proper strain relief. I created some basic designs for a metal chassis to house everything with shock mounting, and Brian Mayton and Tommy Moriarty were instrumental in taking the drawings I made and turning them into reality. We added a customized patch panel with transformer isolated inputs and outputs soldered directly to a DB15 cable for the RME Babyface. Brian made a 3D printed case for the TC Helicon, and we dismantled it, removing its display entirely. Brian came up with a way to connect the display and button panel if needed via a ribbon cable, but we generally programed the TC via MIDI. The capacitive display caused problems as cables would brush against it; so removing it entirely solved some issues we'd experienced where its programming would unintentionally change. We also modified the connectors on the rear panel to remove the guitar inputs, since these were never used and some others were failing. The computer was mounted in this chassis and all its cables were carefully strain relieved. A central power supply was cabled to everything. A USB over Cat5 extender was included for getting the Novation MIDI back to the side stage or FOH, if needed. This was also used to pass the harmonizer MIDI to the video system in certain configurations.

When the computer finally failed, we found a single board industrial PC by Neousys[5] which was actually smaller for the same spec (quad i7 3.4Ghz, 16GB ram). We were able to rebuild the system with that PC but with additional shock mounting right before TED. The system behaved well and has been relatively stable for 2 years. If I were to build a second harmonizer, I would use this machine and case design, since it has finally reached a point where it is stable and roadworthy.

[5] Team, Neousys Marketing. "Nuvo-3120." Neousys Technology, www.neousys-tech.com/en/product/application/rugged-embedded/nuvo-3120.

People often ask me how much custom software and hardware is in the harmonizer and the answer is: not much. It is true that if it were a completely bespoke device, we would be able to thoroughly tweak it as necessary, market it, patent it, etc. But, the circumstances around its development weren't conducive to making hardware and software from scratch. We instead spent our valuable time tweaking and combining components that we knew and trusted, rather than assembling basic building blocks. Now that we have a version of the instrument that we really like and trust, we could look at recreating what we have from scratch. This could be a project for the future, but its motivations would primarily be commercial, rather than creative or research-driven.

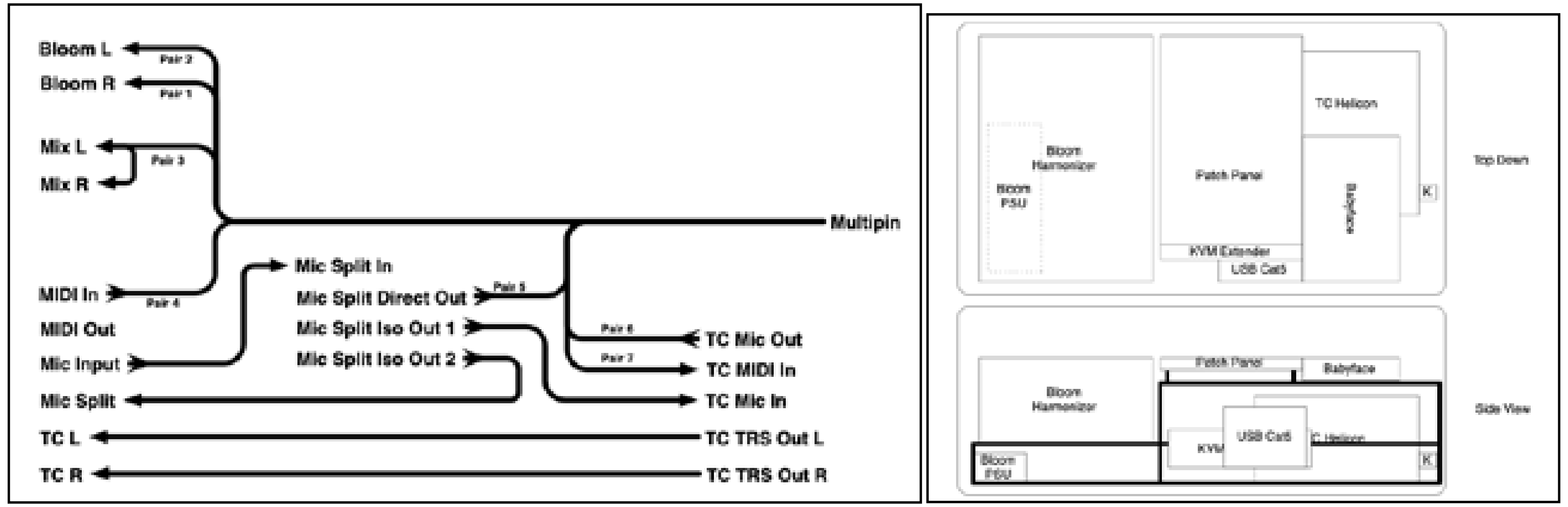


Fig. 4-14: New harmonizer initial designs

If I were to re-design the harmonizer today, I would consider using low latency DSP or FPGA platforms. FPGA leaves the most flexibility but has a longer development process. Our biggest struggle with the current version is the system latency (inherent because accurate FFT on low frequency input takes a relatively longer time). We play some tricks to lower the latency, but this could be done much more effectively by writing a vocoder algorithm from scratch. That would also allow for microtuning, hermode tuning and altered intonations which are features that we've long wanted to include. To develop this and add back in the traditional DSP elements (MS processing, customized enveloping, sustain, glide, freeze pitch shifting, EQ, etc.) would be quite a serious project. It would then need to go through a rigorous debugging and QA testing phase. Finally, it is essential to

consider the non-sound, non-musical features that make the current unit useful for high-stakes performances. We spent a good deal of time refining the sound of the device, but what makes it functional for Jacob are all the other features— its durability, its ability to be remote controlled, its simplicity of use, the fact that it can be used in many different configurations and power systems, its features tailored specifically to the songs and the set, etc. Perhaps down the road we'll start a company with a hardware vendor to do this, but for now, we're happy to have a device that sounds and feels truly unique and works well for our purposes.

4.7 Video Looping System

The video system for the One-Man Show was designed by Will Young and Louis Mustill. It also went through several design iterations as they tried to make it more reliable and more easily tourable. The system uses two depth cameras and two standard cameras to capture Jacob as he moves around the stage. The looping system sends messages to the video system to record clips of video for each audio loop. Using skeletal tracking, the system is able to perform real-time background extraction, so the loops are recorded with only Jacob and not the background. These loops are then overlaid on the live camera footage. The result is a live video of the stage which shows multiple Jacobs playing several instruments simultaneously.

The system originally used two Xbox Kinect 2 cameras connected to two Shuttle PC machines running VVVV[6] to do the background extraction and loop recording. The output of these two machines was connected to a video switcher, which was controlled via OSC, to choose one camera or the other depending on the scene. Later on in the tour, this system was changed, as carrying two Shuttle PCs was too bulky for touring by plane, and the camera quality on the kinect cameras was low. Two Logitech HD920 cameras were superglued to the Kinects and calibrated to be aligned together. Two Intel NUC Core i7 machines were connected (one webcam and one kinect each). The Logitech cameras provided an HD RGB feed and the Kinects were only used for depth. Software running on the NUC cameras relayed the combined depth and RGB feeds to a laptop via gigabit ethernet, with a dedicated graphics card running VVVV. The switch, looping and other effects were handled on VVVV laptop. A control surface was added to allow the video operator to operate some elements of the visuals manually.

[6] "a Multipurpose Toolkit." Vvvv, vvvv.org/.

4.8 Lessons from the *One-Man Show*

Apart from all of the technical considerations of making a show that works anywhere and doesn't break when traveling, we learned a lot about the creative implications of the *One-Man Show*. There are three concepts that emerged: timing, musical communication, improvisation. Our sense of these changed over time. There were three real periods of time where our sense of the show developed; initially when Jacob was learning the show, the serious touring period with several back-to-back weeks of performances, and the end of the tour when the performances started to be a bit more spaced out, but Jacob was intimately familiar with the performance.

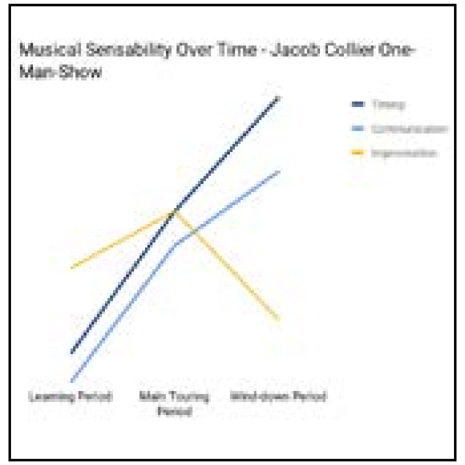


Fig. 4-15: Conceptual plot of performance sensibilities over time

4.8.1 Timing

There is a lot to be said about doing a show so heavily based on a click. For the *One-Man Show*, this specific type of arrangement was inevitable— looping requires the computer to be at the same tempo as Jacob. He is much better at following the computer than vice versa, especially given there were 200 venues with different PA systems, microphones, stage noise, etc. There was simply no way to reliably have the computer analyze audio and determine a tempo. As a result, Jacob had to follow a click track, which he's extraordinarily good at. During sound checks he'd often add and remove subdivisions, going out of time and back in again, just to challenge himself.

The advantages of a click are many, especially from a production standpoint. We always knew exactly how long all the non-vamped sections of the show were. Planning set times was much easier this way. We could do synced lighting and video experiences. Jacob started out following the click in a pretty standard way but really got comfortable pushing and pulling micro timings as he learned the show more completely during the middle parts of the tour. Since the structures are identical, he became comfortable taking risks and improvising in that structure, doing things like running around the auditorium or taking a solo until the last possible moment.

It wasn't all good though—microtimings were difficult to get to feel right, any kind of modulation or permutation, or even non standard subdivision of the downbeat could be incredibly confusing. These challenges meant that Jacob had to think about what he was doing and couldn't effortlessly be musical and feel the beat. He couldn't jump around sections, and even if the audience was demanding it, he couldn't easily play one more chorus, unless we'd already thought of that possibility beforehand. It would require an extensive UI to give him enough flexibility to arbitrarily jump around the arrangements for every song of every set. There are

some interesting examples of this, including systems and performances by Beardyman[7], Tim Exile[8], and Amon Tobin[9]. We could have put all the master clips on launchpad buttons or developed a custom iPad UI. Instead we chose to keep the system simple, avoid more parts to break and UI interactions that would take Jacob out of the moment.

As a substitute, we would quite frequently update the arrangements, loops and playback fragments. This was a careful process. Jacob would have changed playback and loops for every show if he could. I had to be incredibly cautious initially and ensure there was proper time for testing any changes he wanted to make. We got into a very tense discussion before the first show at Ronnie Scott's: Jacob wanted to change the playback content, but there was no time to test it. I said I would not change it because, in my experience, without testing this change in a runthrough prior to the show, we had to assume that there would be an issue either in the timing of the playback or in how I replaced the file. This was a really frustrating moment for Jacob. As time went on, we got better at changing the arrangements and understanding how to test new elements without needing a complete runthrough. Still, it drastically limited the show's agility and Jacob's ability to change things.

Another challenge of the click is wrestling with the impression of "man vs. machine." At the beginning the show went too quickly; Jacob had to remember when to play each instrument and looked like he was scrambling. This was the very last impression we wanted to give the audience: that the technology was moving too fast for him and he was barely keeping up. One of the advantages of Ableton was the ability to alter the tempo very flexibly. At the beginning of the tour, we were able to slow down or speed up arrangements to match Jacob's ability. In the end we didn't need this; instead, Jacob grew comfortable with the arrangements over time, and before long he began to give the impression that he was orchestrating, collaborating with, and even waiting for the loopers. He began to own the arrangements and the loops, so he could take extra time to interact with the audience. Towards the end of the tour, this got better and better. Doing a *one-man show* recently after a string of touring for the current album, we could not believe, upon going back to it, how slow the *One- Man Show* felt.

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[7] "Beatboxer." Beardyman, www.beardyman.co.uk/.
[8] "Performance-Led Musician and Technologist." Tim Exile, timexile.com/.
[9] "A B Ò U T." Amon Tobin, www.amontobin.com/about.

4.8.2 Communication

Jacob and I have both been fascinated with the idea that it is possible to communicate something emotionally without using words, only music. It's a basic idea, but an important one to evaluate when making technology-based and especially looping performances. When multiple players are on stage they communicate among themselves— asking and answering questions, proposing a new idea or different structure or direction, etc. There is also a good deal of communication with the audience. What we have found is this communication is often dependent on subtle details: microtimings, pauses, breaths, harmonizations, tension and release. Jacob has developed a very specific language that he uses to do this. June Lee talks about some of the music theory behind Jacob's language[10]. Jacob has made some descriptions himself, talking about ideas such as the circle of fifths and growing brighter and warmer using 5ths and 4ths[11]. For the *One-Man Show* there was no opportunity for Jacob to communicate with other musicians on stage, but he was able to have a back and forth with the audience which became more nuanced as he became more comfortable with the show.

Jacob became quite good at trying many little experiments throughout the show to see what would get the crowd excited or motivated. Playing at a jazz school, he found people got excited by scales, runs, odd subdivisions or reharmonizations. In the southern part of the US, there was a large contingent of audience members from the gospel community, and they connected deeply with chord progressions that were based on church music. In each place, Jacob tried to come up with some common language with the audience, and use it throughout the show. He could change the arc and articulation of his performance to react or connect with the crowd. With no one else on stage to draw energy from, he depended on the audience for emotional and musical support. The overall engagement and energy of the audience determined a large amount of the details in the performance. Some of the best performances had the wildest, or the most musically talented audiences. The first tour in the US was quite eye-opening for the two of us. The audiences were much louder, rowdier and more engaged than many of the seated audiences in Europe. The first performance like this at the Berklee College of Music in Boston was especially memorable, as was the Bowery Ballroom in New York City and La Cigale in Paris. In all these situations, the energy of the audience pushed Jacob to perform his very best and to start a musical dialog, communicating and surprising the audience members, to their enjoyment.

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[10] "June Lee." YouTube, YouTube, www.youtube.com/channel/UCwmBIysOtf85Bp-9Ibg3PVw.

[11] "Minor Fourths, Major Fifths: Anton Schwartz - Jazz Music." Anton Schwartz - Jazz Saxophone, 13 Oct. 2018, antonjazz.com/2018/01/minor-fourths-major-fifths/.

The solo show was a unique form for trying to do this. We experimented with many tracks from *In My Room* to try and find the right balance to allow communication and engagement. On one end of the spectrum were songs that required too much exact timing or where the pre-recorded elements did not leave enough room for flexibility. We tried and cut many of these songs:

- Michael Jackson's *PYT* was the first song that we decided did not feel right for the show and stopped performing. Because of its complex form and chord progressions, there wasn't really room for Jacob to have any kind of musical moment with the audience. He spent the entire time running back and forth or singing over a progression that was technically impressive but so precisely defined that there were not any reasonable alternate routes or harmonies around the arrangement, and this left Jacob feeling trapped. At one point we added subdivisions to stretch out some 11 beat bars to as many as 15 beats. It was fun to see the audience bobbing back and forth and then getting stuck on one side waiting for the next beat to come, but that was a bit sadistic on our part more than any sort of fun or dialog. *PYT* was also quite heavy on playback elements, which meant there was not enough harmonic room to improvise.

- *Hajanaga* was another track that we tried several times, and ultimately abandoned, for this reason. It leaves a lot of room for improvisation, but its form has very calm sections followed by huge raucous sections. Jacob spent so much time running back and forth from the piano to the drums; it didn't feel like he could really bare his soul in the quiet sections or get the big sections loud enough.

- *Hideaway* was one that we also had to tweak. It started out as a full arrangement with playback. We felt that was too impersonal and decided instead to make it completely loop-free but using a synthesized sine wave for the bass notes played on a set of organ pedals (the pedals were a nightmare to travel with but added an important visual) while starting on guitar. Then he would move to the piano and do the song as a traditional ballad with the whole thing growing and growing in intensity until an explosive climax with no electronics at all. That form really allowed him to read the audience and move as needed. He took a similar approach for the ballad every night, which was always the second to last song in the set. He played it simply, without any accompaniment, at the piano. He chose the song in the moment based on his interactions with the audience. This maintained a degree of authenticity and kept the whole set from feeling like it was on rails.

We went through a period of discovering this phenomenon since it was Jacob's first time doing this type and quantity of performances in such a condensed number of days. It was a honeymoon of sorts, with the most interesting and inventive audience communication techniques being discovered almost every show. On the first US tour, he discovered he could ask the audience to sing with him, which opened a new door to a litany of new activities. Once the tour settled in, we made fewer changes and his interactions became more familiar. I think the audience still enjoyed them, but we felt as though we had reached a comfort zone and had a palette of strategies to choose from.

4.8.3 Improvisation

Musical improvisation was a large part of the One-Man Show. Many of the songs had solos or fills that Jacob changed every single night, often tailored to the crowd. Still we found that with fixed forms and no other musicians on stage, it was hard for him to experiment (except for the ballads, which had a free form.) In *Saviour* we left an open section so he could teach the audience the words to sing along. At one point we imagined this section might be a time for him to experiment, but in the end it didn't work that way. The same thing happened with the large solo sections (*Don't You Know, Fascinating Rhythm*). For many of these solos, he started by sticking pretty closely to the album, especially for virtuosic solos like *Fascinating Rhythm*. This was remarkable because many of the solos on the album were constructed in sections. The fact that he could actually play them live was pretty amazing to see. The album solos required incredible technique but after a while they lost a bit of their spark and he began to experiment more, often slightly simpler in terms of number and speed of notes, but with more thoughtfulness behind the notes that he did use.

One area of the show where Jacob began to take a lot of musical liberty was with the encore. Each night for a long time, he would choose a different song to play. Starting at MIT in 2016, he played *Blackbird* and invited the audience to sing. This was a new dynamic, with the audience involved significantly and responding to the "challenges" that Jaocb presented. They first started by doing simple imitations, but these grew more complex until he had the audience performing harmonies and polyrhythms all within the structure of the chord progression of the song. Every night, the audience pushed for slightly different types of improvisation and Jacob became inspired to try different things. First they would sing, then he slowly added rhythmic elements, and then sound effects. On a few occasions, the audience started doing something unexpected, like making bird noises or stomping their feet, and Jacob found a way to work their contributions into the song. This was a wondrous feeling that Jacob wanted to build into the main part of the next tour.

4.9 Creative Goals for the *Djesse* Tour

When it came time to wind down the *One-Man Show,* Jacob was very nervous about what would come next. The show had many interesting and unique qualities, and he was worried that future attempts had the potential to feel too much like everything else that was already out there. Many times he said to me, "I don't want the next tour to feel like just another band show." Out on the road we had seen so many of those types of performances and while they were incredible, they never felt as fresh or jaw-dropping as what we had created. There is a healthy amount of bias in that judgement, but it was certainly true that we'd proudly made something that included a lot of our original goals. The show truly demonstrated that we could build something with technology that couldn't just be mimed or canned for a better result. The show breathed and changed every night, and once it settled, it felt as though Jacob was performing with the technology with the looping helping him to shine, rather than vice versa. The audience got to see Jacob doing something that required incredible skill—many said they didn't believe what they were seeing at first: that one person could play so many instruments so musically and still move fluidly from one to the next, supported by technology, to allow for unique arrangements and original songs. The whole package showcased Jacob's skills beautifully.

We agreed that any future shows should keep that element of magic so that the audience would immediately understand that this was unlike anything they'd ever seen before. Even with that, the format left us with some unfulfilled desires. Most significantly, there is only so much one can feel "musical" playing to a backing track. The show lacked any kind of on stage musical communication simply because there was no one else with whom to communicate. Seeing the interplay and communication between band members at other performances, this really hit home for us. Jacob felt like he was carrying the whole thing by himself. He also felt that too much of the show was prescribed and this did not leave enough room for him to truly be inventive, in terms of musical structure and feeling. As we began to imagine the next iteration of his tour, we challenged ourselves to think about how to maintain the uniqueness of the *One-Man Show*, while adding the musicality that Jacob desired and keeping the overall feeling from being too much like a conventional touring music show. To give one example of this: other concerts we had seen left the audience behind in terms of involvement, with the musician going off in their own world. Sometimes this can feel really good, but we'd experienced performances where there was a sense of the audience becoming bored. We wished to introduce new elements quite frequently to keep the audience engaged.

Becoming comfortable with one kind of mixer, getting to know the show and developing muscle memory results in a better mix than using a different fancy desk each night.

Lastly, we learned the musical importance of "locking down" as many technical aspects of the production as possible. So much of what you hear in Jacob's music is dependent upon precise details being correct. There really is a "correct" mix, and to get the relationship between sounds accurate every night, we realized that it was imperative to travel with a mixing desk. Even though the show was incredibly small for this kind of approach, we found that the only way to avoid the feeling of scrambling to get the piece together and to allow the entire team (both the crew and Jacob) to be musical was to ensure that everyone (stage, video, front-of-house and Jacob) had a consistent experience, every night. This meant traveling with a MIDI controller for the harmonizer, bringing the mixing desk, using the same kind of microphones and DI boxes each night, and carrying important microphones so they were always consistent, etc. This was especially hard for the crew; on more than one occasion,they arrived at a venue that featured a very expensive and fancy mixing desk, with every bell and whistle one could possibly want (such as Digico SD 7 worth $350,000) yet they would instead be forced to use a $500 mixer that had a lot of limitations. What we learned is that becoming comfortable with one kind of mixer, getting to know the show, and developing muscle memory was preferable to having more advanced equipment every time. For the future performances we hoped to maintain this consistency, but Jacob wanted the next shows to be larger, making it difficult to continue to carry the pieces to to do so.

4.10 Technical Goals for the *Djesse* Tour

We also learned many technical lessons to inform the design of the next tour.

4.10.1 Connectors

We had a number of connectors on stage for the One-Man Show. This included MIDI, XLR, USB, Cat5, DC, various power connections for different countries and even VGA in some cases. We found that for touring, all of the consumer IT connectors didn't hold up well. USB was the biggest problem. The connector is not durable and would gradually loosen over time. A connector that was permanently cabled (inside our mixer rack or in the harmonizer, for example) would gradually wiggle loose after enough road and air travel. It became standard practice to reseat all USB connections.

Where possible, we completely removed USB inside the harmonizer, switching to DIN-style MIDI for connections to the TC helicon and to the MIDI controller on stage, which was much more robust. We did this after a particularly bass-heavy performance in Puerto Galera in the Philippines. The backline company provided a faulty MIDI controller with a bad USB port that was aggravated by the fact that the PA was a Function One system with subwoofers that were not cardioid. There was a collection of low end vibration right where the harmonizer was positioned on stage. Particularly loud low notes would resonate the USB connector and cause the device to disappear. The harmonizer runs Windows 7 and this caused Windows to need to unload and reload the driver each time. After somewhere around 60 to 70 times, the entire machine locked up and the ASIO system failed with a horrible buffer underrun sound. This was a really bad noise on a huge PA system. At the same time, whenever the USB lost connection, the MIDI controller (which was bus powered), would power down. The MIDI controller we were provided sent a string of sysex and program changes when it powered on. This was sent every time the USB connection rattled. Initially to solve this problem, we added PVC tape to the USB connectors in such a way as to add thickness to the plug without interrupting the ground on the shield of the connector. In other cases we deformed the connector (worked best on USB A connectors) to keep it tight.

Becoming comfortable with one kind of mixer, getting to know the show and developing muscle memory results in a better mix than using a different fancy desk each night.

We went through about 5 Novation Launchpads. They did not hold up well in general and the USB connectors would eventually fail by becoming desoldered from the main circuit board. We found that lead solder was significantly more durable and older products (before lead was banned in devices) held up much better than modern products.

With the advent of USB C, we went through many durability issues related to dongles and hubs. Many hubs fell apart or had a very short, non-replaceable cable which would fail at some point. In certain circumstances this could short the power management chip. We ended up with two Apple machines that were unable to charge on the port we had been using for the audio connection. Many of the USB-C combined hub and network devices took several connection attempts before the network interface was functional. Once up and running the system was fine, but determining the correct ports to use and getting all the pieces to function properly was always a challenge. The Ableton computer required a connection to the mixer via USB 2.0, a network connection (for remote control and to communicate with the video system), and a USB 2.0 connection for MIDI from the Novation controllers on stage (used for starting songs and to send MIDI from the harmonizer to the video system). This required several dongles and a fragile power connection.

For the *Djesse* tour, we wanted to drastically reduce the type and style of connectors on stage and our dependence on USB (especially 2.0) connections.

4.10.2 Redundancy and Notable Failures

Another incredibly risky aspect of the One-Man Show was the fact that we had no redundancy on the computer and harmonizer. A failure of the laptop meant that the show would abruptly stop. At the point when we developed the show, Jacob couldn't afford to buy a second laptop; so we did our best with what we had. A failure in almost any cable, connector, computer, etc. was possible and would stop the show. It was miraculous that we were able to perform 200 shows without having to end one early for technical reasons. The risk was compounded by the fact that we did not have a dedicated performance computer— Jacob, Jose and I used our personal laptops to run the show. This was a bit complicated when any of us had to install software updates, and it's not something I would ever recommend "professionally," but we were able to make it work for the first tour.

There were some notable failures that certainly made for interesting stories after the fact:

Malasimbo - Puerto Galera, Phillipines - USB failure caused the harmonizer to need a reboot. Jacob had to move to the piano to finish the encore.

Munich - Video operator engaged the "Link" mode in Ableton on the video system — it caused the audio machine to increase its tempo to catch up to the clock of the video machine. This resulted in a (quite comical) speed up of all loopers and playback with increasing pitch. Jacob tried to stay in time with the loops as they sped up and then stopped the session. The crowd seemed to love it.

Reunion Island (near Madagascar) - The launchpad USB failed. The stage tech did not know how to manually start the session for *Saviour*. Jacob started playing *Saviour* not remembering this, with the intro and had to stop because the session was never started.

Reunion Island - The harmonizer power supply became so intermittent that it failed repeatedly during the performance, even after being resoldered beforehand. To get the PSU to work, the solution that seemed to temporarily fix the issue was hitting the harmonizer or slamming it on the floor. The same stagehand who couldn't start *Saviour* had to walk out on stage, pick up the harmonizer and slam it on the ground during a song. This happened more than once. (The stagehand was Jacob's manager, Michael Peha.)

TED2017 - A stage hand covered one of the video machines with black fabric for aesthetic reasons. This caused the machine to overheat and fail.

Moscow - The video computer connected to the projector started installing windows updates in the middle of a performance.

Bali - A mandatory Apple update (for the touch bar) failed and caused Jacob's computer not to boot during a rehearsal. We had to boot from the recovery partition and give the machine internet access.

Austin, SXSW - A clocking issue in the mixing desk caused all playback and loops to be severely band limited. The problem was so bad we thought the PA had somehow failed and stopped the performance to reboot the sound system. It was hosted by the BBC and meant to be a showcase event for Turbo Sound and Allen and Heath. They blamed the issue on our computer system, even though they supplied a different system than was promised. In the end, they called back to admit it was their fault and not ours.

There were several awkward moments associated with changing playback and arrangements without proper testing:

Miami - Jacob changed the playback but accidentally bounced it with the click enabled. The click played through the PA system and we thought it was a routing issue. It was just the playback.

Boston - We changed playback for *Fascinating Rhythm* and it did not line up with the click track. Jacob had me disable the click track and did the entire song with no click, which the audience found very impressive.

Many of these issues could have been solved by redundancy in a few forms. We do not have a backup harmonizer, but traveling with a second one would provide options in case of hardware failure. For playback and looping computer issues, having a backup machine means that it is possible to try session changes on a single machine with the option to switch to a backup if something goes wrong. If a machine installs an update or has a hardware failure (could be USB, the audio interface, MIDI or any number of things) there is an entire second rig which can be used.

An important goal for the *Djesse* tour was to add as much redundancy as possible. From the production crew's perspective, we were running on borrowed time and it was a miracle we'd made it through every performance with relatively few issues.

4.11 Development of the *Djesse* Album Tour

The music for the next tour would be based on Jacob's nascent album project, *Djesse*. He planned to release four volumes, which together would make up the complete album with about 40 songs. Each volume would have a special aesthetic— Volume 1 was to be orchestral, Volume 2 folky, Volume 3 glitchy and electronic, and Volume 4 a mix of everything. We went to Amsterdam to record the Metropole Orkest for Volume 1, and Jacob spent about six months on the album. When it came time to put the live show together, we had to think about a way to represent the size and scope of the album on stage. Each volume would have its own tour with customized systems. For Volume 1, we now had to try to fit the sonic landscape of the Metropole Orkest on a stage with a small band, a far cry from trying to replicate his YouTube videos.

I was adamant about a few principles when it came to the tour's development:

◊ Nothing new or different works perfectly the first time. We should have as many rehearsals as possible to test ideas and also have live performances to test the feel of ideas and approaches with an audience. We are trying to find a balance of technology and musicality that allows an audience connection, and this is incredibly difficult without an audience as part of the exploration.

◊ New components of the show (arrangements, technology, players, etc.) never feel perfect initially. We should think about how the tour will mature and how the pieces we're experimenting with will feel several months down the line. A smash hit takes a while to get to be a smash hit. We learned this with the One-Man Show, where it took over a year to reach its full potential. In rehearsal the natural inclination is to judge something as if it were the final result. For many technical aspects of the performance (e.g., the reliability of the technology, the capability of interactive elements) this is the correct strategy, but for both the human and musical elements, we need to remember to leave ample room for growth and change.

◊ Rushed elements will feel unfinished and unrefined. It's truly important for Jacob to have creative space to come up with arrangements and show sequencing before getting into the rehearsal room. All decisions made in the rehearsal room have innate compromises based on the timing, rushed nature and ability of the players in that moment, who are generally over-saturated learning new material.

◊ In rehearsal, it's important to allow the musicians to be musical in their own right. We should create arrangements that allow each player to shine through authentically rather than prescribing parts to a level of detail that means the players are catching up. In some instances, this means making parts more complex as the tour goes on. Other times, it means introducing complex parts up front that players slowly get the hang of after several shows.. Ultimately, it can also mean letting the players come up with their own parts. Without this, the feel of the performance can become robotic.

The path to what eventually became the *Djesse* World Tour was quite long and the performance went through an evolution which I will describe in the following sections. In July of 2018, Jacob was invited to perform an entire show at a sold out Royal Albert Hall in London for the BBC Proms. We used this as a way to begin to experiment with some of the Volume 1 songs in a live context. Next, we tried several band performances, with different players around the world. There was a Volume 1 release concert at MIT in December of 2018, featuring

200 live musicians along with the final members of the newly chosen Jacob Collier band (Maro, Rob Mullarkey and Christian Euman). In January, there was a period of production and rehearsal and the first concert of the world tour was January 29, 2019 in Lisbon.

As of the writing of this document, the tour is still ongoing— we have only begun the Volume 1 parts of the tour. With all the sold out dates and amazing shows so far, I believe that we've yet to see the most magical parts of this show. It's still growing and changing and has yet to settle, which is incredibly exciting to watch.

4.11.1 The BBC Proms

Table 4-2: Show plan for the BBC Proms

Jacob was invited to perform Prom 7 in the summer of 2018, and we started to think about the best way to take some of these complex arrangements from the record and perform them, live. The process for making the album was a standard one for Jacob— First, he wrote songs which he orchestrated with a team of people: Jules Buckley, Diamiano Pascarelli, Vlad Nikolov, Stefan Behrisch, and others. These recording scores were tracked over a week in Amsterdam with the orchestra recorded to 60 tracks on a Neve VR72 in MCO 3. He took the recorded tracks of the orchestra and heavily cut and spliced them to come up with some incredible edits of relatively unplayable parts ("the superman orchestra"). Then he added hundreds of tracks of himself playing and singing along with the orchestra. In certain cases, he had bassist Rob Mularkey and others playing on the tracks, too.

The first step for the Proms was to go through the original recording scores and modify them to match the arrangements for the album after Jacob's edits. We generated stems for Jules and his team to do this. Then, Jacob looked at the scores and make comments or changes, as needed.

Next, we created a massive table, similar to the One-Man Show, to describe what was happening in every section of every song. This table became our bible and we used it to determine blocking for cameras, cues for TV and radio mixing, how to structure the stage changes and the show session, etc.. This table went through a number of iterations.

Jacob and I originally wanted to avoid playback and use the orchestra keyboardist to play custom sounds made for the show, but Jules thought it would be best to have the orchestra listening to a click for tempo and playing along with the score. This made a lot of sense, since for live radio and TV the timing of pieces are tightly controlled. Using a backing track and click meant every rehearsal could be counted on for timecode exactly. This also meant that Jacob could add portions of extra sounds (voices, effects, complex rhythmic elements) to the backing which would give a more accurate sense of some of the hundreds of tracks he added to the album after recording the orchestra.

I made the playback rig for the click and backing track using a pair of redundant Logic machines, analog 8 channel audio interfaces and a Radial SW-8 auto changeover unit. We were worried about the rig failing on live BBC radio, so redundancy was critical. All the songs were arranged in the session sequentially, with hotkeys to jump to each location. The machines shared an MTC clock to keep them synchronized. This was a bit complicated: the proper solution would have been to use an external clock to drive both machines. Instead, I would engage the clock to seek to and start each song, then disengage it and let both machines freewheel, so that if one machine stopped (audio hardware fail, Logic crash, etc.) the other would keep running.

We added verbal count-ins on the click track to give Jules and the orchestra a sense of how the session lined up with the arrangement. The input list for the playback rig was the following:

Channel 1: Tone for changeover box
Channel 2: Click
Channel 3-4: Stereo Vocals
Channel 5-6: Stereo Percussion
Channel 7-8: Stereo FX / Everything else

This arrangement worked well because percussion often needed careful processing, and since we were taking things quickly off album sessions it was good to be able to compress that a bit more than other elements.

Once we had an idea of the arrangements and the playback system, we began to work on the sound system design. DeltaLive was the official supplier for the Proms and had a lot of Digico and L'Acoustics equipment on site for the rehearsals and the show. We came up with input lists for the orchestra (see appendix) and a configuration of 8 mixing desks to run the production. The Metropole traveled with an SD7 and a personal mixer system for the orchestra to use— this was maintained for rehearsals and even recording, so it gave them a consistent experience, regardless of the situation. There was a second Monitor desk, also an SD7 for Jacob and other guests. At the front of the house (FOH) there was an SD5 and SD10, with the SD10 being used to handle the orchestra inputs, and to send in stems to the SD5, which had all of Jacob's inputs, the rhythm section and the guests. This desk was driving the L-ISA system in the hall, which was distributed to 5 hangs of L'Acoustics Kara speakers. In the OB trucks the BBC used a Stagetec system and two more Digico desks (sidecars for strings and brass). All splits were digital to the trucks, so care had to be taken to ensure proper clocking for the entire system of desks.

I was given a bay of the SD5 at FOH to manage Jacob's inputs. Since we had 3 vocal microphones, and I knew the arrangements best and how to balance the harmonizer, I was given the job of looking after his inputs and calling the arrangements verbally to the other production crew members. I was also responsible for a vocal bus and an effects bus, including a Waves SoundGrid system and an EMT 250 hardware reverb which was distributed to all desks and trucks for use on broadcast and in monitor mixes.

Fig. 4-16: Front-of-house setup for the BBC Proms

The show ended up being extremely stressful because of visa troubles with one of the guest artists, Mohamed El Kasri, who came from Morocco to perform, for the very first time, in Great Britain. As part of the album process he and Jacob wrote a song together and he came to play it at the concert. Due to border control issues, he arrived only four hours before the concert was set to begin. He was not familiar with the live arrangement

Jacob had made (as there was no recording of it yet) and he did not speak English. The original plan was for him to have attended rehearsals the night before to have some time to learn the parts and the flow of the song.

Instead, the team made the choice to spend valuable time before the show teaching him the song. This involved a lot of gesticulating and a translator. The BBC crew was extremely frustrated because the time spent teaching him the music was supposed to be used for a full runthrough with cameras and lights. We did not get to have a runthrough, so the entire team went into the show having not seen or heard some of the songs in the set. One of the veteran BBC Radio mixers, Marvin Ware, was dismayed at the lack of rehearsal, saying he'd never felt worse going into a show in his 32 year career. He was on a desk by himself with only a single assistant and no sidecar mixers.

Fig. 4-17: Redundant playback system in Tokyo rehearsal hall

In the end, the show went off quite smoothly given the circumstances, with some of the biggest applause for Mohamed's songs— it was the right thing to spend time getting him up to speed. In Marvin's words: "It was a tough show for us as we went out live on radio. As you know, it was a huge challenge given the number of inputs/rehearsal time in the hall and changing nature .... but I think we got there (apart from one bit of early chat from Jacob that I wasn't expecting and was searching for a troublesome mic on layer 5 of the desk at the time [...] There has been some great feedback in the BBC and in the outside world, and the comments on the mix seem to lead to the fact that we got a good mix with all the detail in there .... which is always a huge relief !"

The show was popular with promoters and we performed this version four times, in Tokyo, Japan for Blue Note Jazz, in Bremen, Germany for Musikfest, and in dedicated concerts in Utrecht and Amsterdam in the Netherlands in addition to the BBC Proms at the Royal Albert Hall. This gave some additional perspective to try it with different orchestras. We learned a lot from trying to play the album live this way. The first big lesson was that the click track imposed a unique set of constraints on the performance. It was easy for the orchestra to lose some of the feeling of the pieces when they were playing along with it, and especially with fast moving lines. These felt more subdivided than lyrical— songs like *Every Little Thing She Does Is Magic* and *Hideaway* were especially vulnerable. We also learned that Jacob's movements between instruments had to have real necessity with an orchestra behind him. There were moments that felt as though he was running for no reason, with so many expert players behind him who could have played the lines easily. Texturally, we learned the importance of the signature sounds from the tracks that Jacob added, as well as the scale and scope of the sound of the orchestra;the album needed both to feel authentic. Moving forward, our challenge was to think about the best way to keep those textures, with the extra sounds in sync with the band, but to minimize the way the click impacts the feeling of the show.

4.11.2 Finding the Band

We started thinking of different ways the band could be structured for the tour. The band would need to be relatively small to keep costs low for touring. Fred Harris at MIT suggested that we recruit a string quartet in each city, and we thought it might be possible to find a quartet that could also sing. In the end, we worried about the added logistics of making that work on a bus tour with an already intense schedule and decided to keep the band only.

We tried some test shows with various players from the jazz community: Sam Wilkes, Pedro Martins, Spanky McCurdy, Christian Euman and Rob Mularkey. This crew did some amazing performances. For each show, we'd have a rehearsal period beforehand for the players to learn arrangements and get to know what songs would be performed at that particular show. We'd carefully structure the rehearsals, and usually deviate from our agenda somehow, but try to cover as much ground as possible. Jacob was interested in finding a combination of prescribed arrangements and musical detail while still allowing the band members to freely be themselves and to avoid feeling trapped by the sheer complexity of the songs.

They performed on the roof at Output in Brooklyn, at Oslo in Hackney Wick, and on an outdoor stage for the San Sebastian Jazz Festival in Spain, and these performances were very well received.

These performances were quite instructive and solidified some of our earlier goals even further. We needed careful arrangements to avoid the show feeling too much like a jazz concert; even though we now had several players, having a click and backing track still felt too constrictive; we wanted the players in the band to have some amount of "Jacob magic"— playing multiple instruments, picking up new ideas really quickly by ear and with the ability to follow some musical idea on an unexpected, spontaneous adventure. Jacob also wanted the musicians to have a knowledge of jazz but not to be inclined to play typical solos or anything jazz-like as a default nature. This was particularly difficult. In the end, the touring band would be made up of Maro Secca, Rob Mularkey and Christian Euman.

4.11.3 Djesse Album Release Celebration at MIT

At the end of 2018 we had the new band (MARO, Christian and Robin) come to MIT to perform in a special orchestral concert, celebrating the release of "Djesse, Vol. 1" and his return to MIT hosted by Fred Harris and the MIT Festival Jazz Ensemble. This was a reprise of a concert in 2016 featuring the Jazz Ensemble and a full orchestra of volunteers from Berklee College of Music and the New England Conservatory. We worked from the BBC Proms arrangements but added a new song, *With the Love in My Heart,* that was arranged and orchestrated by Jamshied Sharifi based on the stems created from the album.

For this concert, we made the decision to avoid using a click track. There were a few reasons for this: first of all, the orchestra, big band and choir were not used to playing to click. We were worried about the band following and staying in sync with any playback elements. Second, after the Proms, we felt the click made the orchestra performance quite mechanical. Fred is a very enthusiastic and emotional conductor and the band was used to following him, adding pauses or extending sections. We liked this about the previous MIT concert and wanted to preserve that flexibility for this performance. The challenge was to find a way to preserve the signature sounds and feeling of the album, while still allowing the orchestra to play with flexible timing. To achieve this, we used a handful of strategies that had worked well in previous *Opera of the Future* productions but that I had only tried with Tod Machover's music. We provided MARO with a keyboard controller throughout the performance, and we gave her a combination of triggers and virtuosic parts. In some cases, we combined triggers on the bottom of the keyboard with melodic lines on the top of the keyboard.

For *Home Is*, I cut up each chord of the song and ran it through a very long 45 second reverb to extend the chord much longer than it was on the album. Since the reverb was so dense on the album, this worked well. I recorded only the wet signal from the reverb and assigned each chord to a chromatic note at the bottom of the keyboard. The top of the keyboard had a choir patch from omnisphere. Maro could use the modulation wheel to switch the syllable of the choir, so she could also play chords that way.

This was important because the track was actually recorded at A 432 and then modulated up to A 440. I had to transpose each of the album snippets so the live choir could perform the elements at 440, but that meant some custom voicing was required for the moment where the modulation happened on the record. When we went to rehearse with the choir, Maro could play the chord for reference or help to reinforce certain notes. The effect was needed because we had the choir spaced throughout the auditorium, holding candles with no microphones. For certain voicings they could be quite loud and the acoustic sound was impressive, yet for others, they needed help, especially with low notes. We had a smaller group, the MIT Vocal Jazz Ensemble, positioned up in the choir loft with microphones and close mics on the low voices to help fill out those notes. These voices were panned in surround sound using d&b Soundscape connected to 12 loudspeakers around the hall. Still, Maro's keyboard was needed, especially to help the dynamic grow throughout the piece and to keep all the distributed singers in key.

From there, we segued to *With the Love In My Heart*, where, once again, Maro used a patch that included both played parts and triggers. In this song, the triggers tended to be longer, textural samples that would underpin sections like, "the orchestral glow," which had synthetic sounds that the orchestra couldn't easily make on their own, but really characterized the feel of that part of the album. We extended these textures by looping so they could be played as long as needed, and to ensure that harmonic content within them worked, regardless of the timing of the orchestral parts. Like *Home Is*, we put the triggers on the bottom of the keyboard and left the top for virtuosic parts. In specific cases, Maro would need to transpose only certain voices. This was all implemented in MainStage using a Novation SL MK3 61 key controller. The controller allowed us to color code different regions of the keyboard so Maro could easily know where sounds were located.

Christian had a drum machine (Roland SPD-SX) that he used to play a number of synthetic sounds, such as the 808 section in *With the Love In My Heart*, and large gongs and crash cymbals for other effects. In certain cases, there were very long sections of texture or even playback that we wanted to layer along with the orchestra. Rather than give these to Maro (she'd need to hold down a note for a long time) we put them on some of the SPD pads. Instead of using the internal storage and sampler, I chose to use the SPD as a MIDI controller connected to the same laptop that Maro controlled. This meant I could move sounds around and save them along with Maro's program changes. We used Kontakt as a sampler for the SPD to allow advanced behaviors: for example, a sequence of drum samples that would cut off one another, except for one which could be toggled on and off, or to internally sidechain a textural sample with a kick sample, but only in a chorus. In *All Night Long*, we wanted to have some rhythmic playback; in this case, we made sure to have loud percussion and an affected count-in in the track. Fred got close to the tempo and then Christian triggered the playback so that the full ensemble would then lock in. We were surprised by how effective this method was for getting on and off of playback. This was useful when Jacob wanted a Gnawa feel, but it would be too difficult to expect the live players to learn and play. We could put the complicated and nuanced percussion in playback and not need to worry about spending valuable rehearsal time on details of technique. Finally, we routed MIDI from the SPD to the lighting desk to be able to trigger effects directly on drum hits.

The PA system for the hall was unique and required a lot of work to integrate into the space. We started, two years previously in 2016, to overcome some technical and architectural hurdles in order to rig speakers from the ceiling. Kresge Auditorium, where the MIT concert was held, has a domed roof, and there were not any load ratings or documentation for the catwalk system. Furthermore, in the 1970's the roof actually collapsed, so MIT was quite nervous about putting any extra load on the structure without careful analysis. Working

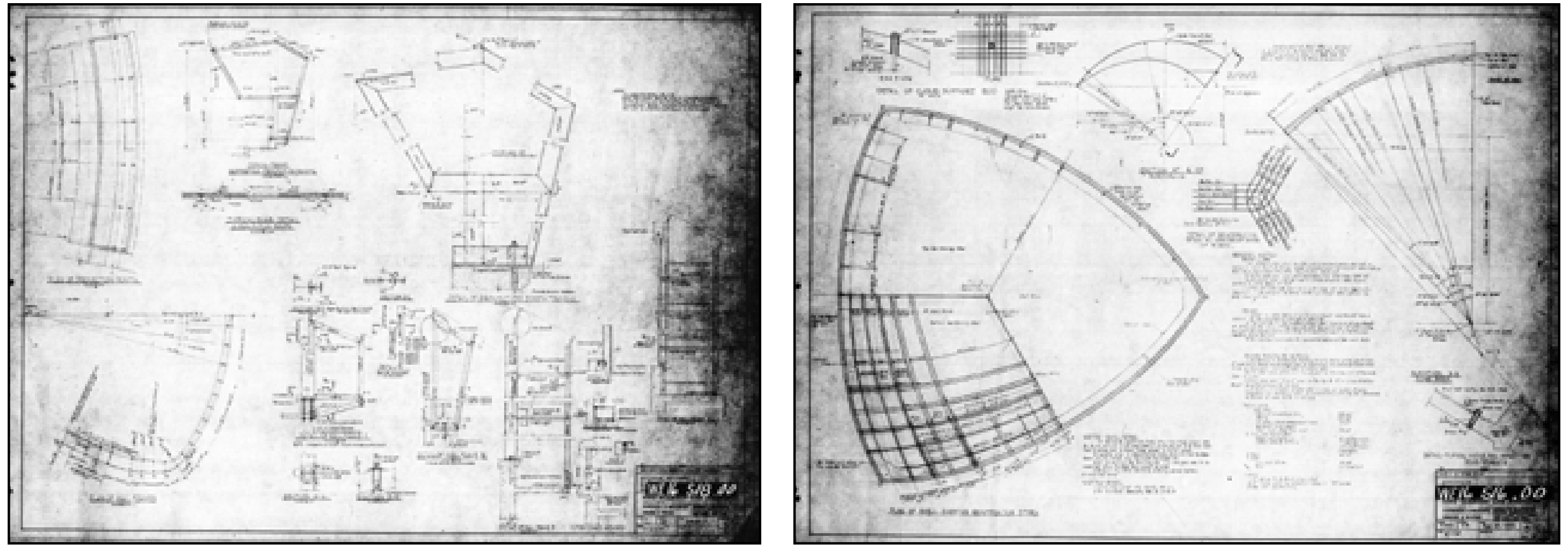


Fig. 4-18: Original 1954 drawings of Kresge Auditorium used for structural analysis.

with structural engineers (SGH, Atelier One) and the architects of the recent renovations to the space (EYP), we came up with a plan and structural justification to hang roughly 2000 pounds of extra equipment from the ceiling. The justification was dependent on the fact that there was no snow on the roof, because a snow load of just a few inches would have added the same amount of weight anyway. Once we were certain of that, we had to analyze the internal structure of the ceiling and catwalk system, based on "pull tests" that SGH had completed earlier in the year for installation of an updated sprinkler system. The analysis showed that the weight of the sound system should be supported by at least four "hangers" - two on each side of the roof. In order to properly distribute this load, we installed a steel strut between beams on the catwalk, with rigging points on each beam to a steel bridle.

It was important to be able to hang the PA in the air for two reasons:

1. With close to 200 performers on stage, there wasn't room to have speakers amongst the chairs and instruments.
2. Putting the PA up in the air helped to equalize the distance from the speakers to the first and last rows of the house, insuring that there would be less issues with volume and spectral discrepancy between front and back of the room.

To optimize the PA system for the room, we created a 3D model of the space using a homemade laser scanner and associated scripts (created by Brian Mayton) to generate a venue model that could be used with d&b audiotechnic's ArrayCalc software. The software optimized level and spectral response over all seating areas, and generated specific DSP for each speaker in the 8 cabinet line arrays on each side to implement the optimization. System tuning and optimization was performed by Shawn Duncan. The result was an incredibly detailed acoustic experience that was unprecedented in the hall.

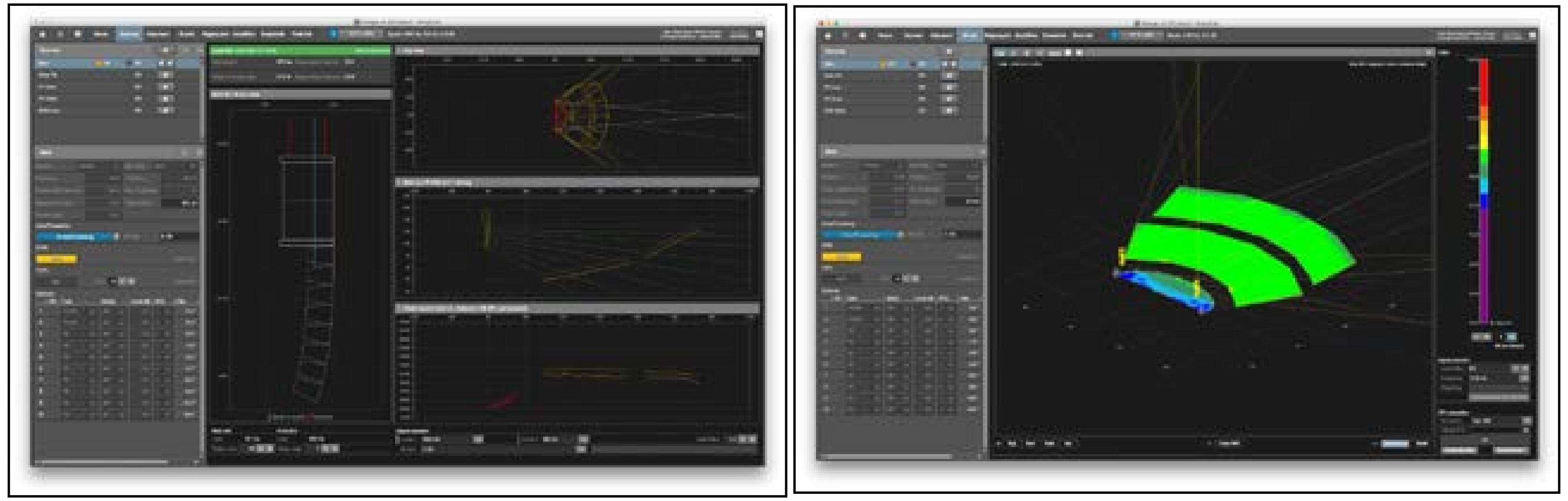


Fig. 4-19: PA system simulations for MIT 2016 Concert

For the 2018 performance, we used these drawings but added some more elements to the sound system: since we wanted to create an immersive experience for the music, we mixed the show in surround using SoundScape, another d&b product. This meant that the placement of all the instruments could be panned in space, individually. The 2018 performance had 112 microphone inputs and the SoundScape processor could only support 64 virtual sources. To accommodate, we had to mix down certain elements and position them as if they were single points. We tried to preserve strings for individual positioning since their sound is dependent on position, but drums and percussion, keyboards, etc., were sub-mixed in groups since they were located in close proximity to one another.

We changed our monitoring for the 2018 performance as well, and we had Jose Ortega come to Boston to operate a second desk, which powered in-ears for the band and Fred. Brian and I put together an 80 output headphone distribution system for the orchestra and choir to use. We asked musicians to bring their own ear-buds and headphones to connect to this system. The distribution boxes were connected to 16 Aviom personal mixers. This made recordings and FOH mix much cleaner; in 2016 we had a problem with floor wedges leaking into microphones. We also transitioned to DPA 4061 microphones which were close mics. This helped because the PA rigging points were actually above the string sections and the close mics meant much more gain before feedback. We used ground-stacked and flown cardioid subwoofers to avoid low end bleed on stage.

Overall, the project was a great success, and included hundreds of student musicians from local universities in Boston. We were able to test many concepts with the band, such as mixed keyboard parts and SPD triggers. We had the amazing privilege of having the choirs, orchestra and big band present and discovered the show worked without a click track. The next phase of the planning would require taking all we had learned and trying to use it to format the tour.

4.11.4 London Rehearsals

Leading up to a period of London rehearsals with the band, Jacob and I did some intense planning to try to come up with refined versions of everything we had done with MIT. There was also the unique creative challenge of performing similar arrangements, having them feel as big and spacious as the orchestra concert, but with no real orchestra on stage. Playback would feel too mechanical and we had already decided that we should avoid click wherever possible. What followed was a long period of testing sounds and writing down arrangements. Rather than use a spreadsheet like before, we made textual descriptions. A spreadsheet wasn't quite good enough at capturing the interlocking nature of the arrangements. Instead we wrote down events as they occurred such as Maro plays this sound, Rob plays that sound and switches to bass, Christian bring in a texture, Jacob plays and sings over the texture until Rob switches to guitar after the bridge.

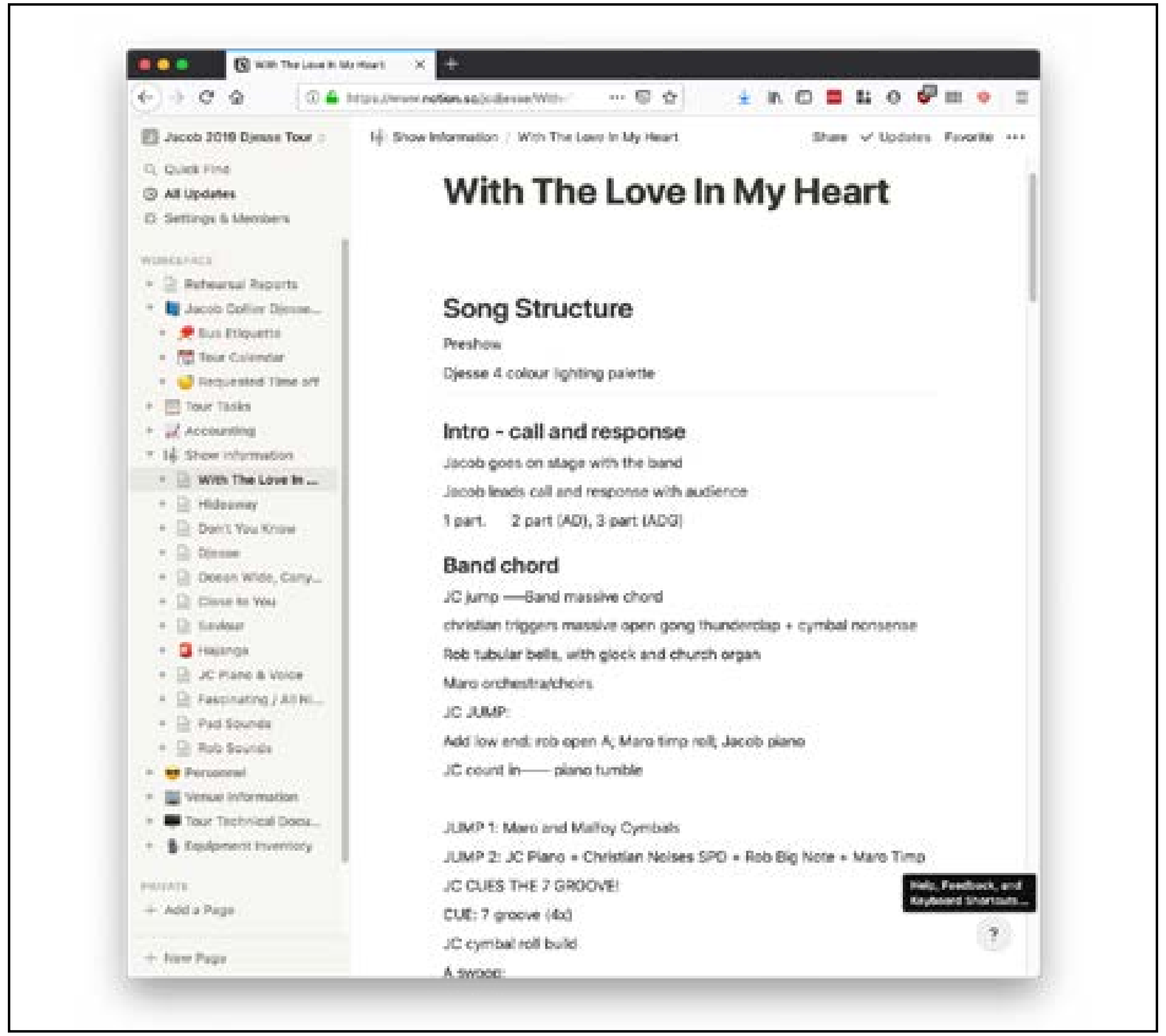


Fig. 4-20: Textual song flow notes from band rehearsals

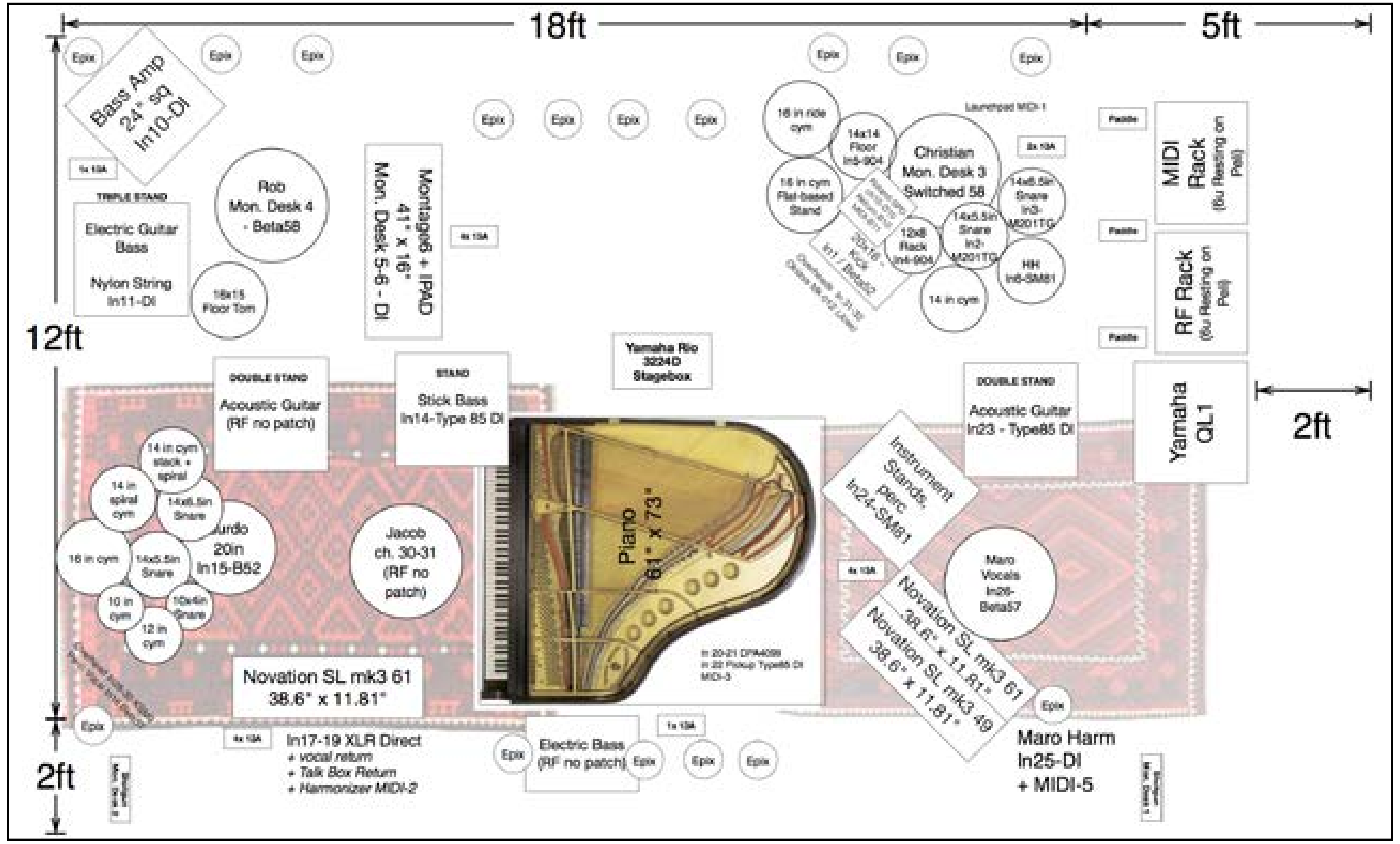


Fig. 4-21: Original Djesse tour stage plot

In the arrangements each band member would switch between many different instruments:

Jacob	**Maro**	**Robin**	**Christian**
Percussion (shakers, tambourine, wood blocks, cowbell) Acoustic Steel String Concert Guitar Electric Bass - 5 string Jazz Bass Upright Free-standing Silent Bass Keyboard Sounds - Talkbox Synth Surdo + Snares + Cymbals SPD Trigger Pad	Percussion (shakers, tambourine, wood blocks, cowbell, chimes) Acoustic Steel String Dreadnought Guitar Keyboard Sounds (Mainstage) Bongos SPD Trigger Pad	Percussion (shells, mini tambourine) Acoustic Nylon String Guitar Electric Guitar - Les Paul Styled Electric Bass - Fender Jazz Bass Upright Silent Bass (same as Jacob's) Keyboard Sounds (Yamaha Montage) Tom-Tom	Percussion (cup chime, shakers) Acoustic Drum set Electronic Drum Sounds (SPD)

Table 4-4: Djesse tour band member instrument grid

We had to create a stage layout to accommodate all the switching. We learned from the San Sebastian and Hackney performances that it was best to put Jacob in the middle of the stage. With Robin ducking in to use the upright bass occasionally, it made sense to place him upstage of Jacob. We also felt good about this because Jacob and Robin had been playing together for much longer than the other members. We wanted Jacob to be able to have eye contact with the other players. Maro and Christian were located near each other because we imagined they would share percussion, although ultimately everyone got their own complement of percussion and no sharing of percussion was necessary.

Jacob decided he wanted to do some looping as well, so we needed to integrate this on top of all of the other elements. From my perspective, I wanted to design a set of gear that would address our concerns from the last tour and recent shows. We wanted redundancy, robustness, portability, and flexibility.

Given all this, the system I made had the following list of requirements:

- Two laptops, connected to two MIDI interfaces, connected to two audio interfaces.

- MIDI mirrored from one machine to the second so both laptops receive the same input from all controllers on stage.
- Instantaneous switching from one laptop to the next, with both producing identical output at all times.
- No USB used for MIDI connections on stage, only DIN style MIDI or XLR MIDI
- Mainstage used for keyboard sounds (allowed us to pull from Jacob's album directly)
- Ableton used for Looping
- Novation SL MK3 for Maro sending program changes to MainStage
- Christian's SPD connected to an instance of Kontakt within Mainstage for triggering electronic percussion sounds to allow for sidechaining.
- Christian with his own set of patches so he could switch back and forth if needed, independently of Maro.
- Jacob and Maro each with SPD drum pads connected directly to Chrsitian's SPD.

We purchased two modern Macbook Pros running Macos High Sierra and used them for our "show computers." I wanted each computer to be able to run completely independently, which meant that they both needed to run Ableton and Mainstage simultaneously. To do this, we had to disable OS level power management features <put terminal command in the margin> which automatically throttled background applications. Each computer had the following applications installed:

- Ableton 10
- Mainstage 3.4.1
- Omnisphere, Keyscape, Trilian - latest versions in Jan. 2019
- Komplete - latest versions in Jan. 2019

All plugins were hosted in Mainstage with Ableton handling only looping. A USB C to USB 3.0 B cable connected to a USB and network hub. The network was used for Dante Virtual Sound Card (DVS). USB connected to a pair of MOTU Timepiece AV 8 channel midi interfaces. These interfaces are economical, simple and have hardware routing, which allowed mirroring of all the ports on one interface to all the ports on the second without the computer. The primary computer had a USB-C Focusrite 16Line Dante interface, which we chose because it had lower latency than DVS and provided redundant Dante support. A network switch connected this to a set of Focusrite 64R Dante to MADI adapters. These adapters were connected to Digico FOH and Monitor desks. The desks had macros to flip between Machine A and Machine B with all channels of both machines accessible via MADI at both desks. Dante is unique in its flexibility to allow this kind of routing. With this setup, any piece of the system could fail (network, computer, USB MIDI interface, audio interface), and the backup system would function completely. There were a few points where we had single redundancy: the network switch, the primary MIDI interface doing mirroring, and MADI interface on the desks. These are all extremely reliable items, and we had no failures in two months of touring.

The MainStage session had 16 patches associated with the 16 session selectors on Maro's SL MK3 controller. Each patch had a combination of Virtual Instruments and Kontakt samples. Some channel strips were global to all patches. These were for Christian's SPD, effects and reverbs, and some end specific routing for sidechain compression. Certain sidechain routing was performed on a per patch basis, so that part of a song might have sidechain and part might not, depending on when Maro would change her patch. One patch has an opening Low Pass filter that resets on new chords. This gives the impression of a dance music break down slowly increasing in intensity.

The instance of Kontakt running connected to the SPD had several instruments loaded with over 200 completely bespoke samples and fragments made just for the performance. The samples were pulled from the album or recorded specifically to use in the show. We had several types of instruments:

- A generalized instrument handled a majority of the sounds
- A special instrument was used for 808 samples, since this required some customized rolls and flams. Certain samples, such as HiHat Open and Close were exclusive, meaning they would cut one another off.

Fig. 4-22: First revision Djesse tour rack design

- A special instrument was used for each of snaps, claps, and subtle claps. These were used in multiple places and also had a round-robin to choose randomly from a small pool of files. For the claps, there were two velocity layers to allow high and low intensity claps. These were selected based on how hard Christian played the respective pad.
- A special instrument was used for noise floors. These were looping togglable samples that could optionally be exclusive. Special care had to be taken to track the on/off state of the toggles. A global reset button (a MIDI CC from MainStage) cleared all the toggles at the start of the show.
- Another instrument was used for one-offs. These samples would play all the way through and couldn't be stopped except by sending an "All notes off" MIDI command or Panicing MainStage. These were samples that we never wanted to stop—Gnawa playback, etc.

Much of this programming could have been done internally in the SPD. However this would have required more analog inputs, and sidechaining would have to be done in the mixer and programmed for each section of each song. Since Maro was already changing patches at these sections, we wanted to lock down the side chain configuration to ensure it would be exactly the same each night. Another big reason that I chose to use Kontakt was for flexibility in rehearsals. The Roland is quite cumbersome to program, with many button pushes. The available computer based editor for it does not allow one to play and edit at the same time, since the internal storage must be mounted as a file system. We spent a lot of time in rehearsal moving samples around, changing them out, stretching them, looping them, and changing their trigger behavior, and this would have been nearly impossible programming only on the SPD. I began the rehearsal period trying to write a script to generate the configuration file for the SPD, but quickly abandoned that in favor of using it as a MIDI interface. Rather than send patch changes, we used the internal patches on the SPD and simply changed the notes sent by each pad on those patches. This meant that no matter what patch Maro was in, Christian could always access all of his sounds, but Maro could still change behaviors of Chistian's sounds, such as side chain or reverbs, based on her patch. This double-tiered patching setup worked well, since it meant that testing and rehearsal could happen independently for Maro and Christian.

The Ableton session was actually identical in structure to the One-Man Show system. My goal was to be able to use the same hardware to run a redundant version of the One-Man Show if that was ever needed, so we based the band show Ableton looping setup off the exact same session and same structure as the One-Man Show. We used a launchpad located at the drums next to Christian to trigger the Ableton loops, exactly like the One-Man Show, however Novation released a Pro version of the controller with DIN MIDI outputs, so we no longer had to worry about USB. That said, the DIN connection used an adapter to non-locking 3.5mm TRS jack. We found if that connector became loose, it would send fragmented MIDI packets. We solved this by taping the cable in for shows. With this setup, we also needed to get inputs from FOH to the Ableton session. We used MADI for this just like the outputs, and because the system was Dante based, it was easy to route inputs to both computers. With this setup, it was actually possible (given a bit of extra input routing) to play a show mixing songs from the One-Man Show and the band show together. We haven't had to do this yet, but we are prepared if that opportunity ever presents itself.

TIME	Load In	Tuesday 22nd	Wednesday 23rd	Thursday 14th	Friday 25th	Saturday 26th	Sunday 27th	Load Out
10AM - 1PM	Load-In/ Set up	CREW CALL 10AM BAND CALL 12PM - Set Up and Plug In	BAND ONLY REHEARSAL NO JC Review Djesse, Don't You Know	BAND ONLY REHEARSAL NO JC Review With The Love, Close To You	BAND ONLY REHEARSAL Review Hajanga, Ocean Wide	BAND ONLY REHEARSAL Review Fascinating, Hideaway	BAND ONLY REHEARSAL TBD - Full Set Review	Pack Down
LUNCH		TEAM WELCOME LUNCH!! W/ JC!						
2PM - 6PM	Programming	Basic Sound Check (1 hr) **Learn Djesse** **-15 min break @4:30pm**	Lighting Run/Record (30 min) - Djesse, Don't You Know **Learn With the Love In My Heart** **-15 min break @4:30pm**	Lighting Run/Record (30 min) - With the Love, Close To You **Learn Hajanga** **-15 min break @4:30pm**	Lighting Run/Record (30 min) - Hajanga, Ocean Wide **Learn Fascinating Rhythm** **-15 min break @4:30pm**	2 Runs w/Notes +Record Run-No-Stopping Close To You (55 min.) Fascinating Rhythm (55 min.) - Break 20 min With the Love In My Heart (55 min.) Hajanga (55 min.)	Notes + Fixes (1hr) Full set tech rehearsal + Record - 3 hours 2 Runs (stops, no stops)	
DINNER				15 break				
7PM - 10PM	Line Check	**Learn Don't You Know**	**Learn Close To You**	**Learn Ocean Wide (2 hrs)** Leave 8:30 for 9pm Dinner @ Finchley!	**Learn Hideaway**	2 Runs w/Notes +Record Run-No-Stopping Hideaway (40 min) Djesse (40 min) - Break 20 min Ocean Wide (40 min) Don't You Know (40 min)	Notes + Fixes (1hr) Break - 30 minutes Final Dress rehearsal (90 minutes)	

Table 4-3: Djesse tour band rehearsal Schedule

We continued to use a wireless IEM system and added four channels of wireless microphones for Jacob's instruments to allow him to keep his mobility. This covered a bass, a guitar, Jacob's headset, and a wireless handheld. Shure was kind enough to loan this and a complement of IEMs, six stereo mixes for the band, the Monitor engineer and a guest, with 8 IEM packs to accommodate the band and stage crew.

The first band rehearsals for *Djesse* were intense. We had a period of 7 days, and our rehearsal studio was booked for 12 hours each day.

We started with a schedule but quickly deviated based on a few lessons that we learned:

- It was quite difficult to be productive for 12 hours per day. We learned this in our rehearsals for San Sebastian, but it became even clearer for this period. We had scheduled morning rehearsals for the band to work alone and afternoon sessions with Jacob. It became clear that it was best just to keep everyone together.
- We structured our original schedule focusing large amounts of time on single songs. Instead the band really wanted to jump around. We found this way of working helped their retention of new material too.

Towards the end of the rehearsal period, we began to pack for the tour. It was only at that point that we realized we had set ourselves up for a huge amount of work. The number of instruments and audio components was, in reality several orders of magnitude larger than the previous one-man show setup.

4.12 Touring *Djesse*

We began a two month long tour over the US and Canada.

Table 4-5: First tour schedule for Fjesse

Tuesday, January 29, 2019	Lisbon, Portugal	Capitolio
Wednesday, January 30, 2019	Pending Masterclass TRAVEL Madrid	
Thursday, January 31, 2019	Spain, Madrid	La Riviera
Friday, February 1, 2019	Spain, Barcelona	Razzmatazz
Saturday, February 2, 2019	OFF TRAVEL	
Sunday, February 3, 2019	OFF TRAVEL	
Monday, February 4, 2019	Germany, Munich	Muffathalle
Tuesday, February 5, 2019	Switzerland, Zurich	Kaufluenten Zurich
Wednesday, February 6, 2019	Lausanne, Switzerland	Les Docks
Thursday, February 7, 2019	Masterclass Paris	
Friday, February 8, 2019	France, Paris	La Cigale
Saturday, February 9, 2019	Germany, Cologne	Live Music Hall
Sunday, February 10, 2019	OFF	
Monday, February 11, 2019	Belgium, Brussels	Flagey
Tuesday, February 12, 2019	Holland, Groningen	Oosterpoort
Wednesday, February 13, 2019	Holland, Amsterdam	Paradiso
Thursday, February 14, 2019	OFF	
Friday, February 15, 2019	Germany, Hamburg	Docks
Saturday, February 16, 2019	Germany, Berlin	Huxleys
Sunday, February 17, 2019	OFF	
Monday, February 18, 2019	Brighton, UK	Concorde 2
Tuesday, February 19, 2019	Bristol, UK	Trinity
Wednesday, February 20, 2019	Manchester, UK	Gorilla
Thursday, February 21, 2019	OFF	
Friday, February 22, 2019	London, UK	Hackney Arts Centre

Friday, March 1, 2019	Boston	Royale
Saturday, March 2, 2019	NYC	Irving Plaza
Sunday, March 3, 2019	OFF	
Monday, March 4, 2019	Detroit	El Club
Tuesday, March 5, 2019	Chicago	Lincoln Hall
Wednesday, March 6, 2019	Nashville	Basement East
Thursday, March 7, 2019	OFF	
Friday, March 8, 2019	Atlanta	Terminal West
Saturday, March 9, 2019	New Orleans	House of Blues
Sunday, March 10, 2019	OFF	
Monday, March 11, 2019	OFF	
Tuesday, March 12, 2019	OFF	
Wednesday, March 13, 2019	Phoenix	The Van Buren
Thursday, March 14, 2019	OFF	
Friday, March 15, 2019	Los Angeles	Fonda
Saturday, March 16, 2019	San Francisco	August Hall
Sunday, March 17, 2019	TRAVEL - BAND/CREW HOME	
Monday, March 18, 2019	OFF	
Tuesday, March 19, 2019	University of North Texas	Winspear Performance Hall @ UNT (solo)
Wednesday, March 20, 2019	OFF - Travel Home	
Thursday, March 21, 2019	Los Angeles	USC - SCALE Conference

We imagined the following divisions of labor:

Tour Manager

- Load and unload the trailer for each show day
- Organize all travel
- Handling Food
- Coordinate all hotel and bus bookings for overnights
- Supervise show "advance" where venue and promoter agree on crew, staff, equipment and schedule for each performance
- Help with backline set up

FOH Engineer

- Organize all aspects of sound for the venues
- Tune and optimize PA system in each city
- Supervise placement and choice of microphones on stage
- Monitor Harmonizer during the performance
- Mix the show
- Create a Multi-track Recording of the show

Monitor Engineer

- Supervise all wireless RF
- Manage stage crew during show build
- Execute instrument setup, including keyboards and MIDI controllers
- Mic up performers and manage in-ear monitoring
- Mix monitors during the performance

Lighting Designer

- Execute the setup and operation of lights for the show
- Integrate the house lighting rig with our lighting console

- Assist with loading and unloading truck each day

We had to fit everything for the show, except for the PA system, in a trailer carried behind the tour bus. This required some creative packing and CAD, as we had to fit the grand piano and all audio equipment and instruments in the trailer. Because of this, the work of loading and unloading quickly became overwhelming. The piano weighed about 1200 pounds and required at least 5 people to move and set up. In certain venues, it had to be lifted onto the stage. It took almost 3 hours to get the show built every day and even after the first 30 minutes, the crew was exhausted.

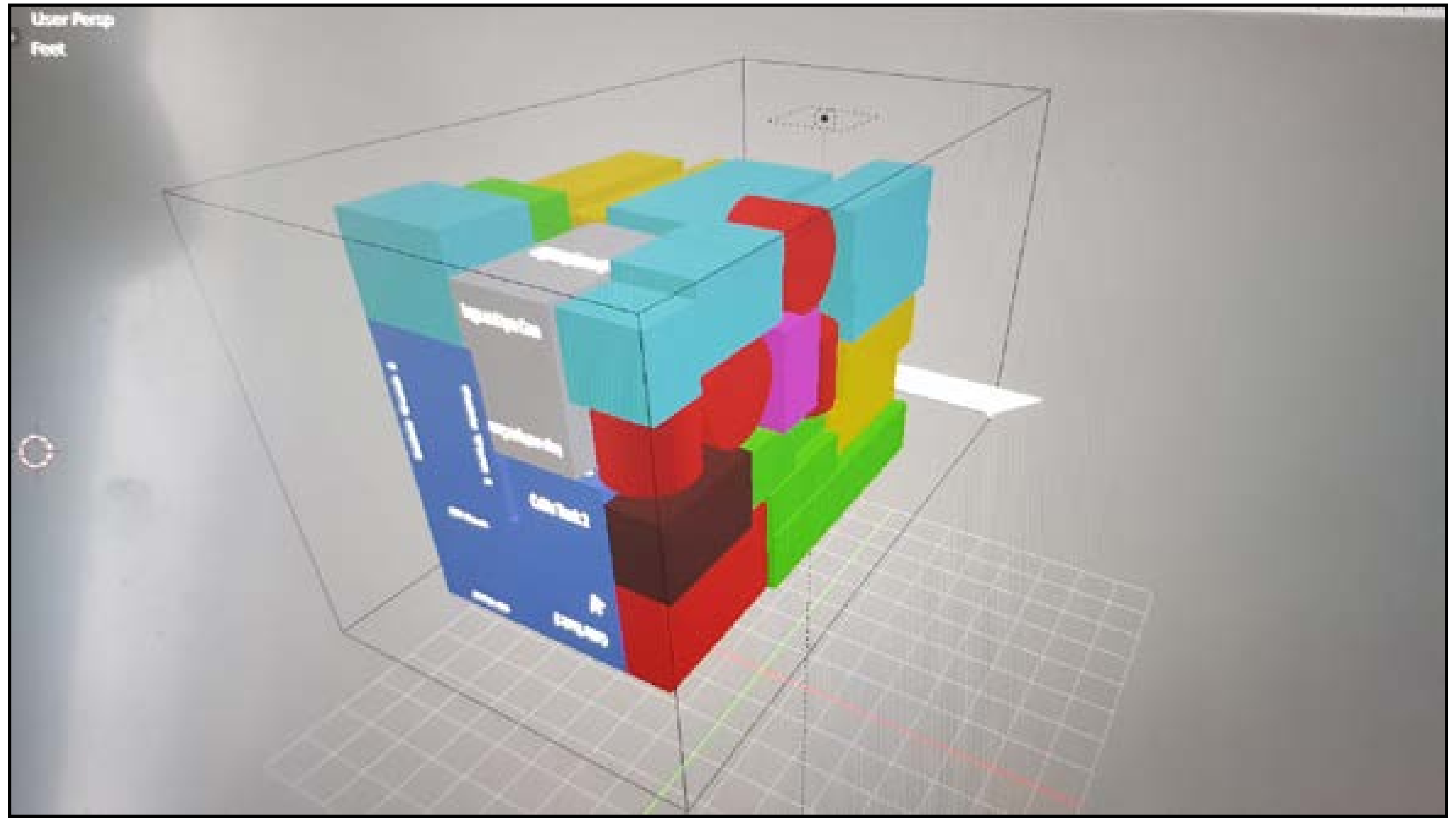


Fig. 4-23: 3D Render of Djesse tour trailer packing arrangement - credit: Rob Fisher

After six performances, we added an extra person to help better divide labor. We initially believed that having extra help with the hospitality aspects of the Tour Manager's job would free him up to focus more on loading and backline jobs. However, this proved not to be true. The packing and unpacking was too much for one person to handle in addition to administrative jobs, such as booking hotels, replying to advance images, etc. The combination of physical and mental exertion every day meant that details were being missed in both places.

We decided to add a person to help with the load, the build and the pack, and to manage the stage during the performance. This freed our tour manager up to stay in the production office during the day handling logistics for current and future shows.

4.12.1 Hardware Revision

Around this time, we began to think about logistics moving forward for future dates. While it was easy to carry everything in a trailer behind the tour bus, we would realistically need to find a way to fit everything in checked bags to make a version of the show that would be transportable by airplane. The performance venues for the summer tour were located throughout the US, Canada and Europe, with no bus or trailer. To keep back fees low, we attempted to structure our baggaged so that as many bags as possible should be under 50lbs. The trailer setup used two 6u racks, once for MIDI and one for wireless, 70 and 76lbs., respectively.

Splitting up the racks into 50lb components required the use of several additional boxes. The majority of the weight was in the case. We found that it was only possible to get 2 rack units of equipment in a case before it was 50lbs. If the components were extra light, we could do a 3u case, but that wasn't the situation for any of the gear we were trying to separate out. Breaking 12u of equipment up into 2u groups would mean a huge number of physical connections during setup.

At the same time, we were also beginning to consider possible accommodations for mixing desks. We initially started by asking venues to source either Yamaha QL or Digico mixing desks. Unfortunately, this became cost prohibitive, with over $10,000 of the tour budget being spent on mixer rentals. To combat this, I had previously spent a significant amount of time trying to condense the show onto a Yamaha QL1. The QL1 could fit in a pelican case as a checked bag under 70 lbs, but it was woefully underspec'ed to run the show. There was one other desk that could do it, with better specifications, but that was the Allen and Heath (A&H) mixing desk that ruined our BBC Showcase at SXSW (see the failures section of the One-Man Show). The interaction with

A&H had been so bad on that project— they blamed me in the moment and later called to apologize saying the problem was their hardware— that I did not feel comfortable using that system on the road. As a responsible producer of the show, I could not let anything like that happen ever again. Due to the uncertainty about both the cause of this problem and the system's reliability in general, especially given the manufacturer's response, the only way to ensure a similar situation would not recur was significant bench testing. We didn't have time for that and I was not inclined to work with the company after interacting with them at SXSW. The Behringer X32 did not have enough IO or mix channels. Although it was heavy and had limitations, the QL1 became our target for a flyable desk. Because we had chosen a desk with native Dante support, we could remove the Dante to MADI bridges from the rig, which saved weight and rack units.

Fig. 4-24: Djesse tour second revision rack design - inner cabling routing

Based on this, I began a redesign of the MIDI rack to reduce it from 6u to 2u. This included creative cable management for power and data. Since the system was purely Dante, we also needed to ensure redundancy on the network, so we added a secondary Dante switch. Photos of the redesigned rack are shown below. Brian Mayton assisted with the construction and cable management in the rack. The MOTU Timepieces were replaced with a single 4-port iConnectivity MIDI interface. We kept the Focusrite Dante 16Line interface.

Fig. 4-25: Djesse tour second revision rack design - front and rear photos

The rack was remarkably compact and fit two managed switches, MIDI, audio, network interfaces, two USB hubs and power distribution for all of that equipment and our two show MacBook Pro computers. The hole cut in the top of the case allows easy access to the rack's internal MIDI connectors to aid troubleshooting, but since this rack has been constructed, there has not been a need to troubleshoot any elements on it (with the exception of some recycled EtherCon jacks).

Several modifications were needed inorder to make the show fit on a QL1.. First, we had to extend the native Dante capability of the desk to add 16 additional channels with a Dante MY-AUD expansion card. We used the stereo return channels of the desk as input channels. We also used spare mix busses as input channels by patching Dante channels directly to the returns. There was a trade-off— returns did not have direct outputs on the desk, and mixes used as channels did not have any input section, since we were using insert returns. Matrixes became aux busses since we were using the mixes as inputs.

Fig. 4-26: Djesse tour second revision monitor desk and rack design

4.12.2 Show Documentation

Around this time, we also began to streamline the show documentation to make it easier to track all the backline and equipment needed at each venue. Since the tour required flights between each city, there needed to be a clear way for promoters to understand what was needed for the show and for our team to understand exactly what was being provided.

I redesigned the show documentation based on two sections: a document describing the ideal configuration of gear and instruments, and an interactive spreadsheet for each venue to fill out showing which items would be provided in each city. The full set of this type of documentation is included in the appendix <??>. This provided promoters with a clear understanding of all the pieces needed for the performance.

This documentation was refined even further in fall of 2019, and separated into two documents: one with basic requirements for all venues, including hospitality, and one interactive sheet for promoters to fill in with exact equipment specifications.

4.12.3 Rehearsal Process in Sound Checks

Moving to the QL1 fully Dante-based system, we held a one day rehearsal to create new mixes based on the new consoles and playback systems. From then onwards, all rehearsals for adding and changing content have been held during sound checks. This saves money, but also keeps the band fresh from day to day. This was made possible because Jacob became organized and methodical with the time spent during sound checks. .

4.12.4 Flying the Full Show

The rhythm of tour changed significantly with the switch to fly-dates. The initial tour was relatively smooth, but with over 30 checked bags, the process of getting from one city to the next quickly became unsustainable. With the crew and our tour manager, Rob Fisher, we developed a strategy for scheduling future tours to include more rest days. We also worked to eliminate many of the cases carrying backline items, such as midi controllers, the upright bass and acoustic guitars. These could be rented locally, so we did not need to travel with them. We also developed rules for ground transport, hotels, and venue safety to ensure the traveling company was well cared for and did not have to spend unnecessary energy in transitions (such as moving 30 cases up a flight of stairs because the hotel has no elevator or storage).

The first test of this new touring strategy was for an Australia tour that I was able to join (in a break from writing this document.) We planned to add a new song, and some additional elements to existing songs by rehearsing in sound checks. s Before the tour, Jacob and I met in London to do some programming, since these songs also required adding new sounds. We were able to go through and refine the existing sounds, as well. After a few months of touring, Jacob had a long list of tweaks that he wanted to complete. We also added a new looping segment in *Fascinating Rhythm* to match the old One-Man Show. That moment was an audience favorite.

From there, we went on tour for six performances in Australia and New Zealand, culminating with a performance at the Sydney Opera House. The first performances went relatively smoothly, as we slowly added the new material as it became ready in sound checks.

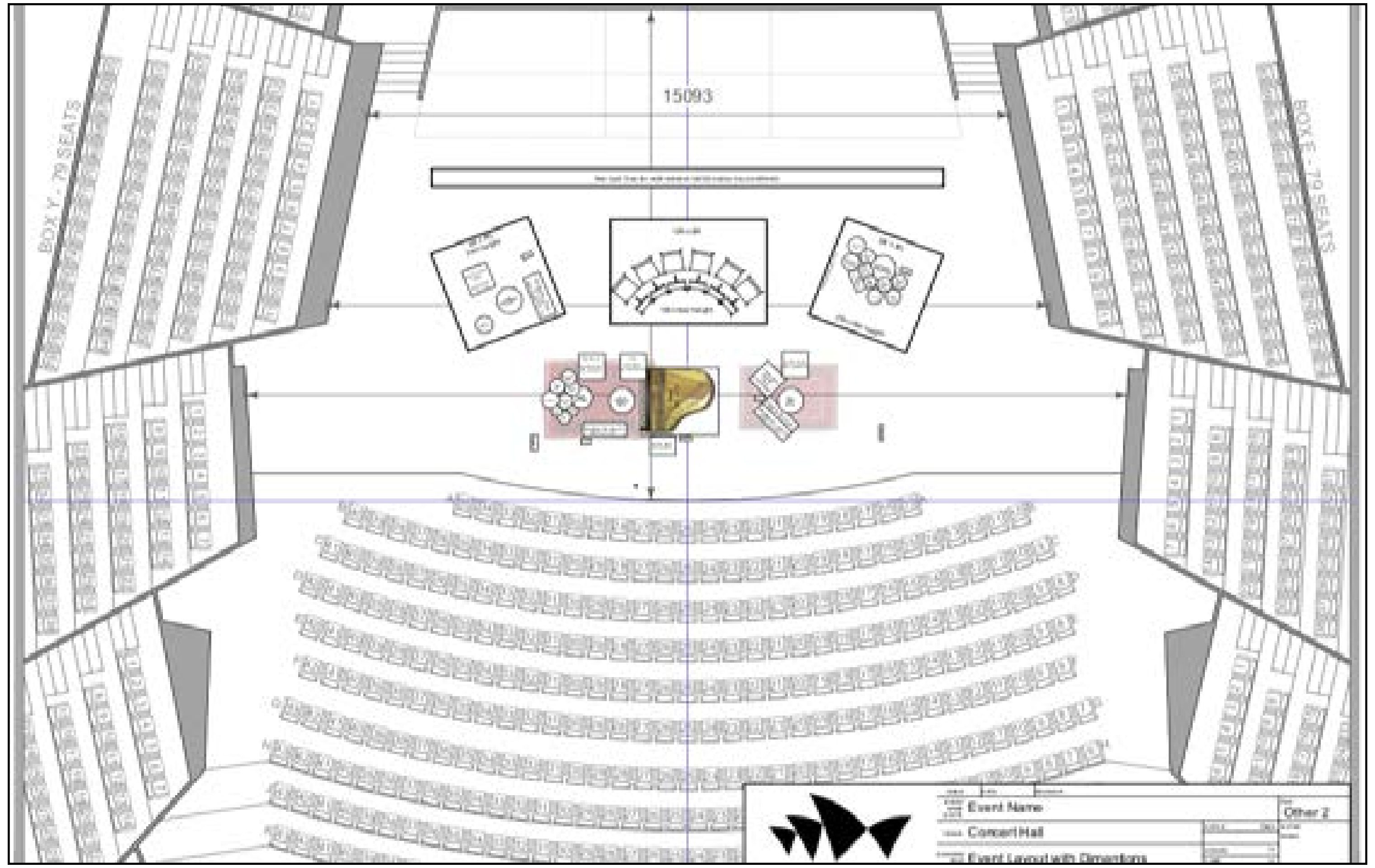
The performance at the Opera House was especially complicated. Jacob had the idea to add a string sextet, including his mother, Suzie, to play on some songs. He also wished to record the audience in more detail than usual. Since we had maxed out our Yamaha QL1s, we worked with the Sydney Opera House team to integrate our Dante System with their Dante system. (A full drawing is given on the next page.) We added microphones and IEM mixes for the strings, and four additional audience microphones. The entire concert was recorded redundantly across two Macs running Reaper. We also added 40 additional lighting fixtures.

The challenge to adding all this was that we landed in Sydney at 12:30 p.m. the day of the concert, so there was very little time to install all of the extra equipment, make sure it was functioning and run rehearsals and sound checks. I created a minute-by-minute schedule and spent my time interfacing between all the house and touring crews to try to get us up and running as soon as possible. During the sound check, I was able to work with Jose (who was mixing FOH) on a few small things I had noticed during earlier performances:

- The piano felt amazing at low volume but the high-mids could get harsh when Jacob was playing loudly. We altered the EQ on the piano and switched from a multiband compressor to a dynamic EQ to tame some of the harshness. On a full range, tonally homogeneous instrument such as piano, I am quite wary of using multiband compression. This is a strategy often used on mixes, but on single instruments there can be challenges in the crossover regions. A dynamic EQ often helps to preserve more of the natural sound of the instrument.

- We worked together on the sound of the sextet, removing high end from the string close microphones. Amplifying strings in a loud amplified setting is always a challenge, and often the microphones must be much closer to the instrument than allows for a natural sound as one would hear it standing in front of a musician playing the instrument. To make it feel closer to being natural, we have to take away high end and usually add a short reverb. Since this was a new addition to the performance, Jose and I focused on ensuring the strings felt right, despite the hurried schedule.

Fig. 4-27: Sydney Opera House stage layout



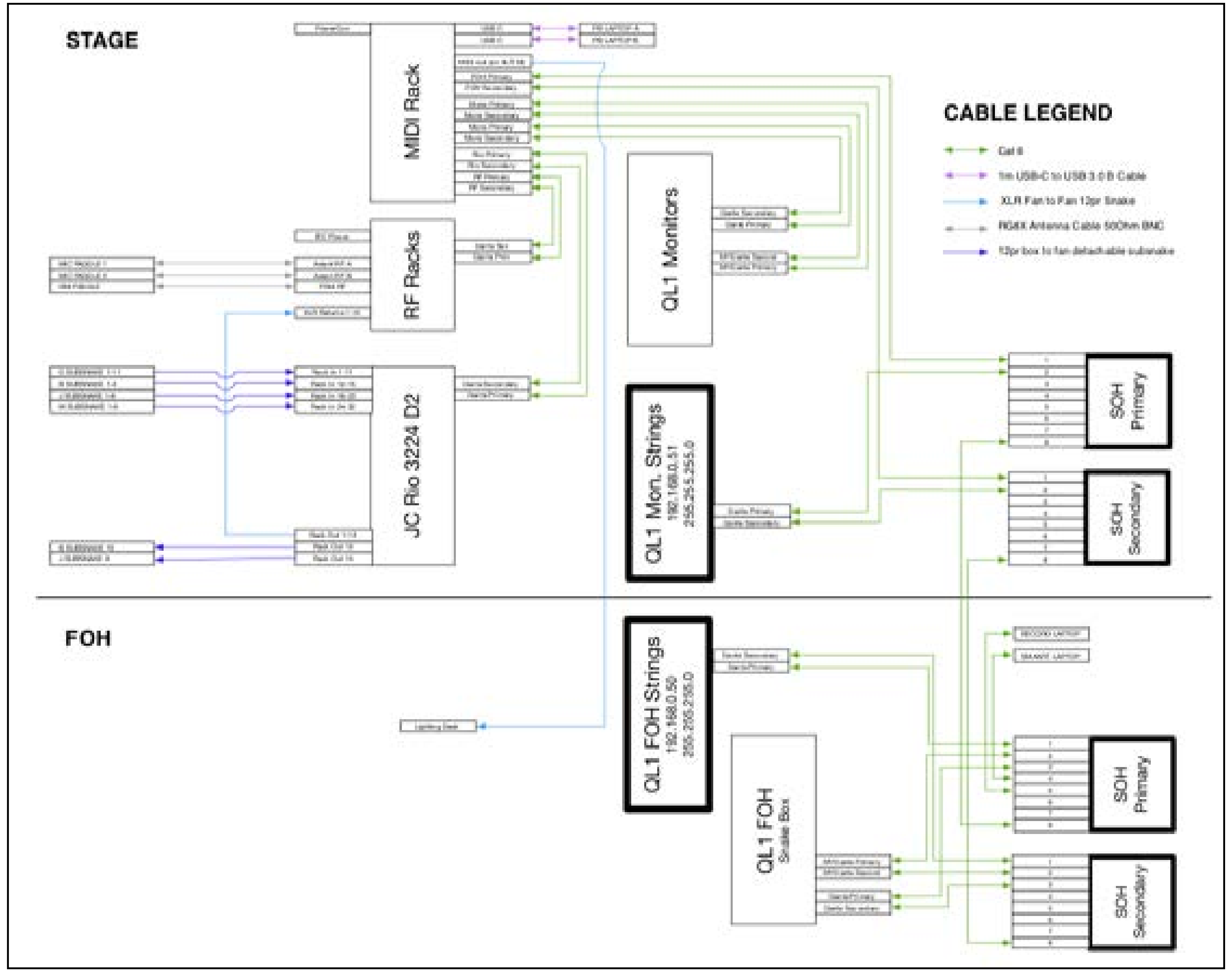


Fig. 4-28: Sydney Opera House schematic

Fig. 4-29: Sydney Opera House Performance Photos

We were strictly limited to two hours for the performance itself, so we had to be quite careful about timing. We planned the set list based on recordings from the Brisbane performance the night before including time set aside for applause and Jacob speaking between songs. We calculated the wall clock time when each song should begin so I was able to track the progress of the concert and give Jacob time updates into his ears.

In the end, both the setup and performance went amazingly smoothly. This was due to our preparation and the amazing crew at the Opera House, who were incredibly accommodating and inclined to work with us in a somewhat non-traditional system. It is exceedingly difficult to do one-off versions of the show without sacrificing the quality of the original elements because the crew often gets caught up in all the changes and forgets the tasks that happen normally. In this case, the house crew made sure all our additions worked well and our crew was able to focus on putting together an amazing show. Given all of that, we were able to keep to the schedule for both production and the show, and the performance went extremely well.

4.13 What Lies Ahead

We are still learning about the *Djesse, Vol. 1* tour and we will continue to grow it with Jacob's vision as he releases more music. We plan to revisit almost everything in January 2020 to evaluate which elements work well for future tours, and which do not. There are some technical challenges that we would definitely like to resolve:

- We are quite dependent on a very specific model of MIDI controller, which has simple patch changes that are color coded and labeled. In the future, I'd like to create a patch change module that has just the buttons on it for the patch changes that are labeled. This would allow us to use any MIDI controller and still be able to do the show consistently. Another option would be to hire in a Yamaha S90 or Motif style synth which has dedicated patch buttons.
- We have an ultimate goal that we would eventually be able to do this performance on house mixing consoles with 48 inputs or more. This is quite complex and would require us to do without some of our more complex console configuration and routing. This will need to be evaluated as we design the next iteration of touring, but I imagine that Jacob will always be an artist who needs to carry his own mixing consoles.

The performance is too detailed and nuanced to be able to use a soundcheck period to get the house gear to function properly.

- We have also considered creating a simplified version of the show, where we sample all of the synth and SPD sounds and make "static" versions of the patches that can be loaded into a Nord and SPD directly. This would eliminate the need for the MIDI rig, special controllers, etc. but it would mean a lot of the effects or behaviors such as sidechaining and exclusive voices would be simplified or impossible.
- For future performances, if the band grows in size, we will need to determine how to add additional inputs while still being able to fly the mixing desks. Currently on the QL1 this is not possible.

Beyond that, we feel incredibly positively about this project. It represents a lot of our initial creative visions, keeping the best parts of the solo show and adding more musicality. There are very few moments where the band must adhere to a form or tempo and as a result, the most amazing things happen each night. We have audience members who will come to 3 or 4 shows in a row, just because they want to understand what stays the same and where the band experiments from night to night. I am incredibly proud that we can do this with the amount of technology present. The tech allows the sound worlds to stay interesting, incredibly detailed and tightly connected to what the players are doing. At the same time, the band has become more and more adventurous as time goes on, which leads to amazing moments and improvisation. The setup makes it relatively easy to add material and songs and we've now devised a strategy to do this efficiently, while Jacob now uses consistent strategies to rehearse and tweak during sound checks.

I feel so fortunate to have been a part of this project, and I hope to continue working to help it grow and adapt, moving forward. I'd like us to be able to have more elements follow the musical intentions of the band even more closely, which will certainly be a project for the next tour.

5 Production Infrastructure for Everyday Life

I have always had a dream that eventually every part of our environment could be choreographed and designed the way we approach design for live performance. So many interactions and experiences outside of this area are aesthetically awkward or do not give us the ability to have infrastructure that supports our emotions and moods. A common example of this is the rapid proliferation of color changing LEDs and the popularity of customizable lighting. People appreciate having the ability to set the color of their lighting to their mood in their homes and workplaces. Even so, the interfaces to do this are rudimentary and poorly designed. They give users access to the direct hardware controls of the LED lighting systems, but these can often be overwhelming. Advanced systems may provide timers or scenes, but there is not a way to create dynamic looks or drive the control from advanced data. Although the application is different, the solution to the awkward control of these environmental systems may lie in the same type of technology used for interaction live production experiences. We can extrapolate and stretch the concept of Hyperproduction to encompass not only performance, but also general non-performance experiences that are embedded in the day-to-day environment around us.

Fig. 5-1: Development of Gamma music experience with Alexandra Rieger and Tod Machover. (Rieger , 2018)

5.1 Gamma Home

As part of an ongoing investigation into the effects of gamma wave stimulation[1] on the brain, we designed an in-home experience based on light and sound stimulation. The design and specific implementation of Gamma stimulation for therapy is well outside the scope of this thesis, but a basic understanding will put the following installation's objectives in context. There is a growing body of research from Ed Boyden[2] that suggests that stimulation of the brain at gamma rates, roughly 40Hz, causes neurons in the brain to fire at the same rate. This effect is called entrainment and it causes increased blood flow in the brain which is linked to a number of positive results, including initial findings that suggest the reduction in Alzheimer's symptoms.

[1] Chen, Angus. "An Hour of Light and Sound a Day Might Keep Alzheimer's at Bay." Scientific American, 14 Mar. 2019, www.scientificamerican.com/article/an-hour-of-light-and-sound-a-day-might-keep-alzheimers-at-bay/.

[2] Iaccarino, Hannah F., et al. "Gamma Frequency Entrainment Attenuates Amyloid Load and Modifies Microglia." Nature, vol. 540, no. 7632, 2016, pp. 230–235., doi:10.1038/nature20587.

Fig. 5-2: ESP8266 based LED bulb - A19 form factor with custom firmware (Rieger , 2018)

An ongoing area of research in our group by Alexandra Reiger looks at the ways these signals can be integrated into experiences that are pleasurable,[3] rather than abrasive and intense as all the experimental tests have been so far. Along these lines, we developed an artistic multimedia experience for everyday living that embeds gamma signals in light and sound around one's home. This was installed in a private home in New York to look at the effects of the stimulation on its inhabitants but also as a model of an in-home experience that is multisensory, party of everyday life and choreographed and shaped in similar fashion to Hyperproduction based multimedia experience. It was our goal to test the hypothesis that a therapeutic multimedia experience could be designed to be ever present in one's home, with certain handles used to customize the experience based on the user's mood or desires. These handles might be abstract and would control a variety of parameters rather than be connected directly to any one property of the system.

There were several restrictions for the installation: most importantly, we had to use existing cabling and infrastructure in the home. This was because the installation had to be completed quickly and without modification to the home's existing electrical or structured cabling systems. We also made the choice to utilize off the shelf products which could be modified to provide automation capability rather than developing our own hardware from scratch.

5.1.1 Lighting Implementation

To create an easily retrofitted system quickly, we found a specific model of LED light bulb based on an ESP8266 microcontroller platform with 802.11n connectivity. These were available in bulk, shipped from China or even via Amazon Prime. The bulb fit a standard E27 incandescent bulb socket, so we planned to replace many of the existing bulbs in the residence with our smart bulbs. The bulbs would communicate wirelessly and could be either standard color changing lights or capable of emitting gamma light. We created a custom firmware for the ESP8266 in each of the lights that allowed them to function this way. Rather than build a control protocol from scratch, the basis of the protocol was ArtNET which is a standard UDP based theatrical lighting protocol which works well over WiFi. This allowed the use of any modern lighting desk to design and test looks on the bulbs. Each bulb used 8 ArtNET control channels which corresponded to Red, Green, Blue and White channels for 20ms frames. The blub would switch from the first set of RGBW values to the second every 12.5 ms which resulted in a transition between colors 40 times every second. Setting the first and second sets of RGBW values to be the same would result in a solid color with no flashing. The bulbs were initially designed to be used in close proximity with a phase-locked-loop taken from the system clock synchronized by NTP. This was enough to ensure millisecond accuracy in timing between the bulbs. However, in the end not enough bulbs were placed in close proximity and this part of the system was taken out to troubleshoot a memory leak in the platform.

[3] Rieger, A. "Explorations of Multisensory Cognitive Approaches in Alzheimer's Disease: Gamma and the Senses" M.S. Thesis. MIT Media Laboratory, 2018

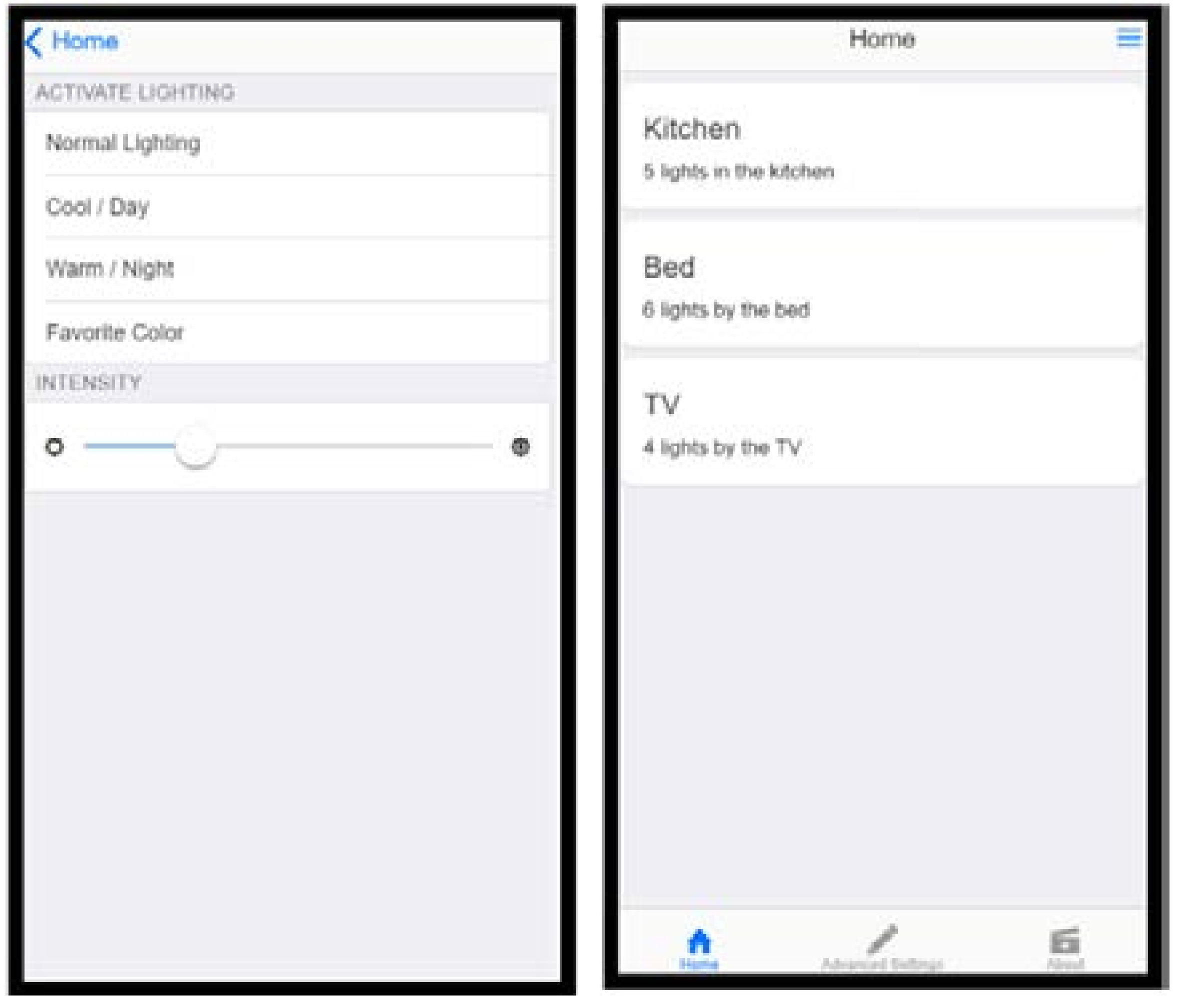


Fig. 5-3: App UI used for controlling Gamma Home system. (Rieger , 2018)

A Node.js based app hosted a web interface based on Onsen UI that allowed users to enable and disable different groups of lights using preset colors and gamma intensities. Communication between the bulbs and server was via ArtNET using the artnet[4] package, and a Ubiquiti wireless access point system installed in the residence. The bulbs were set to boot up in a standard look (2700 Kelvin amber) so that light switches in the residence could be used normally to turn lights on and off. The app was required to enable the gamma aspects of the bulbs.

Fig. 5-4: Chamsys lighting desk used to develop gamma experiences. Custom hardware for lighting desk by Brian Mayton.

[4] "Artnet." Npm, www.npmjs.com/package/artnet.

We used a magic-Q Chamsys lighting desk to experiment with looks and colors on the bulbs, these were then written into the Node.js server which could recall the values easily from the app. A continuous control could take any of the looks and control the amount of gamma effect. For the purposes of this system, that was simply scaling the second set of RGBW values.

5.1.2 Sound Implementation

We created an audio experience in the residence to accompany the lighting experience or to be used in isolation. The system was based on the resident's existing Sonos system, but augmented with professional audio components. A set of Genelec monitors and subwoofer were installed in the bedroom with a DBX DriveRack fed by a Sonos Connect. The Driverack contains an implementation of DBX's 120a subharmonic synthesizer which was inserted on the subwoofer send and set to 40Hz. A subharmonic synthesizer works by recreating fundamental frequencies for its set frequency. It does this by analyzing incoming audio for harmonic series of the chosen frequency and generating frequency content at the fundamental. So where it sees the octave, the fifth the seventh and so on, it will add the lowest note. This is useful for adding low frequency content to material which may have been produced without that content for any reason.

We then created a special playlist of songs with gamma content in them, as well as songs in the key of E (41.2Hz) and the resident was able to pick among these songs to use on the system which significantly enhanced 40Hz content in the songs. The DriveRack had a UDP control protocol for which it was planned that the original Node.js server could alter the level of subharmonic synthesis based on the preset. In practice, it was found that manipulating the overall volume of the system was effective enough.

5.1.3 Results in Practice

The system was used for several hours a day, every day of the week for over a year. Staff and residents found the system easy to operate and were able to tailor the experience to the moment each time it was used. In the lab we set up a research system to refine and improve the experience and were able to use professional production equipment— mixing desks and lighting desks— to experiment with interactive use cases and new looks. A VPN would allow us to push new content and information to the installed system as needed.

For more on this topic, please see the research of Alexandra Rieger in our group. She has specifically analyzed the use and effectiveness of the in-home experience.[5] An excerpt from Alexandra's thesis is provided here which gives an overview of the results:

"While visiting relatives recognized S.R's higher level of communication and interaction, videos were taken by caregivers at various intervals to reveal S.P's gradual changes in behavior and progression. The earliest videos of S.P feature her performing extremely repetitive motions as often seen in Alzheimer's patients [...] In most recent video recordings, more notable findings come to light. One of the tapings shows S.P remarking the fact that she feels tired and would like to go to sleep, after which she leans her head on her caregiver's shoulder and rests. Not only was she able to piece together the sentence, but she also made this remark in correlation with how she felt at the time, and what her intentions were. [...] Further footage shows S.R looking through book of paintings seeing colors and images she admires and expressing this both gesturally and verbally. This behavior is vastly different from earlier footage where she is viewing a photograph of a baby. Here she seems aware that she is looking at a book. She is also connecting with her art therapist and speaking with her about the shared subject matter." [6]

"Creating the installation in an active care setting was illuminating as we quickly learned how intertwined the experience of the care team is with the individual who has Alzheimer's. Although in-home nurses and staff are dedicated to the patient's care, schedules and routines are usually rigid. Therefore, any addition to the care-schedule must be easily accessible and intuitive. We created an app to navigate the gamma installation system easily. The app was uploaded to the mobile devices of visiting healthcare staff as well as a central iPad to control both the Sonos and gamma lights. Despite the fact that the app is self- explanatory, the extra step of adjusting the various settings to convert the lights to their gamma mode was a barrier. Due to this, several bluetooth switches were developed and will be installed."[7]

[5] Rieger, A. "Explorations of Multisensory Cognitive Approaches in Alzheimer's Disease: Gamma and the Senses" M.S. Thesis. MIT Media Laboratory, 2018 (pg. 81-91)
[6] Rieger, A. "Explorations of Multisensory Cognitive Approaches in Alzheimer's Disease: Gamma and the Senses" M.S. Thesis. MIT Media Laboratory, 2018 (pg .88-89)
[7] Rieger, A. "Explorations of Multisensory Cognitive Approaches in Alzheimer's Disease: Gamma and the Senses" M.S. Thesis. MIT Media Laboratory, 2018 (pg .90)

Alexandra makes an important point that the application user experience provided to the care staff was not sufficiently seamless to become completely integrated into the care routine. However, this field is well defined and more research could be conducted to streamline and refine the UI further.

5.2 The Future of Hyperproduction... Borrowing from Social Apps?

Indeed, the public are getting into multimedia production like never before. We can see there is a desire for this type of curation and creation through platforms like TikTok,[8] which allow users to build music driven short videos with effects, editing, performance, multimedia. Users are telling stories creatively through platforms like Instagram,[9] YouTube[10] and others. We see that there is a creative appetite for the general public to make new work. TikTok however, represents a new frontier for these tools, where there is extended timeline based multimedia creation tools built into the app. The available creation tools and therefor expressive capability with TikTok eclipses that of SnapChat,[11] Instagram, YouTube, and all other available social media platforms. It is my hope that we see this creation spill over, especially music based creation, into live experiences made by and for the public, once tools allow us to articulate our environments the same way these social media tools allow for easy creation of videos and screen based media.

We can imagine an app that takes advantage of similar UI patterns used for creative manipulation of video and audio in social media platforms, but instead allows the manipulation of infrastructure in one's environment. Perhaps little fragments of experience, or longer generative experiences could be designed and tailored and then "played" in one's home across lighting, sound and even other systems such as HVAC. This might result in mini experiences much more closely tailored to users' personal preferences or expression. Examples of these mini experiences might include the behavior of lights and sound in one's room that are intended to wake you up peacefully and gradually. Or setting the mood for when a friend is coming to visit (ex. football night, date night, family game night, dinner time, late night reading, etc...). One can easily imagine people customizing their environments across many modalities for these activities the same way they choose an Instagram filter or create a Tik Tok video with very specific creative intent.

[8] "TikTok." TikTok, tiktok.com/.
[9] "Instagram." Instagram, instagram.com/.
[10] "YouTube." YouTube, youtube.com/.
[11] "The Fastest Way to Share a Moment!" Snapchat, snapchat.com/.

5.3 Proliferation of Home Production

As the internet of things takes off, we are seeing more and more that concepts from live performance production are borrowed for the design of architectural and home control systems. However, the truly creative integration of these is still far from reality, with the best approaches found in the home automation systems.

As we make production systems more integrated and more intelligent, they require operators with more creative sensibilities and less technical knowledge. The limit of this curve results in the average user with some emotional needs being able to use one of these Hyperproduction systems to set up an environment creatively with ease and nauce. The ultimate mood lighting and sound. Or, in another scenario, a designer might come in and create a manipulatable environment with a few control parameters that allows users to access a continuum of experiences. Someone might do this for all the LED fixtures in an entire city (see Philips cloud arch. Lighting control platform) or an office building or an airplane. Through these types of designs, we are given the opportunity to explore what it means to create an experience that is present across many spaces and environments and represents our emotions in many mediums simultaneously.

6 Conclusion

During the 12 year course of the work described above, including the designs, the problem-solving, the unusual constraints and the seemingly unattainable goals, I have discovered many new ideas. This section presents those concepts formalized as methodologies that have proven to be useful for multiple projects and are unique to this body of work. The techniques and insights of these methodologies flowed from project to project. This flow of knowledge is represented by the arrows in the diagram below. The refinement and practice of the methodologies demonstrate how they can propel live production into the future.

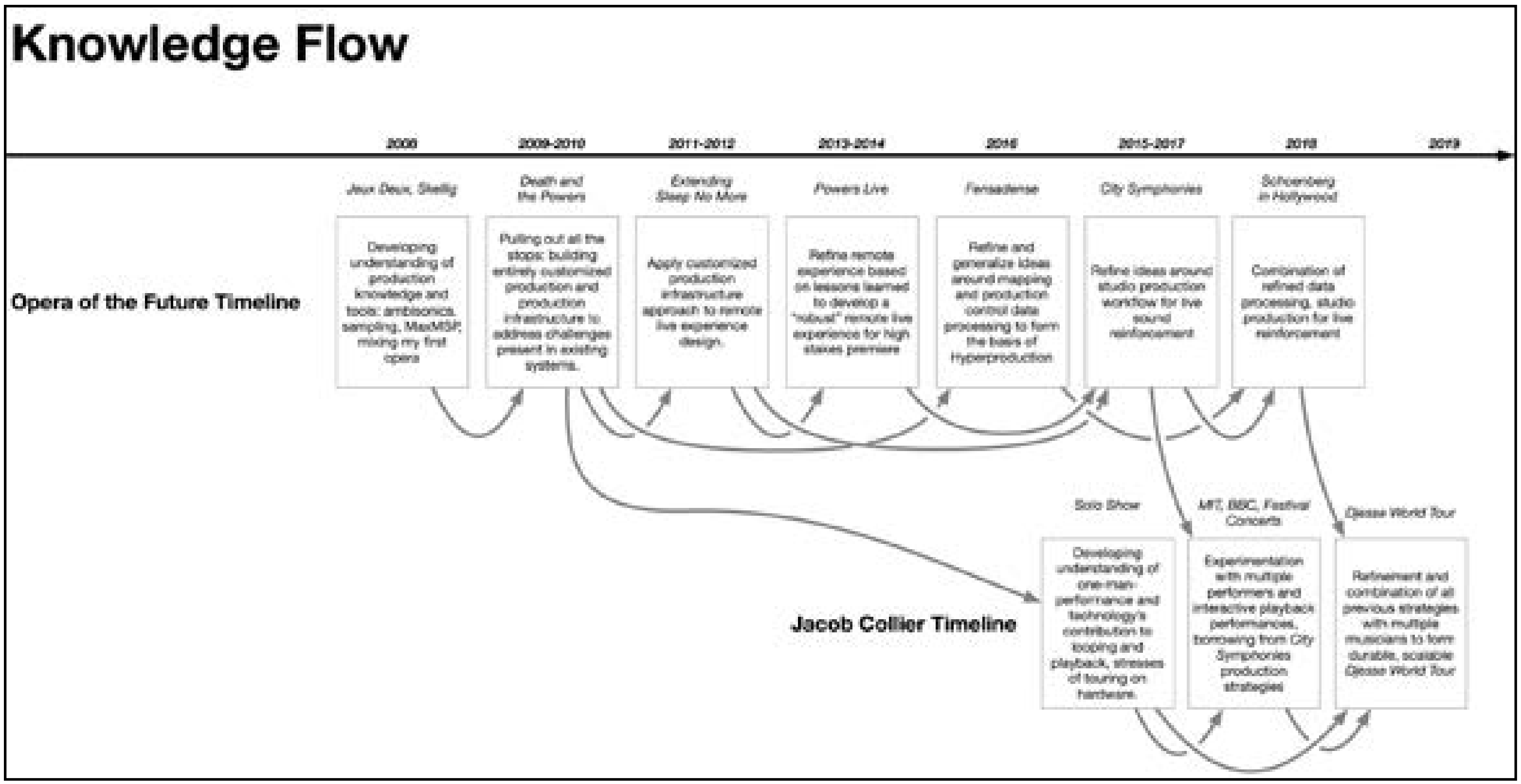


Fig. 6-1: Knowledge flow from one project to the next.

6.1 Methodologies for Hyperproduction

In this work, I have identified the following six objectives for the human-centric integration of technology in performance:

- Universal Connection of Production Systems and People
- Designing Structured Systems for Flexible and Fluid Performance
- Choreographing Live Surround Sound in Large Spaces
- Defining and Supporting Liveness
- Bringing Content Faithfully from Studio to Stage
- Designing Impactful Sound Systems for Large Ensembles and Audiences

I have established a methodology for each objective that is described in this section. The contribution of this PhD work comprises three fundamental components:

- The development of each methodology and the knowledge gained, documented in the previous chapters and summarized below
- The proof of the viability of each methodology at scale through practice in both mainstream and boundary-pushing production environments
- The refinement of and reflection on each methodology for future work

I will continue to incorporate these methodologies in my future work, but their utility extends far beyond me. By discussing these approaches and sharing techniques and tools like Hyperproduction, I envision enabling a vibrant world of experiences by a range of diverse creators. These experiences will be unlike any we have yet seen in live production and will grow from the synergy created by technologies that blend long-standing and novel practices.

6.1.1 Universal Connection of Production Systems and People

This portion of the PhD contribution encompases the work on *Fensadense* to produce a fully functional next generation version of Hyperproduction. This new version includes features and abstractions required to successfully enable measurement and mapping of multiple performers simultaneously in rehearsals and performances, and is connected to every aspect of the production infrastructure.

Knowledge Produced:
From this work I have created the following domain-specific processes and guidelines for the production of live music performances with this style of interconnection:

1. Formulating a process of developing a new production alongside technological tools to enable the production (detailed in section 3.6.2)
2. Determining important factors in developing a virtual rehearsal reproduction system for data driven interactions (detailed in section 3.6.1)
3. Important features of Hyperproduction needed for creative rehearsals that will incorporate design of live mapping and analysis (detailed in section 3.6.3)
4. Handing off a complex technical production with interconnected interactive components to a touring crew (detailed in section (3.6.4)
5. Importance of continual emotional and creative evaluation (detailed in section 4.0)
6. Including technical "performances" by operators in the analysis and shaping of a piece (detailed in section 3.7.2)

Refined and Viability Proven:
Both the new version of the Hyperproduction software, and the new domain knowledge, were successfully applied and refined during the production of six subsequent projects: the *Lucerne*[1][5], *Detroit*[1][5] and *Philadelphia*[1][4][5] *City Symphonies*, *Schoenberg in Hollywood*[3][5], and Jacob Collier's two world tours[1][4][5]. The most recent Opera of the Future project of these, *Schoenberg in Hollywood*, included a Hyperproduction implementation of a MIDI-controlled surround automation system to aid the flexibility and efficiency of production, rehearsal and performance. This enabled *Schoenberg in Hollywood* to be the first opera in the world to

integrate score-driven control of triggered multi-layer, live-encoded electronics; choreographed, rehearsed and recalled dynamically with completely musical timing.

Reflection on Past and Future Work:
Interactively controlled production technology is what allows a production to follow the people on stage and frees performers from having to follow click tracks or timecode. Every project discussed in this dissertation involves some variation of interactively controlled production technology, but the biggest lessons were learned for *Death and the Powers, Sleep No More* and *Fensadense.* These are the three productions for which we were most ambitious about building entirely new technology and systems. For *Death and the Powers,* we were able to propose and validate many approaches, such as Disembodied Performance[1] and IP-based control systems which allow intercommunication, linking every form of production technology in the show.[2] Conceptually, it allowed us to test the idea that sensor-based, human driven systems could "feel" alive.[3] We were able to take the concept of Hyperinstruments and scale it much further than before by linking lighting, sound, visuals, robotics, and scenic automation all to follow the expression of a single performer. In doing this, it also became evident that proper and frequent communication between teams was crucial to ensure the best creative collaboration. *Powers* also highlighted the importance of the operators, with both visuals and mixing desk operators "performing" to creatively shape the piece along with those on stage. It was this interaction (that I first experienced with *Skellig*) that provided the basis for the Hyperproduction software, whose goal was to include input from operators, audience and performers simultaneously to provide a Disembodied Performance-like mapping and intelligent control system for anything that might be in the environment.[4]

[1] Torpey, P. "Disembodied Performance: Abstraction of Representation in Live Theater." M.S. Thesis. MIT Media Laboratory, 2009.
[2] Jessop, E., Torpey, P., and Bloomberg, B. "Music and Technology in Death and the Powers." Proceedings of NIME. Oslo, 2011.
[3] Torpey, P. "Digital systems for live multimodal performance in Death and the Powers." International Journal of Performance Arts and Digital Media, Volume 8, Number 1. 2012.
[4] Bloomberg, B. "Hyperproduction: an audio-centric framework for the abstract modeling of live performance to guide audience attention and perspective using connected real-time systems." M.S. Thesis. MIT Media Laboratory, 2014. (pg. 83)

Both *Sleep No More* and *Fensadense* pushed the limit of logistical and operational complexity, with more lines of code and systems created in a shorter time than any other project. The overall scale of both productions was much smaller, but the ideas from *Powers* were ultimately pushed and scaled larger, with completely individualized production articulations (i.e. the sound and video mix, storyline and interactions) for every single audience member in *Sleep No More*, and the measurement of ten performers in *Fensadense* (as opposed to just one in *Powers)* to control an entire production environment. A side effect of working so quickly was a lack of time for creative iteration and development as well as for debugging. These productions were ultimately successful, but fraught with many challenges and incredible intensity. We were able to validate many conceptual ideas, investigate the sense of connection between a remote and local audience, and develop the ability to make a reasonable interaction (see the trinity of bad experiences in Chapter 3) for a large number of performers given the same type of measurement device. However, most significantly, I learned that it is *incredibly* difficult to make a piece with new technology and interactions as nuanced as in *Powers* in such a short amount of time. For both productions, we realistically spent significantly more time refining the technology (since it was all new) than refining the "feel" of the experience.

For Jacob's live tours, I was able to use the lessons learned from the implementation of technology, personnel and performance measurement strategies for *Powers,* software development for *Fensadense* and the production design for *Sleep No More* to determine which technologies and interactions to borrow and which to create from scratch. We still had the goal of bringing the best creative concepts from the existing projects, but I now had the experience to know what was easy and what was more difficult, especially given the timeframe. Significant technical components of Jacob's project, such as triggers, basic audio analysis, and universally communicating production systems (via Open Sound Control), were based on a blueprint from systems I created for the earlier productions. We had a clearly articulated sense that moving to completely click-free production would allow the performers to be more musical, and I had all the knowledge to integrate small fragments of audio and interaction into the set, knowing what would feel good and what might not. Most importantly, we knew that we had to continually evaluate the "feel" of the experience for both Jacob and his audiences, and to be wary of using the technology to do anything that wasn't serving that purpose.

6.1.2 Designing Structured Systems for Flexible and Fluid Performance

This portion of the PhD contribution encompasses work to design performance systems that allow an artist to appear to engage in improvisation within the confines of a carefully structured architecture. This goal has been developed and refined for every Opera of the Future production and has been brought to the mainstream through Jacob Collier's touring and music.

Knowledge Produced:
From this work I have formed the following domain-specific processes and guidelines for the production of live music performances with this quality of experience and organization:

1. Organizing timed and free electronic material so that it flows together smoothly (detailed in section 4.4)
2. Arranging performances and performance numbers to contain seamless transitions between many categories of content: live performed material, recorded materials, pre-recorded materials, synthesized materials. (detailed in section 4.2)
3. Crafting digital systems to implement materials and content with specific feeling (detailed in section 4.0)
4. Crafting digital systems to feel uncrafted or humanized (detailed in section 3.3.3)

Refined and Viability Proven:
With both Tod's productions and Jacob Collier's *One-Man Show* and *Djesse* World Tour, I have demonstrated that these techniques are not just for contemporary classical music, but can be applied in many styles of music. They can also accommodate the rigorous requirements of both a bus tour and a fly tour using the airlines to get to each venue. These techniques can clearly be used successfully with future large-scale professional productions.

Reflection on Past and Future Work:
An important theme emerged in the design of these productions, starting with *Jeux Deux*. It is central to Tod's work, and I have helped Jacob incorporate it into his performances as well. We must be intentional about how we craft interactions and electronics so they feel organic and improvisational. Electronics do not know how to "feel" so it is important to try to break down the feeling we intend to communicate to basic components, to

unpack it so that we can give very specific instructions to both people and systems. Often the most seamless, flowing experiences are the most carefully planned. *Sleep No More* is an excellent example of this: the audience experiences something that feels natural and improvised, when in fact every second is carefully choreographed to "feel" natural and improvised. The "shadow line" in Tod's electronics is another example where the fluidity and the variation are carefully sculpted to feel natural, as if several musicians were improvising together.

Jacob's live performance is designed so that audiences feel this sense of natural movement between a combination of playback, synthesized sounds, looped live recordings, and acoustic instrumental performance. We very carefully structure every single element so that transitions are not jarring and it can be difficult to tell which elements are which. This keeps the technological aspects of the performance subtle and nuanced and actually allows him to improvise on top of them. This specific style of structure allows for freedom even when the timing is rigid. As many viewers have stated to me, the degree of intertwinement of elements in Jacob's performances is unparalleled for a live tour that has reached such a large number of people and that is supported by popular music industry representatives. Through this work, I have demonstrated that these techniques are not just for contemporary classical music, but can be applied in many styles and with the rigorous requirements of a bus or airplane tour. These techniques are viable to consider for the future of large-scale professional production. Furthermore, this work proposes that we are only beginning to see how these techniques will enable a new future of multimedia, multisensory experiences by applying these techniques at scale throughout the world.

If there is one aspect of production where machine learning could have an application, I think it is creating these types of structures and smooth transitions that feel organic, but are not overt and obvious. It may be possible to spend less time carefully crafting these structures by providing a learning algorithm with many examples and letting it create them with a Generative Adversarial Network. This is an interesting area for future research. It remains to be seen if having such a structure generated by an AI would enable the same amount of freedom for the performers. Jacob's complete understanding of the structure and capabilities of the live systems gives him the ability to improvise on top of them. A system that is somewhat non-deterministic might need to approach human levels of musical communication before it would work well in a similar scenario.

6.1.3 Choreographing Live Surround Sound in Large Spaces

This portion of the PhD contribution encompasses work beginning with the integration of lessons learned from *Death and the Powers* and extending through *Fensadense* and *Schoenberg in Hollywood* to formulate a working knowledge about multi-channel live mixes.

Knowledge Produced:
Formation of domain-specific processes and guidelines for the production of live music performances with surround sound:

1. Concepts that separate group surround sound and traditional sound design (detailed in section 3.3.6)
2. Mixing desk architecture for controlling large surround sound systems (detailed in section 3.3.2)
3. Strategy for controlling surround sound based on musical triggers (detailed in section 3.8.1)

Refined and Viability Proven:
This workflow and aesthetic was included in discussions with d&b about their Soundscape product in 2015. The device was officially released to the public in 2018 incorporating certain features that were used to influence studio workflow and production practices for Opera of the Future and Jacob Collier productions in 2018 and 2019. A customized Hyperproduction automation interface to Soundscape was created to facilitate dynamically controlled score driven surround sound choreographies. This was based on the aesthetic principles that sources should be able to be mixed and shaped in real time before being encoded as surround sources. We pioneered this technique through the work for *Philadelphia Voices* and enhanced and formalized it with the addition of score-based automatic triggered movement for *Schoenberg In Hollywood*.

Reflection on Past and Future Work:
This investigation began with my experiments with ISCT Ambisonics. Hearing the results of that experiment in *Skellig* led us to try and scale the approach significantly for *Death and the Powers*. With *Powers,* the compelling aspects of large audience surround sound began to clash with some of the basic rules of traditional sound design. I had to think carefully about which rules to preserve (ex. time alignment of all speakers) and which to throw out (ex. all audience members should experience exactly the same sound). With *Fensadense* and the *City Symphonies* I was able to experiment with commercial products and start to develop a sense of aesthetics for multichannel sound design. Rather than integrate certain triggers or effects, as was done for *Powers* and *Skellig*, several of these projects were mixed entirely in multi-channel sound (some surround, some with multi-channel frontal system). This involved evaluating the creative impact of concepts such as "binaural unmasking" and placing live sources dynamically. After the performance of *Fensadense* in particular, we agreed that the multi-channel system used (and the mix for it) was not a significant contributor to the artistic impact of the piece. If *Fensadense* is ever done again, it will probably be performed with a stereo sound system.

Developing the surround and multi-channel systems required new infrastructure to adapt some of the workflow and tools we'd created for the studio and translate them to be useful in a rehearsal and live production environment. Many different types of control systems and user interfaces were tested, ranging from iPads, to keyboard and mouse, joysticks, touch screens and mixing desks encoders. For many of the productions, we were not able to choreograph movement at all because it was just too cumbersome to program and recall so many trajectories, or it was too hard to move sources manually while mixing.

All of this was taken into account during the production development for *Schoenberg*, and using Hyperproduction we created a system to audition, store, recall and automate trajectories and movements based on triggers from the orchestra. This enabled continuous movement and choreography of 64 live and prerecorded elements in a combined frontal multi-channel and surround system, which had not been previously achieved live. Because we were able to have a seamless and agile workflow, it allowed us to get the most detailed sense so far of surround aesthetics. The properties mentioned in Chapter 2 were important to define the feel of the acoustic and electronic sounds in the sonic environment of the piece. The triggering of the choreography was entirely based on the performance of the orchestra, not requiring any specific operator or oversight. This also freed the mixing engineers to focus on the musical balance of the piece, rather than moving multiple sources around. It will be exciting to work on future productions with this type of workflow, and I think we now have a platform on which it will be possible to create music driven performances with the capability to focus on creative mixing and choreography rather than the mechanics of organizing the technology.

6.1.4 Defining and Supporting Liveness Traditional,forma

This portion of the PhD contribution encompasses work to incorporate *Powers Live* internet compatible infrastructure and communications protocols to Hyperproduction, to enable the use of Hyperproduction for personal production experiences. It also includes the hardware and software design for the Gamma Home personal production system that was installed in a New York City residence.

Knowledge Produced:
Formation of domain-specific processes and guidelines for creating interactive remote and personal production experiences:

1. Development of drop-in compatible LED bulb for home fixtures that speak industry standard lighting protocol (section 5.1.1)
2. Creation of server and web application with a general user interface for operating production systems by non-technical operators (section 5.1.1)
3. Workflow for moving from professional production hardware to end user controlled infrastructure (5.1.1)

Reflection on Past and Future Work:
With *Sleep No More* we first investigated the concept of "personal production" where each audience member was given their own customized version of an experience. This included personal binaural audio mixes using triggers with a separate sampler for every audience member, live and pre recorded video streaming with the switch controlled for every audience member independently, telephone interactions, automation of the story, and script interaction. This system worked but we discovered the biggest challenge was in linking those experiencing the show online with those experiencing it while physically present. We wanted to foster a sense of connection because we thought this was one of the most important parts of live, real-time experiences. With all of the bells and whistles, we had trouble helping participants understand what was happening and there was not a sense that the performance was actually taking place live with real human performers.

For *Powers Live*, we designed the experience with this in mind and spent a lot of time iterating on the various interactions and the role of the mobile experience. We carefully chose moments where the mobile screens would go dark or would try to distract from the main performance. Since this remote experience was scaled to thousands, we developed methods of algorithmically defining the individualized experiences for each mobile device. Every remote audience member was provided a customized experience, but the variation was defined with probabilities rather than explicit definition as used in *Sleep No More*. This difference was due in part to the universal timeline, since all members were watching the opera via satellite link. For the final interaction where remote audiences could control the live experience and the Dallas audience was supposed to be aware of the presence of the remote users, we struggled, but ultimately I do believe this experience improved (if only slightly) on the sense of connection between live and remote audiences. This is remarkable given the relative size of audiences for *Sleep No More* and *Powers Live*.

Furthermore, this work paves the way for a new type of production, where experiences are personalized for us no matter where we may be. The work in the Gamma Home is a very basic example of what may be possible for the future: using the internet to link infrastructure across our environments and to design every-day life experiences in a similar way to how we think about live production of performances. Even better, as user experience for production designs systems improve, borrowing from creative tools already used by the public we can imagine audiences having more than just cursory impact on these personal production experiences. You might send an environment choreography to your mom for her birthday, or pick from your list of favorite environments, designed by you or your friends, to wake you up in the morning. I hope to continue to explore the idea that every device can be a platform for a Hyperproduction, but also to consider the importance of how such an application of technology impacts the sense that a performance is live and supports the qualities discussed in section 2.1.2.

6.1.5 Bringing Content Faithfully from Studio to Stage

This portion of the PhD contribution encompases work to redesign Tod's home studio, as well as the production of Jacob Collier's *In My Room* album at Remote Control Productions in LA and the following redesign of his home studio (not covered in this dissertation). Following the studio changes, the contribution encompasses the development of workflow for both Machover and Collier to bring studio creations to the stage in a flexible, shapeable, musically timed fashion.

Knowledge Produced:
Formation of domain-specific processes and guidelines for the production of live music performances based on compositions designed and refined in a studio rather than specifically for live performers:

1. A process and workflow for multi-channel triggers, preserving both the original composition balances and the ability for real-time mixing and spatialization. (detailed in section 3.7.3)
2. Strategies for moving sound direction from album/studio sessions to live automation systems for keyboards, looping and playback (detailed in section 14.12.1)
3. Thought-process for taking work produced for Youtube/Album, determining what elements are important to preserve on stage and how to preserve them, including:
    - ◊ The creation of customized instruments such as the harmonizer (detailed in section 4.1, 4.3, 4.6)
    - ◊ The creation of customized looping systems (detailed in section 4.4)
    - ◊ The creation of universally connected keyboard and drum automation to trigger many kinds of sounds intelligently (detailed in section 4.12.1, 3.8.2)

Refined and Viability Proven:
This knowledge was refined and proven to be viable through the production of six projects: Lucerne, Detroit, Philadelphia City Symphones, Schoenberg in Hollywood, and Jacob's two world tours. Each of these projects involved a new approach based on previous attempts to translate material produced in the studio to be used in live production systems for performance. These approaches are detailed in the respective sections. *Schoenberg in Hollywood* and the *Djesse* World Tour represent the final implementations of this technique with multi-layer, dynamically spatialized, automatically choreographed triggers used for *Schoenberg in Hollywood* and combined looping, triggered, multi-layered samples and direct-from-album keyboard sounds in a unified interactive system for the *Djesse* World Tour. Both approaches are novel and have not been used before in live production.

Reflection on Past and Future Work:
Starting with Skellig, the concept of triggers has been a constant in all of these projects, simply because it is the most straightforward way to allow technical systems to follow humans. Until *Fensadense*, the triggers did not evolve and remained stereo recordings (the exception was that *Powers* did have 6 multichannel textures to go

along with the stereo triggers). For that period of time we created triggers in the same way as well. Recording them in Tod's studio in stereo through the Yamaha 02R mixing console, which could only record digitally in stereo to Tod's studio computer. This involved trying to imagine (as we had with Ambisonics) the "feel" of the triggers in a much larger room, and mixing them down in the studio. Tod's studio was actually a good environment for this because it was relatively large and the speakers are arranged to fill the space diffusely, which is common in larger venues. In the studio at the Media Lab, it was much harder to imagine the feel of something in a large room while creating it in a small room. This is one of the reasons I became so interested in Ambisonics: it allowed much more consistent results moving from a small to large space. Still, once we had committed to a mix in the studio, we were stuck with that mix in the larger venue. Many times I would have to play tricks with EQ or compression on the stereo mix to try and rebalance a trigger. Occasionally after getting to the hall, we would return to the studio and re-record certain triggers.

When redesigning Tod's studio, I took into account how we'd been using triggers and how we had been producing performances, and devised a new workflow that would allow us to record multi-channel triggers, providing much more flexibility when we arrived at the performance venue. In addition, the added ability to shape more layers of triggered electronics helped them to feel more natural and connected to the players on stage. At first these layers were simply mixed down to stereo live, but for *Philadelphia Voices* and *Schoenberg* these elements were spatialized. For *Philadelphia Voices* the spatialization was static to increase the resolution of the sonic image. With Schoenberg we had the best of all worlds, being able to mix the layers of the triggers to follow the musical energy of the performers, while having all the layers spatialized dynamically following choreography triggered by the same MIDI playing the layers. This resulted in many layers changing in position and volume, directly influenced by the performance of the music on stage.

For Jacob's one-man-show, moving from the studio to the stage took on a new meaning because we were trying to develop fragments of recordings that perfectly lined up and complemented loops. This involved choosing which elements to include in the recordings and which he would play live. It also involved thinking about the quality of the live recordings and the loops and trying to match them so the recordings would not feel out of place. For the *Djesse* tour, we preserved this style of pulling fragments from the album, but we augmented it by having the fragments be triggered (just like in *Skellig*) rather than played automatically by the looping system. This enabled the band to improvise, always knowing which elements they could trigger at any time. Rather than using Ableton to organize the triggers and looping, we added MainStage, which was able to run channel strips imported directly from Jacob's album session. This allowed us to pick and choose exact elements of the album to bring to the live performance in an interactive way, preserving all editing and routing capabilities.

6.1.6 Designing Sound Systems for Large Ensembles and Audiences

This portion of the PhD contribution encompasses work to develop large audio systems for all projects starting with *Fensadense*. This often includes selling the concept of advanced systems to venue and production outfits that may believe such an approach is "overkill." This therefore requires developing systems that show very clear improvement over typical approaches, and includes selection of speakers, processing, control, mixing consoles, automation systems and other decisions and designs which are completed for every production.

Knowledge Produced:
Formation of domain-specific processes and guidelines for the production of live music performances based on compositions designed and refined in a studio rather than specifically for live performers:

1. Formulation of guidelines and recommendations for spatial resolution required in a creative piece with many sources or layers (detailed in section 3.7.3)
2. Formalizing the concept of the shadow line and active mixing (detailed in section 3.6.4)
3. Working recommendations for how to choose hardware for high quality audio reproduction systems (detailed in section 3.7)
4. Strategies for defense of advanced systems in spaces that may not be originally suited for them (detailed in section 3.7.3)
5. Determining mixing console architecture required for advanced performances (detailed in section 3.3.1)
6. Interactive rider format specifying equipment for touring productions (detailed in section 4.5)

Refined and Viability Proven:
This technique was refined through the production of live performance projects from *Fensadense* onward. Each project had unique audio requirements and care had to be taken to design systems to satisfy these requirements, then communicate the importance of the design to suppliers, producers and venues and work to determine the

best method of installing the system. The system design for Jacob Collier's MIT Concerts represents work that involved significant justification and investigation. The lessons from this and from *Symphony in D* were used to aid in the justification, design and planning of the installation of a 40 channel advanced spatialization system at Carnegie Hall for *Philadelphia Voices*, and the refinement of that system design for inclusion in *Schoenberg in Hollywood*. Simultaneously, the interactive rider format originally developed for the Jacob Collier's One-Man show was refined and improved for the *Djesse* World Tour, to ensure high quality production in hundreds of venues around the world. These strategies

Reflection on Past and Future Work:
Even with interactive technologies and careful production design, I believe that a major component of the "feel" of these productions is contributed by the quality of the sound environment. This comes down to the selection of specific hardware and the workflow used to create content and mix live performances. There is a certain amount of science and technical understanding needed, which I was able to develop during many of the earlier projects and hope to keep developing into the future. This started with *Skellig,* where I had my first experience with high-end professional audio equipment, and continued with *Powers,* which was the most complex audio system I ever created. It wasn't until the *City Symphonies* that I had the chance to really explore the many capabilities and properties of speakers and their effects on the feel of a piece.

I believe it is important to react emotionally to mixes and equipment. Just because the spectrum analyzer says it sounds flat does not mean it "feels" good. I have learned the most about this from both Tod and Jacob, who have incredible ears and hear music, texture and balance with more weight than sound quality. The hardware used for a performance (or to mix an album for that matter) must satisfy the musical intentions of the artist first and foremost. In many scenarios, I will ask the artist how they want to feel. I will then try to design hardware that will make me feel that way. Then we are able to iterate on the exact details and I get a sense of how my feelings differ from the artist. An important part of this is having enough capability in one's equipment to provide enough scale, through volume and impact, where its needed. At the same time, with scale comes the loss of fine details, so it is important to carefully consider the space, the desired feel, and the content itself. With the proliferation of systems such as Soundscape that allow more resolution for fine detail at tremendous scale, these problems are beginning to be less of a concern, but we must still be careful to consider the quality and implementation of the production infrastructure itself in the evaluation of an experience.

6.3 Creative Evaluation of new Hyperproductions

The objectives in the previous sections provide insight into how and why to build new Hyperproductions, and a documentation of methods resulting in compelling work. However, they also provide insight into several creative qualities that are important to consider when building performances with humans and technology. While these are implied in previous chapters through the process and decisions that were made, we shall state them explicitly here and briefly explore their application to new work.

6.3.1 Analysis through Questions

We may find it tempting to try to create a rubric that can be used to "score" projects on creative impact, but such an analysis would be ineffective. I hope it is clear after reading about the process of completing the projects in previous chapters that every project is drastically different and no single set of guidelines or requirements could ensure high quality work.

Instead, I will simply list a series of questions that I feel are important to ask and consider— they are not exhaustive, but merely represent the most important creative questions from the projects described in this dissertation. These questions form an interesting parameter space, since many imply continua. I will visualize some interesting contrasts and reflect on creative choices that were made for a selection of projects. I hope that future creators may find this useful in guiding the creative conception, development and evaluation of new creative works involving technology.

Does the work effectively utilize the qualities of humans and machines?

The systems described in this dissertation all involve the combination of humans and machines. The most compelling collaborations of machine and human are those where natural tendencies of both are enhanced by one another.

Fensadense (3.6) and *Death and the Powers* (3.3) represent productions that take advantage of the best aspects of machines and humans. While both are ultimately driven by human timing, technology (in the mapping system) adds some intelligence, and it provides a machine interpretation of human behavior accessible to the technological systems. This enhances the ability of the human performers to express themselves, and simultaneously adds nuance to the behavior of the technology. The balance enhances the creative impact and capability of both the technology and the human performance.

The City Symphonies are human-driven, with very simple electronic interactions using triggers. *Schoenberg in Hollywood* (3.8) systems were trigger based as well. In these cases, the human performers use any timing and the electronics follow along. Without a mapping system, the content of the triggers is not altered based on human performance. This is another interesting combination: once the performers tell the machines when to start, the machines then dictate the timing required of the performers. By increasing the number of trigger points and carefully tailoring the content of the triggers, this can result in much more flexibility than if there were only a single trigger. However, because the triggers are created ahead of time, they can be quite complex and detailed without every component needing to be fashioned live in the moment. The creative result is incredible synchronization and high production value along with human-centric operation.

Jacob Collier's *One-Man Show* (4.8) was machine-driven, since Jacob had to follow the timing and structure dictated by the computer very carefully. And yet, in this particular case, his ability to follow the machine is very highly developed, so it is still a fruitful collaboration. The inflexible timing allows the computer to perform more advanced behaviors such as looping, which in turn allow Jacob to focus on micro-improvisations within the structure provided. The performers in *Sleep No More* are a similar example of this. They are able to perfectly follow the second-by-second choreography and still improvise and create personal connections with the audience. However, the act of translating Jacob's *One-Man Show* to the *Djesse* tour with a full band — removing the structured timing and letting the band control technology with spontaneity and flexibility — ultimately freed Jacob to be more musical and creative. This could possibly provide a blueprint for a future iteration of *Sleep No More*, where the cast controls the production infrastructure rather than vice versa.

For new work, this balance and arrangement is worth intentional consideration. In the projects described, we have seen three examples of different arrangements of human and machine collaboration. When developing a new performance or experience, we should carefully think about exactly which roles humans and machines will play and how they affect one another and the experience.

Does the work relate to deep inward feelings and at the same time engage our intellect?

When experiencing a performance the audience may have a range of reactions. Although this categorization is entirely subjective, there are two types of reaction that I believe are important to consider and balance:

Intellectual reactions relate the experience to existing knowledge or the pursuit of information. Examples reactions might include:

- "How does Jacocb's harmonizer work?" / "How is he doing that?"
  - we are trying to figure out what something is or how it's being achieved
- "This performer has incredible technique!"
  - we must already know what technique means and have some ranking of talent in our minds
- "What is the connection between the sound and the visuals?"
  - we are trying to discover if there is a link between two elements
- "This is an impressively difficult time signature to play live!"
  - we understand what time signatures are and what it means to play them
- "This reminds me of Boulez."
  - we must know who Boulez is and what his music sounds like
- "I wish the sound guy would turn up the lead vocal!"
  - we are focused on the balance of the sound, and must understand that someone is working on it
- "The oboe is flat"
  - we are focused on a piece of the full sound and understand that it feels out-of-place or wrong
- "Where is that sound from?"
  - we are trying to recognize the origin of a piece of the performance

Emotional reactions are more difficult to fully define, but are more basic in nature and often relate to the feeling of the performance or personal emotions of the audience without significant external influence:

- "I feel on edge."
- "I have goosebumps."

- "I can't stop smiling!"
- "This guy is really baring his soul for us."
- "I am angry!"
- "This is so soothing and warm!"
- "I want to hug everyone here!"
- "Wow, we've all been through a really special moment together."

Traditionally, audience members would primarily react to content: the melody line of the music, the mannerisms of the performer, the arc of the piece, the lyrics to a song, etc. However, when it comes to building performances with technology, we design our environments to enhance or even directly contribute to the content and therefore change the perception of the audience. This can have an impact on the feeling of the performance and the reactions of the audience.

In the production of the projects in this dissertation, we spent significant time considering the intellectual and emotional stimulus of our technology choices and their impact on the overall feeling of each piece. To be clear, we did not want to prescribe specific audience reactions or even plan them, but we did want to help the feeling of the production technology support the artist's intentions with the content. A conspicuous and convoluted system might result in the audience spending more time thinking about the technology than the music, and this is rarely a desired outcome.

For example, *Schoenberg in Hollywood* (3.8) is an incredibly intellectual piece, not only the music but the narrative as well. The piece asks complex questions about the creation and consumption of artistic works. The music, although very Machovarian, also quotes Schoenberg himself and requires active listening. The text moves quickly and uses double entendres and metaphors. The music combines many moving layers of electronic and acoustic sounds that have huge emotional impact but are also quite nuanced and deep (3.8.2). The production involves projected video, and the relationship of the video projections to the live performers is continually shifting; the audience members are at times viewing the video content, at other times reacting and acting as if they were inside it, and at other times using it as dressing for the stage. The combination of all of these properties of the piece engages one's mind quite substantially. At the same time, the narrative themes of the piece are deeply personal, and the music is impactful and wide-ranging in texture and style. This appeals to one's emotions as well.

Death and the Powers had significantly more technology running behind the scenes, but the majority of the complexity of that technology is not apparent to the audience. Rather than showcasing the technology directly, we use it to provide a unique quality to the experience: technology behaving as if it were human. This is intuitively sensed by the audience and supports the narrative of the piece, since the main character is a technological being that was formerly human. During the process of designing *Powers* it was a very intentional decision that we did not want the technology to be featured explicitly— it should fade into the background and become a storytelling tool. The very existence and completely fluid integration of the robots, visuals, and surround sound were enough to immerse the audience in the world of the piece. This supports but does not overshadow the cast, as each character disappears into the system. In the end, the piece is intensely emotional, with Miranda having to make a very difficult decision as to whether or not to remain human.

On the other end of the spectrum, Jacob Collier's One-Man Show (4.2) is emotionally engaging, but it also engages the curiosity and intellect of audiences as they try to determine exactly how Jacob's actions connect to what they hear (4.8). This discovery is aided by the combination of looping and pre-recorded fragments of audio, which are crafted to sound like one another. By the end of the opening number, the interactions become familiar and the audience learns the "rules" the show. At that point, they can appreciate the emotional nature of the music. As the performance progresses we hope the audience will take the technology for granted and focus on what it enables Jacob to do musically on stage.

For *Fensadense*, we actually gave a demo of the technology before the performance to intentionally familiarize the audience with the interactions that were possible. Once they understood the "building blocks" of the piece, they were free to focus on listening and considering the contrasts of the piece as described in section 3.6.

The production for each of these projects treats the intellectual component of the work differently: *Schoenberg's* technology is entirely subliminal, letting the narrative and music engage the intellect of the audience. *Powers*'s technology fully informs the underlying fabric of the piece, yet is never front and center, and the narrative grows and grows in emotion until the explosive climax of the piece. Jacob's *One-Man Show* allows the audience to discover the technical interactions over time. *Fensadense* provides an explanation before the piece even begins, and then allows the audience to apply their new understanding. This is an important set of considerations

when designing new work: When do we want the audience to feel? When do we want the audience to think? In reality the audience is always doing both, but to what extent, how, why and when are things that can be crafted. I personally believe that the most interesting experiences carefully consider this intellect-emotion balance, especially where technology is involved.

Does the work take risks in design and execution?

We must think about which risks we are taking and why. There are three categories of risk that I typically consider: artistic risk relating to the content of the project, implementation risk relating to the development of the production elements, and performance risk

Fensadense (section 3.6) is an example of a production which contained significant implementation and artistic risk. Not only was all the technology and hardware newly created for the production (3.6.2), but the process of creating the composition was also quite risky, involving a large ensemble in the creation of the piece's structure (3.6.3). On top of that, the timeline was quite compressed and ambitious compared with previous projects. It also was a risk to base the content of the piece completely on the two interactions described in 3.6.2 with no other sounds or systems. However, the performance and execution of the piece became quite refined with many rehearsals and ultimately resulted in detailed documentation, so the performance could tour successfully.

The BBC Proms (4.11.1) is an example of a performance where no new technology systems were used. The workflows were all well thought out. In this case, the challenge was the short amount of time in rehearsal and adapting Jacob's music to the traditional format of click and backing track. Risk could not be tolerated in the design or implementation of the technology because the performance was broadcast live on national radio and recorded live for television. However, the format of Jacob's arrangements added a level of risk to the performance because they required precise balances and active mixing. Without this, parts of the recording would sound very obviously wrong to the casual listener. There were a total of 8 mixing engineers (including myself), and I had to communicate the important actions in the mix to all of them separately. Each engineer was located in a different part of the complex, with some in trucks outside, and we had to separately execute the mix for each platform.

The *One-Man Show* was less of a risk from an artistic standpoint because we adapted songs that were already quite popular on Jacob's YouTube channel. We knew the material could stand on its own. The risk in implementation of the technology was quite high, as was the execution of the performance. Looping performances are quite unforgiving for any kind of variation in timing or structure; any small mistake becomes magnified very quickly. There was also no redundancy in any of the technology, so there were many components whose failure would completely stop the show.

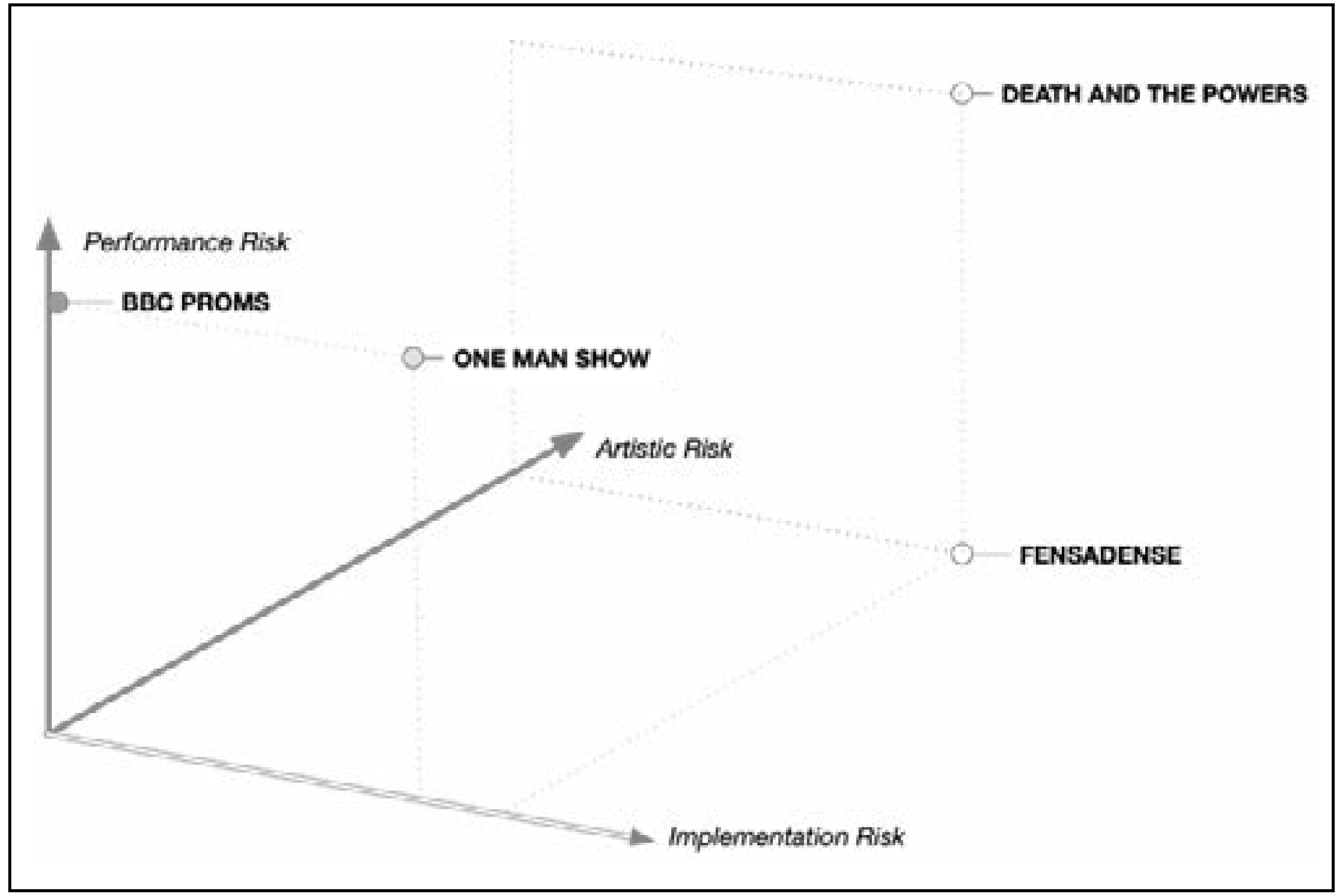


Fig. 6-2: Plot of performance, implementation and artistic risk for performances.

Death and the Powers is an example of a project with high implementation, performance and artistic risk. The work was completely new, and the musical textures were dependent on customized DSP that had not been tested in large venues before the final rehearsal period. The rest of the production was dependent on the implementation of custom technology with very little redundancy, and it required a high number of operators, all with specialized knowledge. This project took an incredibly long time to develop, and the premiere was pushed back several times because the pieces (artistic, conceptual and technologically) were not finished or functioning. Once it was finally finished, however, it was hailed as a feat of creative and technical innovation, truly providing a new paradigm for performance with all of its totally new, finely tailored, and completely integrated elements.

When it comes to new work, we often must make decisions about how much risk to take on. This can be proportionally based on the amount of time that is available (i.e., the less time to design, build, and rehearse before performing live, the greater the risk). However, I believe it is important to take some sort of risk in every project. Without this, we end up creating similar experiences and having to grow other aspects such as spectacle to stay relevant and interesting. I hope that Hyperproductions will continue to retain their "experimental" nature, where the testing of new ideas is an accepted part of the creative process.

Does the medium support the content?

Importantly, do the technology and media support the full intensity and vision of the work? There are many who will try to do a video project with dim projectors, or a large sound installation with small speakers. It is important that the elements supporting the experience be intentionally designed and chosen to achieve a desired outcome. Often this can involve more work than one would imagine, but it is the key to a striking and engaging work.

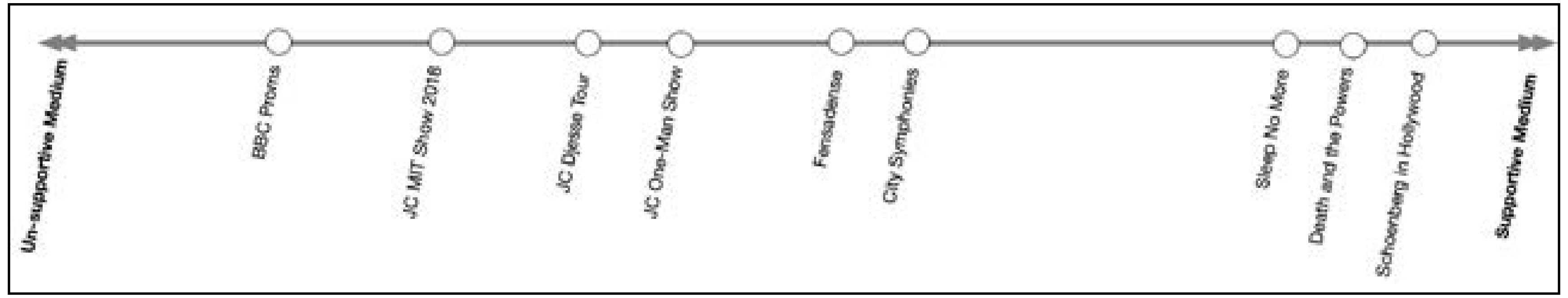


Fig. 6-3: Continuum of production from Un-supportive medium to supportive medium

There are situations where it is very difficult for the medium to support the content. Due to rigging limitations at the BBC Proms, the PA system was mounted 6 meters too high. Since we were aware of the issue, we were able to compensate in other ways by adding exciters to certain channels to increase intelligibility and high end. Even if it is difficult to control one's circumstances, it is good to understand if the gear being used is working for or against the creative intent of the piece. In other cases, limited media may become part of the artistic intent, and an unsupportive medium may become supportive. Many sounds are now designed specifically for cell phone speakers, especially on social media applications. This is a case where the content has been modified to adapt to the medium.

For a piece such as *Schoenberg in Hollywood* (3.8) the systems were carefully chosen for both the space and the content of the piece. This technical design required careful collaboration with composer and technical crew at the venue, as well as at the sound company supplying equipment. It required an understanding of how the choice and operation of equipment affects the artistic outcome and the communication of those consequences (along with logistics and cost) to the creative team. Modifications to these systems for any reason (logistical, technical, creative) almost always have creative ramifications, and the importance of communication and collaboration cannot be underestimated.

On a world tour, such as the two for Jacob, we attempted to design technology that is robust in many environments and quite flexible in terms of sound reinforcement, equipment durability and configuration. This results in the best possible production in the widest range of venues. For example, if we were to have an extremely feedback prone microphone configuration, some venues may have more leakage of the house PA sound onto stage and require that the entire show be quieter. It is important to consider all the environments and systems that will be used and find a good balance of implementation to satisfy the requirements of the performance.

When it is known that the medium will be completely unsupportive, it is important to make sure the content is not completely dependent on the medium and can stand by itself. If Jacob is asked to do a performance and we know the sound system will be very low-quality, we do not use the harmonizer for that performance, since it really shines with a full range sound system including subwoofers. Instead he might perform a more basic set with only the piano, which is much less demanding on the audio system.

For new works, this is an important concept to consider, especially since many more complex production endeavors only come across positively with the right selection of equipment. If the composition of the content is also able to be tailored, this can help even more. For example, arena rock developed with quite simple rhythm and large amounts of reverb simply because that is the type of music that worked well in arenas. There are many compositional aspects of a piece that the production medium and environment can either support or hinder. In the past, pieces were produced and performed with little thought to the environment. We must not ignore this connection as we move forward.

6.4 New Technology for Hyperproduction

At the beginning of this document, I claimed that we are on the cusp of creating a new artform, Hyperproduction, with the magic of the best truly live music performances and the production scale and live effects only currently possible for pop tours. Ultimately it will be technology which enables creatives to do this. While many of the technologies currently exist, the completion of the projects in this dissertation has led me to consider another cusp in technological development as well.

Fifty years ago there were many composers, including Stockhausen, Babbitt, Berio and others, who experimented with multiple forms of electronic music. Some of them focused on synthesis, and a different group focused on manipulating recorded audio, and the two disciplines were not mixed together. Even as recently as the late 1990s, tools were developed either for synthesis (Digital Performer, Logic, OMS) or audio (Sequoia, ProTools, Pyramix). However, are now living in an age where audio and synthesized sounds can be manipulated and crafted cohesively together. Tools like Ableton Live allow creators to yield a unified "paintbrush" working with audio and MIDI connected in a wholly integrated way and in real-time. Meanwhile, other products like Protools and Digital Performer have tried to add previously lacking MIDI or audio support respectively, but from the perspective of either MIDI or audio rather than rethinking the approach like Ableton did.

Fifty years ago, it was uncommon to find any level of sophisticated inter-media technology in live performance. Today we take quite complex systems of technology for granted, and almost everyone is using them for live performance. One might claim that we are now in a golden age of live production. Indeed, with significant advances in computation, networking and user interfaces, it is possible to quickly and economically build instruments, shows, tours, and experiences.

I believe that we are in the "Stockhausen, Babbitt and Berio age" with live production technology. The systems are powerful, and the interfaces are improving. However, the approach for developing new systems is fundamentally based on using old paradigms and bolting new capabilities on top, while keeping antiquated ideas about how the capabilities should interact. In truth, we are on the cusp of redefining live, emotional, musical experiences using technology to bridge many mediums together. That is *exciting,* because it enables us to create live productions in a completely new way. Once there is a generalized, easily operated method to manipulate environments the way *Powers, Fensadense, Djesse,* and all of the other projects documented here have linked together many systems based on human expression, we will be in a new age of production technology.

Given this coming revolution, we *must* start from the ground up and think about what it means to have inter-media, with many articulated forms used together and created together. Many of the projects outlined in this dissertation discuss possible directions and new methods and concepts for folding together technology and music, but I want to impress that this is *just the beginning.* We now get to imagine what the future can hold. We now get to throw away existing tools and start from scratch. We now get to really begin to think about how technology can contribute to the creative, emotional and experiential causes that we care about. The work I've described in this document is not a final product. Rather I hope that it opens the door to many ideas that we can continue to build upon. Or, even better, that it may inspire us to develop the next systems and art work that weave together many media using musical sensibilities, nuance and articulations. This could be for a new popular music tour, or it might be the way our lights turns on with our alarm clock.

6.5 Future Work

I hope to continue on scales large and small with this work. My greatest passion is to design and develop live experiences and performances. I hope to continue to do this with both Tod Machover and Jacob Collier, and additionally, to seek out boundary pushing artists with whom I may collaborate on the development of new projects.

6.5.1 Jacob Collier

We will begin to imagine the next 5 years of Jacob's touring with an expanded band and musical systems. We must take what we have and shrink it and make it easier and more compatible to deploy within the requirements world of touring. That will leave room to expand upon the ideas, both technical and artistic, which will allow the introduction of more new ideas and experiments to test. In this case, we will build upon what exists, keeping elements that work well and letting others fall away as we come up with equipment and combinations of technology and performance strategies to best support Jacob's next era of music production and performance.

At the same time, we will continue to develop inter-media studio and production systems that Jacob currently uses to make and create music. We will enhance his in-room video and audio systems and our notation strategies for performance design, devices and instruments that allow new interactions.

6.5.2 Opera of the Future

There are several other exciting Opera of the Future projects that will involve large scale musical collaboration and production:

- After years of traveling about the world, we are planning to bring City Symphonies home to Greater Boston. Tod Machover says: "Building on our past successes, we will make the best City Symphony ever, uniting institutions and individuals, long-time residents and recent arrivals, well-traveled neighborhoods and secret spots, songs and sounds and wails and whispers, all to make the Hub sync-and-spin with ecstatic energy and profound connection. So join us now to help create HUBsymphony, which will premiere at HUBweek 2021. Let's show the world what our HUB really sounds like!"[5]

- Another symphony project in Chennai will focus on the connection of language, film and community through music and involve performance of video and audio simultaneously.

[5] "HUBsymphony." CITY SYMPHONIES, citysymphonies.media.mit.edu/hubsymphony.html.

- In the demilitarized zone between North and South Korea another symphony aims to bring together musicians from both countries as a catalyst for bringing change and unification and helping to end the humanitarian crisis in the region.[6]
- At the Dubai Expo in 2020, symphony entitled *"Flow Symphony"* and a multimedia experience will be developed to link many communities through a common understanding of how ideas and goods flow around the world.
- There will be an expansion of the study of Gamma related treatment and multi-sensory experiences.
- Hyperproduction as a software platform will be enhanced to allow more flexible, simple and creative uses by the general public.
- A new opera using innovative multisensory technologies to explore the relationship between humans and trees.

6.5.3 Beyond

I will continue to practice and refine the methodologies detailed in this dissertation and think about what constitutes the most human, most compelling live musical performances. While much of this investigation is related specifically to production of performances and experiences, I also intend to continue the process of making production and inter-media experiences simple to design and integrate in our everyday lives through development of tools and systems. This may enable another avenue of musical expression, even in non-musical media, that I believe is both critical to our existence as emotional beings and important for our development in community and society.

The work documented here is made possible and is uniquely special because of the connection among the many different systems and layers, and the connection between the technical and creative individuals working together on the implementation of the piece. For example, the experiences created for *Death and the Powers* and *Fensadense* were possible only because *every aspect of every system* was able to connect together and influence one another, and all the creative and technical individuals worked extraordinarily closely throughout the conception of both projects. The same is true for Jacob's tours, and for every other project described in this document. It is not enough to connect one or two components together, it is not enough to have the creative team interact with a designer who speaks to a technical team. It is the connection of *all systems* and *all disciplines* that results in the most interesting and ground-breaking product.

[6] SEUNG-HYE, Y., "Innovative composer hopes to unify the Koreas." KOREA JOONGANG DAILY, March 2019.

Furthermore, these connections throughout the production process result in greater connection between the artists and audiences who perform and experience the works, because the technologies are able to *entirely* work together in support of creative goals. This is the core of the ***Universal Connection*** objective in section 6.1.1 above. We can imagine a future where many kinds of systems and projects are designed to maximize this type of synergy. The modalities of our environments will be linked together and intelligently reacting to our emotions and expressions. Here are some scenarios that I imagine along these lines:

- A live tour that provides scale and spectacle of the largest pop sensations, but features musicians that are incredibly talented and excel at musical communication and interaction with other musicians performing on stage and the audience. This type of show is incredibly flexible and organic, while still incorporating the most impactful techniques and practices that traditionally require timecode. Technicians working on such a production focus their efforts on creative input, mixing and shaping performance systems and the overall experience rather than triggering material at the right moment.

- A high-bandwidth network of interlinked spaces that each excel at one particular function. These functions may be acoustic (i.e. the best room to experience and/or record a live orchestra), operational (i.e. a space that allows audience and performers to fly) or pertaining to history and *vibe* (i.e. a historic place where something important took place that informs the feeling and impact of events that currently take place, for example, the Stonewall Inn or Abbey Road Studio). The spaces may be used together for distributed, connected experiences across the world. Participants gain a sense of the special properties of certain spaces remotely while experiencing their local space in person. If each space is considered an instrument, the complete ensemble of them forms an orchestra. A "symphony" may be written for an orchestra of these spaces.

- Home automation systems that create a real time adaptive experience based on our mood and emotion. Infrastructure links together modalities for multi-sensory experiences that are intelligently articulated based on intuitive expression by the average individual. This turns our living, working and playing environments into expressive media to allow these spaces to represent us more faithfully and to enhance our ability to communicate and live in harmony with our environments.
- A next-generation classical music institution that has greater flexibility to document, share and perform new and experimental works involving massive collaboration and technology. This is combined with a flexible performance venue that can morph acoustically and supports many types of arrangements and configurations of ensemble. It also includes a new type of organization of personnel, fluidly technical and creative, and a system of governance and hierarchy to ensure musicians are treated fairly but that leaves flexibility to change the style of rehearsal and performance, and to adapt to the changing world of music.
- Technology that allows musicians starting out to develop a more advanced tour, which does not require a cost-prohibitive crew to travel, and simultaneously allows musical performances to achieve advanced multi-sensory synchronization without backing tracks or timeline based media.

6.5.4 Adoption

While it is very good to analyze and discuss these objectives and projects, it is also important to consider if the industry and the world is ready (or will ever be ready) to adopt these ideas and see them become more common. I am lucky to get to work with many professionals and artists at high levels in the industry, and through my conversations I do believe there is an appetite for a new approach to production design. Artists and designers alike are growing tired of the same process and are constantly looking for interesting techniques to allow them to be more creative while performing. The individuals I've spoken with range from professional DJs, content

and production designers, recording artists, agents and managers, audio engineers and sound designers, choreographers and dancers, classical musicians and conductors, and many others.

The key to bringing new approaches to these typically conservative operations is careful design of fallbacks and redundancies. If we're adding Hyperproduction to a DJ set controlling all of the lighting and visuals as a replacement for a timecode system, it makes sense to keep running timecode and slowly migrate each system over. As a backup, we can devise systems that allow the operators to revert to the prior existing control system if errors arise. In this way it is possible to guarantee that the system will be at least as functional as the existing infrastructure, while possibly adding much more capability. For big artists in situations where failure cannot be tolerated, this takes away the uncertainty of deploying a new system without seeing it tested over years of touring.

With this type of mindset, and given conversations and interest from the industry, I am excited to see just how far we can push these ideas. I hope to get them into as many hands and on as many stages as possible in the coming years.

6.6 Final Words

While much of this work revolves around creating and using technology, I hope that it is clear that the humanity and connection it brings to us is the fundamental goal and mission of my investigation. Technology affords a great number of advantages—the ability to reach many, scale up our expression to infinite heights, achieve feats and sensory experiences that have never been felt or seen— but this is pointless if we cannot preserve the human connection in such endeavors. Technology for technology's sake will never be meaningful, and it does not work alone as a tool to connect us.

Ultimately we want to build experiences that weave technology and humanity together, and design tools that allow us to create such experiences with a single paintbrush focusing on emotion and creativity rather than nuts and bolts or integrations. Rather than using scale to impress, we can come up with simple and elegant interactions that enlighten audiences and leave conceptual room to explore, discover and experience our humanity. I believe that technology can and will support and enhance the human aspects of these coming endeavors.

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