THIRD EDITION

SOUND SYSTEMS: DESIGN AND OPTIMIZATION

MODERN TECHNIQUES AND TOOLS FOR SOUND SYSTEM DESIGN AND ALIGNMENT

BOB McCARTHY

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Sound System Design and Alignment

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Dedication

*To the woman who knew me back when this journey began, and is still there through
three editions, the love of my life,
Merridith.*

In memoriam

*During the course of writing the first edition our field lost one of its most well-loved
and respected pioneers, Don Pearson a.k.a. Dr Don. I had the good fortune to work with
Don and receive his wisdom. He was there when it all began and is still missed. More recently
we lost Tom Young, and Mike Shannon who contributed so much to the industry and to me
personally. And finally, I lost my brother Chris who was an inspiration to all who knew him.*

This book is about a journey. On the one hand, the subject is the journey of sound as it travels through a sound system, then through the air, and inevitably to a listener. It is also a personal journey, my own quest to understand the complicated nature of this sound transmission. The body of this text will detail the strictly technical side of things. First, however, I offer you some of the personal side.

I was supposed to build buildings. Unbeknownst to me at the time, this calling was derailed on February 9, 1964 by the appearance of the Beatles on *The Ed Sullivan Show*. Like so many of my generation, this landmark event brought popular music and an electric guitar into my life. I became a great enthusiast of live concerts, which I regularly attended throughout my youth at any chance presented. For years, it remained my expectation that I would enter the family construction business. This vision ended on a racetrack in Des Moines, Iowa on June 16, 1974. The experience of hearing the massive sound system at this Grateful Dead concert set my life in a new direction. On that day I made the decision that I was going to work in live concert sound. I wanted to help create this type of experience for others. I would be a mix engineer and my dream was to one day operate the mix console for big shows. I set my sights on preparing for such a career while at Indiana University. This was no simple matter because there was no such thing as a degree in audio. I soon discovered the Independent Learning Program. Under the auspices of that department, I assembled a mix of relevant courses from different disciplines and graduated with a college-level degree in my self-created program of audio engineering.

FIGURE 0.1
Ticket stub from the June 16, 1974 Grateful Dead concert in Des Moines, Iowa that led to my life of crime

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By 1980, I had a few years of touring experience under my belt and had moved to San Francisco. There I forged relationships with John Meyer, Alexander Yuill-Thornton II (Thorny) and Don Pearson. These would become the key relationships in my professional development. Each of us was destined to stake our reputations on the same piece of equipment: the dual-channel FFT analyzer.

I would like to say that I have been involved in live concert measurement with the dual-channel FFT analyzer from day one, but this is not the case. It was day two. John Meyer began the process on a Saturday night in May of 1984. John took the analyzer, an analog delay line and some gator clips to a Rush concert in Phoenix, Arizona, where he performed the first measurements of a concert sound system using music as the source with audience in place. I was not destined to become involved in the project until the following Monday morning.

From that day forward, I have never been involved in a concert or a sound system installation without the use of a dual-channel FFT analyzer. I haven't mixed a show since that day, resetting my vision to the task of helping mix engineers to practice their art. For Don, John, Thorny and many others, the idea of setting up a system without the presence of the FFT analyzer was unthinkable. Seeing a sound system response in
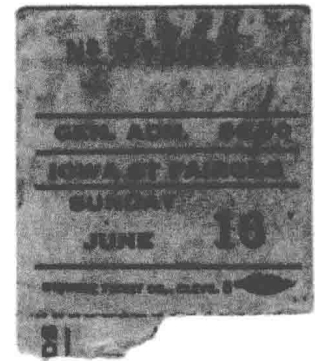

FIGURE 0.2
The author with the prototype SIM analyzer with the Grateful Dead in July 1984 at the Greek Theater in Berkeley, California (Clayton Call photo)

high resolution, complete with phase, coherence and impulse response, is a bell that cannot not be un-rung. We saw its importance and its practical implications from the very beginning and knew the day would come when this would be standard practice. Our excitement was palpable, with each concert resulting in an exponential growth in knowledge. We introduced it to everyone who had an open mind to listen. The first product to come from the FFT analysis process was a parametric equalizer. A fortuitous coincidence of timing resulted in my having etched the circuit boards for the equalizer on my back porch over the weekend that John was in Phoenix with Rush. This side project (a bass guitar preamp) for my friend Rob Wenig was already six months late, and was destined to be even later. The EQ was immediately pressed into service when John nearly fell over as he saw that it could create the complementary response (in both amplitude and phase) to what he had measured in Phoenix. The CP-10 was born into more controversy than one might imagine. Equalization has always been an emotional "hot button" but the proposition that the equalizer was capable of counteracting the summation properties of the speaker/room interaction was radical enough that we obtained the support of Stanford's Dr Julius Smith to make sure that the theory would hold up.

Don Pearson was the first outside of our company to apply the concepts of in-concert analysis in the field. Don was owner of Ultrasound and was touring as the system engineer for the Grateful Dead. Don and the band immediately saw the benefit and, lacking patience to wait for the development of what would become the Meyer Sound SIM System, obtained their own FFT analyzer and never looked back. Soon thereafter, under the guidance of San Francisco Opera sound designer Roger Gans, we became involved with arena-scale performances for Luciano Pavarotti. We figured it was a matter of months before these techniques would become standard operating procedure throughout the industry. We had no idea it would take closer to twenty years! The journey, like that of sound transmission, was far more complex than we ever expected. There were powerful forces lined up against us in various forms: the massive general resistance of the audio community to sound analyzers and the powerful political forces advocating for alternate measurement platforms, to name a few.

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In general, the live sound community was massively opposed to what they conceptualized as an analyzer dictating policy to the creative forces involved in the music side of the experience. Most live concert systems of the day lacked complexity beyond piles of speakers with left and right channels. This meant that the process of alignment consisted of little more than equalization. Because all of the system calibration was being carried out at a single location, the mix position, the scientific and artistic positions were weighing in on the exact same question at the same point in space. Endless adversarial debate about what was the "correct" equalization ensued because the tonal balancing of a sound system is, and always has been, an artistic endeavor. It was an absurd construct. Which is better—by ear or by analyzer?

This gave way to a more challenging and interesting direction for us: the quest beyond the mix position. Moving the mic out into the space left us with a terrible dilemma: The new positions revealed conclusively that the one-size-fits-all version of system equalization was utter fantasy. The precision tuning of parametric filters carried out with great care for the mix position had no justification at other locations. The interaction of the miscellaneous parts of the speaker system created a

FIGURE 0.3
November 1984 photo of Luciano Pavarotti, Roger Gans, the author (back row), Drew Serb, Alexander Yuill-Thornton II and James Locke (front row) (Drew Serb photo)

highly variable response throughout the room. Our focus shifted from finding a perfect EQ to the quest for uniformity over the space.

This would require the subdivision of the sound system into defined and separately adjustable subsystems, each with individual level, equalization and delay capability. The subsystems were then combined into a unified whole. The rock and roll community was resistant to the idea, primarily because it involved turning some of the speakers down in level. The SPL Preservation Society staunchly opposed anything that might detract from the maximum power capability. Uniformity by subdivision was not worth pursuing if it cost power (pretty much nothing else was either). Without subdivision, the analysis was pretty much stuck at the mix position. If we are not going to change anything, why bother to look further?

There were other genres that were open to the idea. The process required the movement of a microphone around the room and a systematic approach to deconstructing and reconstructing the sound system. We began developing this methodology with the Pavarotti tours. Pavarotti was using approximately ten subsystems, which were individually measured and equalized and then merged together as a whole. Our process had to be honed to take on even more complexity when we moved into the musical theater world with Andrew Bruce, Abe Jacob, Tony Meola, Tom Clark and other such sound designers. Our emphasis changed from providing a scientifically derived tonal response to maximizing consistency of sound throughout the listening space, leaving the tonal character in the hands of the mix engineer. Our tenure as the "EQ police" was over as our emphasis changed from tonal quality to tonal equality. The process was thus transformed into *optimization*, emphasizing spatial uniformity while encompassing equalization, level setting, delay setting, speaker positioning and a host of verifications on the system. A clear line was drawn between the artistic and the scientific sectors.

In the early days, people assessed the success of a system tuning by punching out the filters of the equalizer. Now, with our more sophisticated process, we could no longer re-enact before-and-after scenarios. To hear the "before" sound might require repositioning the speakers, finding the polarity reversals, setting new splay angles, resetting level and time delays, and finally a series of equalizations for the different subsystems. Lastly, the role of the optimization engineer became clear: to ensure that the audience area receives the same sound as the mix position.

In 1987, we introduced the Source Independent Measurement system (SIM). This was the first multichannel FFT analysis system designed specifically for sound system optimization (up to sixty-four channels). It consisted of an analyzer, multiple mics and switchers to access banks of equalizers and delays. All of this was under computer control, which also kept a library of data that could be recalled for comparison of up to sixteen different positions or scenarios. It thereby became possible to monitor the sound system from multiple locations and clearly see the interactions between subsystems. It was also possible to make multiple microphone measurements during a performance and to see the effects of the audience presence throughout the space.

This is not to say we were on Easy Street at this point. It was a dizzying task to manage the assembly of traces that characterized a frequency response, which had to be measured in seven separate linear frequency sections. A single data set to fully characterize one location at a point in time was an assembly of sixty-three traces, of which only two could be seen at any one time on the tiny 4-inch screen. Comparison of one mic position to another had to be done on a trace-by-trace basis (up to sixty-three operations). It was like trying to draw a landscape while looking through a periscope.

The multichannel measurement system opened the door to system subdivision. This approach broke the pop music sound barrier with Japanese sensation Yuming Matsutoya under the guidance of Akio Kawada, Akira Masu and Hiro Tomioka. In the arenas across Japan we proved that the techniques we had developed for musical theater and Pavarotti (level tapering, zoned equalization and a carefully combined subsystem) were equally applicable to high-power rock music in a touring application.

The introduction of the measurement system as a product was followed by the first training seminar in 1987. A seminal moment came from an unexpected direction during this first seminar as I explained the process of system subdivision and mic placement for system optimization. Dave Robb, a very experienced engineer, challenged my mic placement as "arbitrary." In my mind, the selection was anything but arbitrary. However, I could not, at that moment, bring forth any objective criteria with which to refute that assertion. Since that humiliating moment, my quest has been to find a defensible methodology for every decision made in the process of sound

system optimization. It is simply not enough to know *that* something works; we must know *why* it works. Those optimization methodologies and an accompanying set of methods for sound system design are the foundation of this book.

I knew nothing of sound system design when this quest began in 1984. Almost everything I have learned about the design of sound systems comes from the process of their optimization. The process of deconstructing and reconstructing other people's designs gave me the unique ability/perspective to see the aspects that were universally good, bad or ugly. I am very fortunate to have been exposed to all different types of designs, utilizing many different makes and models of speakers, with all types of program materials and scales. My approach has been to search for the common solutions to these seemingly different situations and to distill them into a repeatable strategy to bring forward to the next application.

Beginning with that very first class, with little interruption, I have been optimizing sound systems and teaching anybody who wanted to attend my seminars everything I was learning. Thorny, meanwhile, had moved on and founded a company whose principal focus was sound system optimization services using the dual-channel FFT systems. Optimization as a distinct specialty had begun to emerge.

The introduction of SIA-SMAART in 1995 resulted from the collaboration of Thorny and Sam Berkow with important contributions by Jamie Anderson and others in later years. This low-cost alternative brought the dual-channel FFT analyzer into the mainstream and made it available to audio professionals at every level. Even so, it took years before our 1984 vision of the FFT analyzer, as standard front-of-house equipment, would become reality. Unquestionably, that time has arrived. The paradigm has reversed to the point where tuning a system without scientific instrumentation would be looked at with as much surprise as was the reverse in the old days.

Since those early days we have steadily marched forward with better tools—better sound systems, better sound design tools and better analyzers. The challenge, however, has never changed. It is unlikely that it will change, because the real challenge falls mostly in the spatial distribution properties of acoustical physics. The speakers we use to fill the room are vastly improved and signal-processing capability is beyond anything we dreamed of in those early days. Prediction software is now readily available to illustrate the interaction of speakers, and we have affordable and fast analyzers to provide the on-site data.

And yet we are fighting the very same battle that we have always fought: the creation of a uniform sonic experience for audience members seated everywhere in the venue. It is an utterly insurmountable challenge. It cannot be achieved. There is no perfect system configuration or tuning. The best we can hope for is to approach uniformity. I believe it is far better to be coldly realistic about our prospects. We will have to make decisions that we know will degrade some areas in order to benefit others. We want them to be informed decisions, not arbitrary ones.

This book follows the full transmission path from the console to the listener. That path has gone through remarkable changes along its entire electronic voyage. But once the waveform is transformed into its acoustic form it enters the very same world that Jean-Baptiste Joseph Fourier found in the eighteenth century and Harry Olson found in the 1940s. Digital, schmigital. Once it leaves the speaker, the waveform is pure analog and at the mercy of the laws of acoustical physics. These unchanging aspects of sound transmission are the focus of 90 per cent of this book.

Let's take a moment to preview the challenges we face. The primary player is the interaction of speakers with other speakers, and with the room. These interactions are extremely complex on the one hand, and yet can be distilled down to two dominant relationships: relative level and relative phase. The combination of two related sound sources will create a unique spatial distribution of additions and subtractions over the space. The challenge is the fact that each frequency combines differently, creating a unique layout. Typical sound systems have a frequency range of 30 to 18,000 Hz, which spans a 600:1 ratio of wavelengths. A single room, from the perspective of spatial distribution over frequency, is like a 600-story skyscraper with a different floor plan at every level. Our job is to find the combination of speakers and room geometry that creates the highest degree of uniformity for those 600 floor plans. Every speaker element and surface will factor into the spatial distribution. Each element plays a part in proportion to the energy it brings to the equation at a point in the space. The combined level will depend upon the relationship between the individual phase responses at each location at each frequency. How do we see these floor plans? With an acoustic prediction program we can view the layout of each floor, and compare them and

see the differences. This is the viewpoint of a single frequency range analyzed over the entire space. With an acoustic analyzer we get a different view. We see a single spot on each floor from the foundation to the rooftop through a piece of pipe as big around as our finger. This is the viewpoint of a single point in space analyzed over the entire frequency range.

This is a daunting task. But it is comprehensible. This book will provide you with the information required to obtain the X-ray vision it takes to see through the 600-story building from top to bottom, and it can be done without calculus, integral math or differential equations. We let the analyzer and the prediction program do the heavy lifting. Our focus is on how to read X-rays, not on how to build an X-ray machine.

The key to understanding the subject, and a persistent theme of this book, is sound source identity. Every speaker element, no matter how big or small, plays an individual role, and that solitary identity is never lost. Solutions are enacted locally on an element-by-element basis. We must learn to recognize the individual parties to every combination, because therein lie the solutions to their complex interaction.

This is not a mystery novel, so there is no need to hide the conclusion until the last pages. The key to spatial uniformity is control of the overlap of the multiple elements. Where two elements combine they must be in phase to obtain spatial uniformity. If the elements cannot maintain an in-phase relationship, then they must decrease the overlap and subdivide the space so that one element takes a dominant role in a given area. There are two principal mechanisms to create isolation: angular separation and displacement. These can be used separately or in combination and can be further aided by independent control of level to subdivide the room. This is analogous to raising children: If they don't play well together, separate them. The interaction of speakers to the room is similar to the interaction of speakers with other speakers. Those surfaces that return energy back toward our speakers will be the greatest concern. The strength of the inward reflections will be inversely proportional to our spatial uniformity.

There is no single design for a single space. There are alternate approaches and each involves tradeoffs. There are, however, certain design directions that keep open the possibility of spatial uniformity and others that render such hopes statistically impossible. A major thrust of the text will be devoted to defining the speaker configurations and design strategies that maximize the potential for spatial uniformity.

Once designed and installed, the system must be optimized. If the design has kept the door open for spatial uniformity, our task will be to navigate the system through that door. The key to optimization is the knowledge of the decisive locations in the battle for spatial uniformity. The interactions of speakers and rooms follow a consistent set of spatial progressions. The layering of these effects over each other provides the ultimate challenge, but there is nothing random about this family of interactions. It is logical and learnable. Our measurement mics are the information portals to decipher the variations between the hundreds of floor plans and make informed decisions. Our time and resources are limited. We can only discern the meaning of the measured data if we know where we are in the context of the interaction progressions.

We have often seen the work of archeologists where a complete rendering of a dinosaur is created from a small sampling of bone fragments. Their conclusions are based entirely on contextual clues gathered from the knowledge of the standard progressions of animal anatomy. If such progressions were random, there would be nothing short of a 100 per cent fossil record that could provide answers. From a statistical point of view, even with hundreds of mic positions, we will never be able to view more than a few tiny fragments of our speaker system's anatomy in the room. We must make every measurement location count toward the collection of the data we need to see the big picture. This requires advance knowledge of the progression milestones so that we can view a response in the context of what is expected at the given location. As we shall see, there is almost nothing that can be concluded from a single location. The verification of spatial uniformity rests on the comparison of multiple locations.

This book is about defined speakers in defined array configurations, with defined optimization strategies, measured at defined locations. This book is not intended to be a duplication of the general audio resource texts. Such books are available in abundance and it is not my intention to encompass the width and breadth of the complete audio picture. My hope is to provide a unique perspective that has not been told before, in a manner that is accessible to the audio professionals interested in a deeper understanding of the behavior of sound systems in the practical world.

There are a few points that I wish to address before we begin. The most notable is the fact that the physical realities of loudspeaker construction, manufacture and installation are largely absent. Loudspeakers are described primarily in terms of acoustic performance properties, rather than the physical nature of what horn shape or transducers were used to achieve it. This is also true of electronic devices. Everything is weightless, colorless and odorless here. The common transmission characteristics are the focus, not the unique features of one model or another.

The second item concerns the approach to particular types of program material such as popular music, musical theater or religious services, and their respective venues such as arenas, concert halls, showrooms or houses of worship. The focus here is the shape of the sound coverage, the scale of which can be adjusted to fit the size of the venue at the appropriate sound level for the given program material. It is the venue and the program material taken together that create an application. The laws of physics are no different for any of these applications, and the program material and venues are so interchangeable that attempts to characterize them in this way would require endless iterations. After all, the modern-day house of worship is just as likely to feature popular music in an arena setting as it is to have speech and chant in a reverberant cathedral of stone.

The third notable aspect is that there are a substantial number of unique terminologies found here and, in some cases, modification of standard terminologies that have been in general use. In most cases the conceptual framework is unique and no current standard expressions were found. The very young field of sound system optimization has yet to develop consistent methods or a lexicon of expressions for the processes shown here. In the case of some of these terms, most notably the word "crossover," there are compelling reasons to modify the existing usage, which will be revealed in the body of the text.

The book is divided into three parts. The first part, "Sound systems," explores the behavior of sound transmission systems, human hearing reception and speaker interaction. The goal of this part is a comprehensive understanding of the path the signal will take, the hazards it will encounter along the way and how the end product will be perceived upon arrival at its destination. The second part, "Design," applies the properties of the first part to the creation of a sound system design. The goals are comprehensive understanding of the tools and techniques required to generate a design that will create a successful transmission/reception model. The final part, "Optimization," concerns the measurement of the designed and installed system, its verification and calibration in the space.

From the viewpoint of my publisher, Focal Press, this is indeed the third edition of *Sound Systems: Design and Optimization*. From my perspective it feels more like the thirtieth edition, because I have been writing about these same subjects for thirty+ years. You might think I would have figured this subject out by now but I can assure you I am still learning. This field of work continues to evolve as we get better tools and techniques, which is exactly what I find most interesting about it. Study in this field is a moving target as new technology opens doors and removes obstacles and excuses. The more I learn about this, the more I realize how much I have to learn.

Adding new areas is the easy part of creating a new edition. It's what to do with the previous material that presents a challenge. There are two ways to approach the old material: innocent until proven guilty, or the opposite. The former approach leaves things in unless they are conclusively out of date or irrelevant. The latter approach throws the old material out unless it can prove it is still current practice and up to date.

I studied the later editions of several other authors and noticed a troubling trend. Although new information was added in later editions, a lot of old information remained in place. Seeing vacuum tube circuits from the 1960s in a current-day pro audio text was a tipping point for me. The decision was made to trim out the old to make room for the new. If it's the way we do things now, it's in. If we've moved on, it's out.

The surprise for me was how much we have moved forward in this time, which meant entire chapters were bulldozed and rebuilt. One of the hardest decisions was to let go of the perspective sidebars that colored the previous editions with the wisdom and insight of so many of my friends and colleagues. The bottom line is that there is simply too much new information to be added. I take comfort in knowing that there are many other places where those voices can be heard and that optimization is now firmly ensconced as part of the audio landscape.

There have been no updated laws of physics and our audio analyzers still compute things the same way as they did in 1991. But today's analyzers are faster, easier and able to multitask, which means we can get much more done in a short time. We can tune methodically, and methods are what this book is about. We have far better loudspeakers, processors and steadily better rooms to work in. All this leads to the primary goal of this third edition: current methodologies and techniques for sound system design and optimization.

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