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Recording and Reproduction over Two Loudspeakers as Heard Live

Part 2: Recording Concepts and Practices

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ABSTRACT

For a half century, the crucial interaction between recording engineer and monitor loudspeakers during two-channel stereophonic recording has not been resolved, leaving the engineer to cope with uncertainties. However, recent advances in defining and improving this loudspeaker-room-listener interface have finally allowed objectivity to inform and shape the engineer's choices. The full potential of the two-channel stereo format is now accessible to the recording engineer, and in a room that is just as normal as most consumers' rooms. The improved reproduction has also allowed a deeper understanding of the merits and limits of spaced and coincident/near-coincident microphone arrays. As a result of these and earlier observations, a four-microphone array was conceived which exploits natural hearing processes to achieve greater auditory realism from two loudspeakers. A number of insights have emerged from the experiments.

1. INTRODUCTION

This report is offered to demonstrate the importance and utility of accurate monitors to a recording engineer. There is this conundrum at the heart of the profession: the job depends upon listening to two loudspeakers in a room, yet one soon learns that what is heard from two loudspeakers in a room cannot be taken literally due to unspecified inaccuracies unique in every setup. This has been a dilemma for a very long time, and has unavoidably affected two-channel stereo recording from its beginning. When considering the unknowns involved, there is every rational reason to despair of a solution. No matter what qualities might be designed into the loudspeaker itself, it has to be placed in a room that will have its own influences, mixing and multiplying the potential distortions and distractions. This just didn't seem solvable at all. The future appeared to be a continuation of ad hoc arrangements.

All of that ended in 2005 with the loudspeakers described in Part 1 of this paper [1], [2]. I came to recognize their capacity as true monitor loudspeakers, and adding to improbability, did so in my own living room, a remarkable development on its own. I heard them as able to take charge of the room – or ignore it - and produce whatever the source material called for. After a period of appreciation the speakers faded from notice as I began to re-evaluate recordings and their microphone techniques with a confidence that only objectivity can sustain. A career-long pursuit that had ended ten years earlier with my retirement gradually resumed, and led to a series of experiments to see if accurate monitors could really make the difference I imagined they would during those earlier years.

2. LEARNING FROM ACCURATE MONITORS

A great deal of time was spent listening to many CDs in my collection, which amounted to an extended period of relearning. Most were classical, but eventually just about everything was sampled out of curiosity. The sum of the experience uncovered several particular areas that I found to be consequential:

- There were fewer distractions from involvement in the music. This observation prompted a new awareness of how noticeable the absence of something can be. It was my first experience with the effect of an absence of audio artifacts.

- The stereo canvas was larger and more complete than I had known, and the context of the recording was the most valuable and convincing component. It is a prerequisite to hear and know the entire capability of the format before trying to make the most of it. In other words, accurate monitors are a prerequisite for accurate recording.

- Despite this improved display of the context of the recording venue, most recordings still sounded deficient in this respect. Direct sounds in isolation are not very useful; it's the context of those sounds that defines them for us. In classical music, that relationship is familiar to audiences everywhere, yet most recordings emphasize the direct sound at the expense of its context.

- Most importantly, the performance of coincident/near-coincident and spaced microphones attracted attention. Of course their fundamental characteristics of coherence and spaciousness have been heard all along by engineers using every kind of loudspeaker. But heard in complete context through accurate monitors, it was their limitations, more than their merits, which were now more comprehensible. The conclusion: both are correct as far as they go, but they ultimately reach a dead end; neither is capable of producing a complete recording. The continued use of – and debate about – these two techniques for over fifty years is historical evidence that we desire both coherence and spaciousness in two-channel stereo recording. Does it really have to be one or the other?

The monitors demonstrated a comprehensive capacity, but none of the classical recordings I heard were able to take full advantage of it. Once again, it was the incompleteness, the absence of something that drew attention. The missing quality was realism. This might have been less noticeable but for the fact that everything else was so convincingly reproduced.

One wonders at this point if it is possible to capture the quality of realism. If not, then a truly complete auditory experience is not possible, and we will have to reassign our expectations to lesser outcomes. Even then, incompleteness will remain as an audible reminder; from what I heard, it cannot be glossed over or hidden.

3. A PLAN FOR REALITY BASED RECORDING

The only way to resolve this issue was to perform some experimental recordings to see if the missing realism could be recorded, using the monitors as a guide in the process. Any plan for recording must begin by choosing the aural perspective. The vast majority of recordings I've heard are based upon the podium perspective, or in close proximity to the stage. There are many reasons for this choice, but none are based upon what one is familiar with. The reference resides exclusively with the audience. Figure 1. To illustrate this, I have watched a few young, idealistic conductors begin a passage in rehearsal and quickly run out into the auditorium to judge the real effect. They know where reality exists, and it isn't near the podium, which an engineer can confirm by taking a similar walk. If it is good enough for audiences, then surely this is the perspective worth adopting.

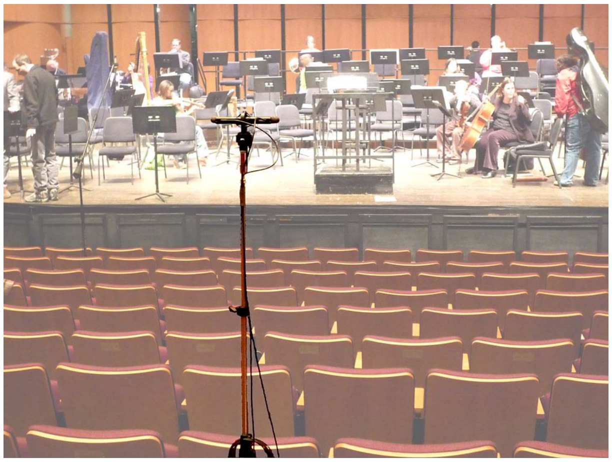

Figure 1: The front cardioid microphones capture the "source" from the audience perspective.

When considering microphone technique, several observations in section 2 provide a direction. It seems we want the apparent contradictory properties of coherence and incoherence. The question becomes: how to combine them beneficially? Here, an obvious division of labor into source and response suggests itself: a pair of near-coincident cardioid microphones, providing both coherence and directivity, to pick up the initial signal from the performers (the source) and a pair of spaced omnidirectionals behind the front pair to pick up the response of the hall (the response). This permits concentration upon two very different yet fundamental sonic scenes with separate microphone pairs that are optimized for each. Both pairs would form a source and response or S+R array, Fig. 2.

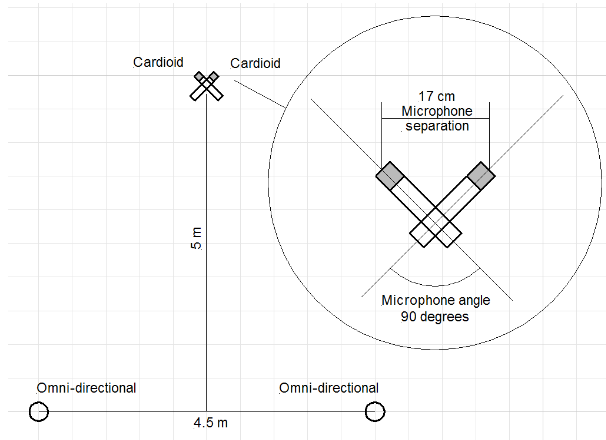

Figure 2: The S+R microphone array that separately captures the initial sound source and the response of the venue.

With guidance from accurate monitors, a preliminary plan was made to attempt accurate recording using these original specifications:

- The audience perspective will be honored; it can be recognized and confirmed. The sound also arrives pre-mixed at that location with no need for spot microphones, which is another advantage.
- The source will be picked up by a pair of near-coincident supercardioids, their location determined by walking back from the podium or stage during rehearsal until a fair representation of the entire ensemble is heard.
- The response will be picked up by two rear omnidirectionals decorrelated from the front pair as well as each other.
- The height of the microphones will be ear height, removing a variable during evaluation. The front pair will be pointed straight ahead; the rear pair will be pointed to the ceiling.
- The session will be recorded on a battery powered four channel recorder, allowing later mixing under controlled circumstances and of course using accurate monitors.

A series of experimental recordings were conducted at three locations in the Washington D.C. area: Washington National Cathedral, Schlesinger Concert Hall (a 1000 seat facility), and a local church. Ensemble sizes ranged from string quartet to orchestra and chorus. All were necessarily carried out during rehearsal with no audience, which is not the intended use of these venues, so the conditions were worst case.

4. RESULTS

These were the initial observations from the first two sessions:

- The two microphone pairs do combine constructively. Their mixed coherence and incoherence do not represent a foreign language at all.
- The live experience heard near the front pair's position was in all essentials replicated in the living room. Isolation of differences was difficult.
- The absence of a recording engineer's creative contribution was very noticeable and welcome.
- The stereo seat is of reduced importance to the overall auditory experience. The performers and performance remain in place off axis to a high degree. This was a welcome development, as it mimics the live experience.
- The results demonstrate the capacity that the two-channel stereo format is capable of. The response of the hall plays an enormous role in this result, to a degree that one wants to say "It's all about the response." Actually, this is true of the various music surround formats, whose sole purpose is dealing with the response of the hall; the source is almost a given. The surprise is that the same degree of importance applies to the two-channel stereo format as well, helping to transform it into a sound-field format, though of course not a surround format.

After the second session, I was confident that these results would be consistent, so all future sessions were dedicated to fine tuning and optimizing various details. The interactive nature of these experiments complicates their recounting, which follows.

4.1. The Cardioid Front Microphones

Supercardioids were originally used for the front pair, chosen for their directional qualities and excellent polar pattern. During mix-down, however, it was difficult to balance the front and rear microphones, and the blend was not as clear as anticipated. The reversed channel orientation of the supercardioids' rear lobes are at odds with the rear omnidirectionals' stereo image, such as it is. To see whether this was the explanation, cardioids were tried, using the ORTF configuration (110 degree angle between capsules, 17cm spacing between capsules). There was good improvement in the areas mentioned. In some later sessions, the supercardioids were tried again using a different angle and spacing, but with the same problems as before. Cardioid, then is the preferred pattern in this application.

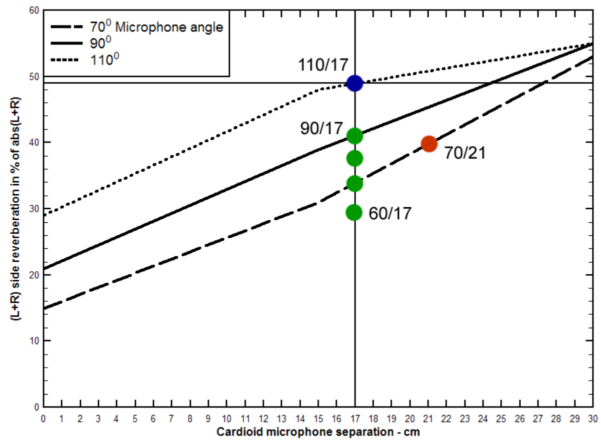

Figure 3: Left and Right Loudspeaker mono output signals relative to the total (L+R) output, as function of cardioid microphone capsule separation and included angle [5].

The question was raised as to how the ORTF and other potential configurations mapped the recorded sound-field into left and right loudspeakers. [3], [4], [5]. The S+R array uses two stereo pairs that must co-ordinate to produce the intended whole. A better understanding of the parameters of angle and separation of the capsules and their relative influence was necessary to make more informed choices when experimenting with the front pair. A graph was prepared which showed the ratio of mono to stereo output from the loudspeakers with a given configuration of microphone separation and angle. Figure 3. Using it as a guide, a 70 degree/21cm mounting was selected for its lower ratio over the

ORTF, and a test was made at the Cathedral comparing the two. Cathedral acoustics are not very representative, but the reverberation is long and memorable, providing an extended opportunity to evaluate this aspect in a recording. The result seemed to improve the quality of the reverberation and confirm the mono to stereo ratio to be an "active" parameter with this array, so a decision was made to decrease it further. Another mounting bar was made to 60/17 specifications and compared to the 70/21 at Schlesinger Hall, with some interesting lessons. The 70/21 came closest to replicating the live experience while the 60/17 sounded cleaner than was heard live and with a narrower hall than was heard live. This helped to settle several issues:

- The 60/17 bar apparently represents a step too far in narrowing the angle between capsules. The ORTF 110/17 by comparison could represent a limit when widening the angle since it was likely intended to provide as inclusive a pickup as possible. This is not necessarily our goal for the front pair, which must share the workload with the rear pair.

- The angle between the capsules is the most influential parameter. It establishes the width and size of the recorded venue, and must do so in a way that the rear omnidirectionals are able to complement. Next in influence appears to be the mono to stereo ratio of the configuration.

- It was decided to expand the 17cm family of mounting bars to include 70, 80, and 90 degrees for further experimentation. (These bars were made locally, following the Schoeps ORTF mounting bar design. They allowed precision mounting and quick changes from one to the other in less than a minute.)

There was one other notable lesson from this session. This venue lacks a center aisle, which makes walking back from the podium difficult, so half of the stage width - which seems to be a representative distance – was measured back from the podium to initially place the 70/21 pair. Seated next to the stand, the sound was not pleasant. It was eventually discovered that only one row back the sound was surprisingly more acceptable and the stand was moved there at the next opportunity. Later during mix-down, the difference between the two was just as noticeable as heard live. This underscores that there is no better guide than the ears when placing the main pair. It is normally a matter of meters rather than centimeters, but in either case, you have effective, predictable control over the results. It should be added, though, that raising the microphones to a height necessary to accommodate a live audience was not tried. It is not known whether the front microphone placement, for example, might be affected.

At the next Schlesinger Hall session, four 17cm bars were tested having 60, 70, 80, and 90 degree capsule angles. The 60 and 70 degree bars gave too narrow a presentation to be usable, the 80 degree displayed some promise, and the 90 degree bar gave the best impression of the hall. This result settled a few more issues. The controlling parameter of the S+R array is the distance of the front microphones from the ensemble, which is determined by ear at the venue. The experience to date shows that this distance is likely to be half the stage width or greater, and at this distance, arrays with a stereophonic recording angle greater than about 118 degrees (such as the 60, 70 and 80 degree bars) will pick up too narrow an image. For them to display a proper width, they must be placed closer to the ensemble, but the ear tells you that closer is not good; the sound is too rough, too raw, too in-your-face, lacking the civilizing quality of context. This is a recurring lesson from the project: if you don't like what you hear where you are standing, then don't put the microphones there and expect any improvement. Those microphones and your ears may not hear in the same way, but agreement on what constitutes hostile audio territory is not difficult.

Another session was carried out at the Cathedral to compare the ORTF 110/17 and 90/17 bars and to see if the effect of the mono/stereo ratio could be better isolated and understood. This time the results led to a good understanding of the interactions at work. The 90/17 pair provided a somewhat narrower venue than heard live, but the reverberation decay was unlocalized, just as heard live. The 110/17 pair presented a larger cathedral, but the reverberation tended to die between the loudspeakers. Of the two, the 110/17 pair was preferred; the larger venue was closer to what was heard live. The unlocalized reverberation decay of the 90/17 pair was ultimately not as important in this instance.

The 90/17 worked well at Schlesinger Hall. It's possible that 100/17, if available, would have been preferred. The 110/17 was preferred at the Cathedral; it's possible that 100/17, if available, would have been the best compromise. Obviously, a 100/17 bar will have to be made for future comparison to see if it possesses a universal characteristic, or if all three angles, 90, 100, 110 should remain available for consideration. At least the tradeoffs involved – width/size of the venue versus unlocalized reverberation - are now more clearly understood.

4.2. The Rear Omnidirectional Microphones

The rear omnidirectionals are 20 mm diameter, so they have a high frequency directivity related to that dimension. Therefore they are mounted vertically, pointed to the ceiling to reduce high frequency conflict with the front microphones while still maintaining their overall spectral balance. Also, the capsule's 90 degree axis becomes the same as the horizontal axis of the array, making it uniform in that plane through 360 degrees. The high frequency response of the capsules may need to be increased; there is more of this detail in the section on mix-down.

The original location of the rear omnidirectionals was 11 meters behind the front pair (about one wavelength at 30 Hz), where they were spaced 11 meters apart. Initial evaluation showed that the front-to-back distance was too great. The sound fields of the two pairs were too different to combine effectively. The rear microphones were moved forward in experiments (listening to both pairs individually and together), settling on 5 meters to the rear of the front microphones. The 11 meter spacing of the rear microphones was also too great, leaving a hole in the middle of their sound field, so similar experiments were made using 3, 4.5, and 6 meter spacings, with 4.5 meters being preferred. There were no further adjustments of these spacings in preference for other experiments, so these distances should be viewed as variable.

4.3. The Mix-Down

This is the crucial stage of the process, and fortunately the engineer has the key for success in achieving reality. Figure 4. Having been present near the front microphones throughout the session provides the memory that becomes the controlling authority during the mix-down, and accurate monitors placed in a normal room can confirm when the closest approximation has been achieved. The session was recorded on an Edirol R-4 four channel recorder and transferred to a laptop with Adobe Audition 1.5 software. Some details and observations about this process are grouped below.

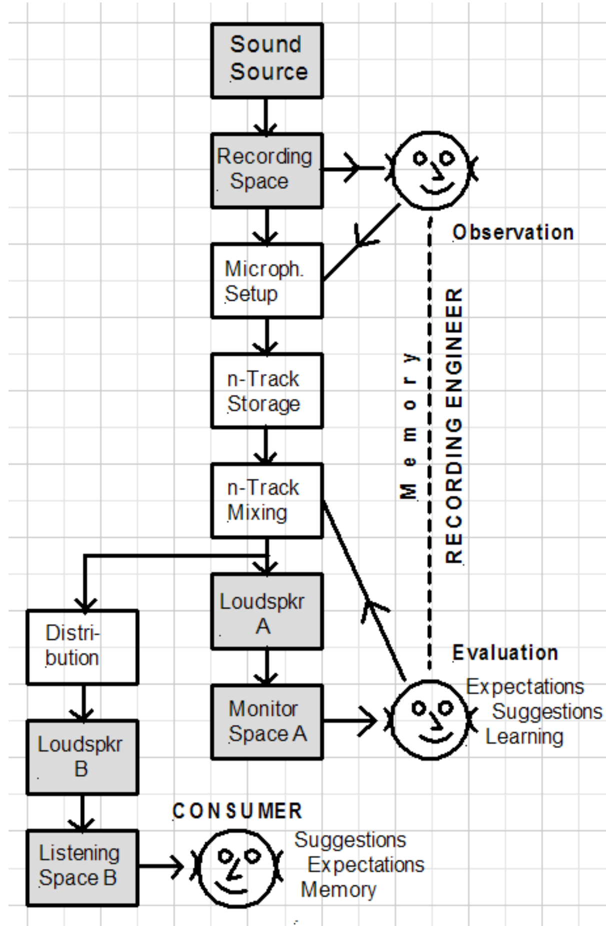

Figure 4: Process steps in recording and reproduction

4.3.1. The Omnidirectional microphones

The original omnidirectional capsules used were the Schoeps MK 2, which have a flat frequency response. It was later determined by experiment with equalization that the rear omnidirectionals must spectrally blend with the front pair, therefore requiring a high frequency boost to the MK 2. These were subsequently replaced with the MK 2S which has a 4 dB high-frequency rise, and it was found that even these benefited from an additional 2 dB of boost, with the frequency break point optimized by ear. Whether this combination ended up precisely simulating the MK 3, which has a 6 dB rise, is unknown. The MK 2 and MK 2S both require equalization in this application, and there could be a benefit from being able to choose the frequency and amount. Would different venues benefit from slight changes? For example, a Cathedral session was tried using as much boost as 5 dB in addition to the MK 2S response with quite interesting results. Both capsules are usable in this application, given shelving equalizers with an adequate choice of frequencies. The MK 3 was not tested here.

The degree of influence of these microphones' high frequency response was unexpected. Changing it can sometimes even be heard as equalizing the front pair instead, or at least affecting one's perception of them. It's an important influence on the overall perception of reality, so this is an area that requires attention.

4.3.2. The Cardioid Microphones

The cardioids capsules were Schoeps MK 4. Equalizing their low frequency roll-off required several attempts. I found that the most reliable technique was to set the shelving equalizer to its lowest frequency and add boost until it sounded right. Testing the mix with this equalization in and out of circuit confirmed that it should be in circuit. Recalling the experience with the omnidirectional microphones, I experimented with high frequency equalization of the cardioids as well, since they are also located at some distance from the source and a shelving boost of this sort can be relatively benign. Using a Cathedral session for the test, even a slight 1-2 dB boost was easily noticed as different from the live experience, and at the higher boost levels it became somewhat reminiscent of typical commercial recordings. It did provide a clarity that might be found preferable, so with difficult venues it could serve as an optional bridge between typical recording and reality recording. Finally, due to the cardioids' susceptibility to mechanical noise, attention was paid to shock mounting. It was found that adding a second shock mount, the Shure A53M, significantly increased the isolation from microphone stand disturbances. Figure 5.

Figure 5 Double shock mounting of the near-coincident cardioid microphone pair.

4.3.3. Balancing the S+R Array

The balancing process presented a novel experience that could be described as retrieving reality, which was considerably different from previous mixing experiences. It took quite a while to become acclimated to it. This process stage generated a number of details and observations:

- The balance between left and right channels of each pair should be checked first.
- The playback level should be the same as the experienced live level. More on this subject is addressed below.
- Attention should be paid to maintaining the overall level when making individual adjustments. This can be a nuisance depending upon the software involved, but is necessary.
- When searching for the optimum balance, the restrictiveness of the stereo seat can be a useful indicator. If moving away from the center causes too much disturbance or interference, this could be an indication that more from the rear omnidirectional microphones is needed.

The results from all mixes show that the contribution of the rear microphones is always less than the front pair. The difference can be described in this way: muting the front pair is always noticed (at the moment of muting); muting the rear pair requires greater attention to notice.

4.3.4. Monitor Levels

Adjusting the monitor level to simulate the live event is not as straightforward as I expected. During an experiment, I began advancing the level in 1 dB increments over the level earlier settled upon as proper. Each advance sounded "better": bigger, wider, more real in some way, taking over more of the room, each step suggesting that maybe this was the more correct level. This was continued until the level greatly exceeded the real event, but was still giving the same result. How to explain this? I've certainly played with levels before, but this effect was different: each increase sounded like another step in a gain path that had no limit. Weeks after this episode, I heard a tutorial seminar given by Jens Blauert [6] and particularly noticed this passage: "Spaciousness. When the orchestra in a nice hall plays something loud, then you hear something which is more extended than where you see the orchestra….and we know that this is caused by lateral reflections." And: "…..apparent source width…..we called it spaciousness in those days….goes up when correlation goes down." It struck a chord. Could something like this explain what is happening in my living room? Whatever the explanation, this behavior introduces a variable that one should remain aware of as it could influence parts of the mixing process.

4.3.5. The Role of Memory

The memory of the live performance is an indispensable feature of this recording technique. It provides the path back to reality during mix-down. Figure 4. The difficulty, it has been noted is that auditory memory can be fallible and fleeting. I have experienced examples of it during mix-downs. There are times when the original memory of this or that has become corrupted and is no longer reliable. Recognizing when it occurs is not a problem. But I've learned that these moments do not mark the end of useable memory from the session or that the time for accurate mix-down has expired. In my case, memory corruption usually involves a matter of detail, and often detail that I didn't pay enough - or any - attention to during recording, but now wish to consider. Well, that ship has sailed. But it doesn't mean that everything learned during that session is gone. I found that more important and consequential impressions remain, which days later can still be recognized when they are approximated. Think of this example: a voice not heard in years can be immediately recognized, even over a telephone, indicating a robust auditory memory. The auditory memory learned from being present at the acoustic event is much more a strength of this recording technique than a weakness. Absent this memory of the actual event, the real concern should be over what replaces it.

5. CONCLUSIONS

The low frequency behavior of the S+R array deserves separate mention. As a generalization, I find that omni based recordings can have too much low frequency response, in effect exaggerating reality. This is the sin of commission. Pressure-gradient or gradient based recordings are guilty of the sin of omission; they have the effect of miniaturizing reality. Thus the effect from combining omnidirectionals and cardioids was a primary curiosity concerning the proposed array. Mixing them provided the opportunity to compare this aspect of the two pairs, and the difference between them was hard to reconcile. Depending upon program content, the front pair appeared to have no low frequencies while the rear omnidirectionals sounded like they had nothing but low frequencies (I exaggerate). Yet when listening to each pair alone, both results could be accepted. Each sounded plausible in their own contexts so long as the alternative was not available for comparison. Well, the solution once again is to combine them. The mix from the S+R array serendipitously avoids both sins and does not draw attention to itself. This is particularly welcome because low frequency reproduction seems to be one of the earliest and most reliable indicators of something wrong or unnatural.

5.1. Envelopment

After listening to reproduced sessions that so closely approximated the live experience, the question left standing is: what is missing from the live experience? The obvious answer is envelopment. But at these live events, that quality did not seem to matter as much as I expected, and I think the reason is that live music is predominately – very uniquely – a forward biased acoustic event for the audience. The performers are in front of us, and they occupy our full attention. The S+R array enables the two-channel stereo format to replicate this forward biased experience very closely. It may be theoretically incapable of envelopment, but it does seem able to at least turn a corner. So I would answer the question this way: it is my sense that the major difference between the live and reproduced experience is the quality of freedom to the sound. There is no sense of restriction with live music. One's relationship to it represents a physical and perceptual freedom so integral to the experience - so part of our physiology - that separate notice is not taken. But over loudspeakers, historically, this freedom becomes noticeably truncated, always contrasting with the live experience. Recalling this quality from the live event when mixing might be useful. It could be the ultimate descriptor of achieved reality in recording. And the most difficult to describe.

5.2. Direct and Reverberant Balance

This advice circulated during my early years as an engineer: The first and most important thing to get right is the balance between the direct and the reverb. It was on my mind before the first session of this project, planned as a worst case test of this theory: in an empty cathedral, with a pair of near-coincident microphones improbably located 20 meters from the chorus, the ill-advised addition of a pair of spaced omnidirectionals placed another 11 meters further back would be welcome and improve matters, not worsen them. This theory proved true. So what does this say about the earlier advice? As insightful as it appeared then, it occurs to me now that it was derived from, and therefore its utility is still limited to, inadequate monitor loudspeakers. If the "reverb" is portrayed accurately, then the amount of it is no more an issue at home than it is when sitting in an empty cathedral listening to a rehearsal. In either place, it's just as enjoyable. It is the realism of the recorded effect that naturally and intelligibly incorporates the "reverb", reducing its likelihood of becoming an unrelated, quantifiable source of confusion.

5.3. Closed System Recording and Reproduction

This recording project might earn skepticism because it is the product of a closed system , which refers to the process whereby errors in the monitor system unavoidably affect the engineer's choice of microphone technique and mixing, producing a result unique to that circumstance and with no value beyond it. But what if the monitor system was, for the sake of argument, perfect? It would also be a closed system, but now beyond reproach. So the problem is not with a closed system (they must always be so), the problem is the level of contamination introduced by the monitors. The recording system described here serves as a control for this contamination by allowing the engineer to monitor both the live performance and its playback, a technique that was used when developing this loudspeaker. If both live and reproduced experiences are the same or nearly so, then inaccuracies are very low or nonexistent. The recordings from this project have a heritage of two distinct interventions, one by the loudspeaker design engineer and the other by the recording engineer, where referral to the live event was instituted as a check against inaccuracy. This is how the "circle of confusion" can be broken. [7]

During most of our waking hours, our hearing apparatus goes about its business with no difficulty at all and is only rarely - momentarily - surprised. The only time during the day that it is presented with confusing information is while listening to the stereo system, trying to deal with a constant source of foreign and contradictory information. The message of most recordings seems to be: here's an auditory experience you probably haven't had before, while the message of the recordings from this project is: here's an auditory experience that is very likely familiar to you.

5.4. Acceptable Recording Venues

The S+R array is intended to be maximally sensitive to the acoustics of the recording venue. It has to be if accuracy is the goal. The idea is to capture the reality of the event in such a way that when reproduced the brain can readily decode it. Does this mean venues without quality acoustics are even less usable now? Not necessarily. Obviously the more flattering acoustic is always preferable. But I recall a session at a local church with terrible acoustics and a super-busy highway just a few meters away. The recording sounded as unflattering at home as it did there, and yet it is surprisingly listenable because of its accuracy, which eventually wins trust and involvement. It's remarkable what the brain can ignore and what it can do for you if given a chance. Accuracy is such a durable value that it can overcome accurate reproduction of unfavorable acoustics. This could possibly modify what one perceives as acceptable acoustics for recording.

5.5. Further Experiments

This report is one example of how accurate monitors can be used to guide the recording process. The described S+R array is robust and relatively forgiving in some aspects. Its essential qualities were evident in the very first session, even though many of the details were yet to be refined. This suggests there is room for variation in some of the details. It is hoped that enough information has been provided to enable others to plan their own experiments with greater efficiency. If you would like a preview before beginning or to satisfy your curiosity, find a group in rehearsal, walk back from them until you hear a fair representation of the ensemble, and listen for a while. What you hear is what I have been describing. If you are not impressed by what you hear, you may be in an unfavorable acoustic environment. Or perhaps more likely, you have been listening to too many CDs and not enough to live music. This would be an example of how unreality in audio is perpetuated.

5.6. Artifacts have contaminated Audio

There is no question that the most indelible impression from this project came from recognizing the existence and influence of audio artifacts. Nothing is as effective at revealing and defining artifacts as their absence, at which point nothing that comes after can be the same. For example, it led to the realization that my professional audio life had been spent in a parallel universe of audio artifacts, rearranging or modifying them or reacting to them and not at all solving the puzzle of reality in audio. This project served to highlight two basic sources of artifacts and their influence upon the recording and playback process. The first source is the monitor system and its interaction with the room. The second source is microphone techniques. Only when the artifacts from the loudspeaker and room are eliminated can the recording engineer identify the nature and cause of the recording artifacts and intelligently reduce or remove them. The S+R array proposed here was only possible with the availability of accurate monitors. The goal was to eliminate the remaining artifacts in the recording process, which would finally satisfy this curiosity: what does an artifact free recording sound like? The answer: it only sounds like music.

6. SUMMARY

In Part 1 of this paper, an accurate loudspeaker is described, and its interaction with a normal room is explained. In Part 2, this loudspeaker is recognized and accepted as a standard for monitoring and used to formulate a four-microphone recording technique for accurate recording and 2-channel reproduction. Experimental recording sessions are described that produced this result: the live experience heard near the main microphones' position was in all essentials replicated in a normal living room, thus verifying the accuracy of the recordings and the steps taken to achieve it. The outcome represents more than just another alternative in a 50-year history of recording. It challenges current practices.

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