Tremblay, Pierre Alexandre and McLaughlin, Scott

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THINKING INSIDE THE BOX: A NEW INTEGRATED APPROACH TO MIXED MUSIC COMPOSITION AND PERFORMANCE

Dr Pierre Alexandre Tremblay, Dr Scott McLaughlin

CeReNeM
University of Huddersfield
Queensgate Campus
Huddersfield, HD1 3DH
United Kingdom
p.a.tremblay@hud.ac.uk
s.mclaughlin@hud.ac.uk

ABSTRACT

The Thinking Inside the Box project (TItB) seeks to address pragmatic concerns inherent to mixed music performance, and proposes ways to better consider the sound of the acoustic reality of the concert space at studio composition time. This is achieved through empirical investigation into subversive use of recent developments in hardware and software technologies.

The primary concerns are (1) optimising the integration of live instruments and electroacoustic sound in the concert hall environment for both the performers and the public, by carefully choosing loudspeaker types and placement at commission time, and by avoiding sound reinforcement; (2) minimizing for studio composers the in-situ trauma of the first live rendition of the piece, by bringing the concert hall acoustic environment into the studio composition process, using convolution reverb to reproduce in the studio the given loudspeaker setup through its impulse responses.

This paper presents the conclusions of the project's early experiments in the form of three case study sets, and describes how this approach will be of use for any composer of mixed music.

1. INTRODUCTION AND CONTEXT

Taking a work of mixed music from the composition studio into a concert hall is an act of translation, from one acoustic reality to another, and translation always involves alteration. Thinking Inside the Box (TItB) proposes some practical approaches to making this translation easier, by altering some ways of working at the outset of the compositional process. It also addresses the difficulties that mixed music can create for the performer when the piece is brought into the live environment. It is important to note that throughout this paper we define mixed music as a straight translation of the French term *musique mixte*, where it implies music for acoustic instruments with live processing and/or fixed media.

For this project, we have assumed a very simple mixed music concert setup, with a single performer and a limited number of mostly frontal loudspeakers. Our assumption of a less complex auditory field allows a better assessment of the blending, as minimizing the number of elements in the auditory scene affords us greater powers of discrimination with reference to the sound environment, as implied by Bregman's research on auditory scene analysis [3: especially chapter-5]. We deliberately set the most discriminating blending condition to be able to assess it thoroughly.

Historically, mixed music has suffered from a dichotomy between the sound of what is composed in the studio, and its subsequent rendition in the concert hall. Our primary concern is the problem of blending the electronic sounds with the sound of the live instrument: an effect that is often sought in this type of music, yet rarely convincingly obtained in the concert hall, even under the best presentation conditions. While it goes without saying that many seminal and excellent works have been written despite this issue, mixed music performance and composition can be brought to a new level of fluency by taking advantage of current developments in hardware and software.

We believe that the dichotomy stems from two main sources; (1) music destined for live performance is composed in the studio, an acoustic environment that is radically different from that of a concert hall; (2) instruments and loudspeakers have different modes of sound production that undermine their ability to blend in the concert hall. We address these problems using commercially available new loudspeaker designs that radiate sound rather than using axial projection, and software advances that allow desktop computers to run many instances of convolution reverbs using impulse responses (IRs). The methodology of TItB includes case studies on an existing piece of mixed music, two sets of purpose-written composition studies, and a full-length commission written using this new approach.

2. PROBLEMS

The main source of the problem of blending live instruments and electronic sounds in the concert environment lies in the major differences in the manner in which loudspeakers and instruments excite the room's acoustics; as has been noted by previous researchers (Caussé et al) [4]. Traditional axial-firing loudspeakers project sound along an axis, while instruments are, more often than not, radiating sound sources whose energy is emitted in many directions and patterns. It is our assertion that addressing this difference is often overlooked by composers during the studio composition process. This dichotomy will be discussed below in relation to three specific and pervasive issues in mixed music performance:

1. The non-portability of the stereo standard from the studio into to the concert hall: "the sweet-spot problem".
2. The position and type of loudspeakers, and the palliative use of amplification of the instrument as sound reinforcement.
3. The need of foldback monitoring to cope with the problems created by sound reinforcement.

2.1. Problem #1: the Sweet-Spot

Problem #1 is the main issue at composition time. The music is composed in the highly controlled acoustic of the studio, with the composer sitting in the sweet-spot and perfectly aligned to perceive all the spatial detail in the stereo image. However, in the concert hall most listeners are not positioned so perfectly and lose much, if not all, of the spatial detail composed in the virtual stereo image: the more the listener is off axis, the more the image is distorted. Psychoacoustics informs us [15: pp.92-94] of the very limited efficiency of the stereo image for in-hall representation: the interaural time difference localisation cues will induce problems for anyone who is off-axis by a difference smaller than the width of the human head. Further than this distance, the precedence effect implies that all other sources will be dismissed as early reflection. This means that even slightly off-axis listeners will hear an amplified instrument—typically sent to all frontal loudspeakers—and every object in the stereo mix that is in both loudspeakers, as coming from the loudspeaker nearest to them rather than from its actual position or its intended position in the stereo field.

Simon Emmerson notes the problems that stereo can bring to mixed music performance:

> *The virtual imaging of two (even high quality) loudspeakers at the periphery of an auditorium is insufficient. Any off-centre listener will lose [...] the sense of the performer being the source* [6: p.95].

When the composers are in their composition studio, they are always in the sweet-spot and as such the position of the listener is not a variable that they are obviously presented with, therefore this is often not factored into the composition. Léo Küpper's research suggests that under studio conditions the listener can differentiate between up to 70 positions [11: p.290], and most mixed music composers will be tempted to use a subset of these in the composition studio. Considering that in most halls the majority of the audience will be off-axis, this is a major problem for the live rendition of a studio composition.

Much fixed-media music (acousmatic and other practices) circumvents the sweet-spot problem by diffusing the music across multiple loudspeakers surrounding the audience. This situates the listeners inside the sound rather than projecting it from a stage; using the desk as an instrument to multiply the inner space of the studio mix into the real space of the concert hall, the diffusion artists will do a performed distortion of the studio image, as justified by one of its main defendants:

> *Events carefully oriented by the composer within the space of the stereo stage will simply not be reproduced appropriately in a large concert space unless something more radical is done* [9: p.121].

However, diffusion is much less successful once a live instrument is introduced into the equation. An acoustic instrument in a diffusion context either chokes the diffusion by virtue of being an immobile point source, or is simply drowned out by the electronics: diffusion artists may see the problem as lying with the instrument, as Christian Clozier states; "mixed music [...] militates against and neutralises creative diffusion" [5: p.233]. The acoustic instrument is alien to the diffused sound by being, effectively, a native of the room sound—as described in the following section—so its excitations undermine the loudspeaker created illusion even for people on-axis. TItB proposes ways in which the composer will be able to consider at composition time the acoustic reality of the hall, to ensure that the spatial content of their music can be perceived by every listener in the room, however off-axis they are.

2.2. Problem #2: Sound-Reinforcement

Problem #2 follows from Problem #1 when at concert time the intention is to recreate the studio sound in the hall. The loudspeakers are chosen with this intention and usually positioned in a wide pattern in front of the performer to optimise the stereo image, but only for those fortunate enough to be in the sweet-spot. Most of the audience will be closer to one of the loudspeakers than to the performer, thus creating the acousmatic dislocation presented above. At the same time it also creates problems of balance and localisation for the performers, as they are behind the loudspeakers.

This tendency to use sound-reinforcement can be explained by the fact that, in the composition studio, most composers usually work with a very good quality recording of the instrumental part for their mock-up. Recording reduces the instrumental sound to a stereo-field reproduction, which appears in the studio to blend successfully with the electronic part because both are reproduced through the studio monitors. With an acoustic instrument in the live environment, this blending may not be as successful due to the differences in acoustic modes of production between loudspeakers and instruments. As Simon Emmerson stresses in Living Electronic Music:

> *The electroacoustic part has in all likelihood been created without room ambience information, whereas the amplified instrument (however closely miked) is bound to take something of its space and throw it back, making integration of the two very problematic (if that is the aim)* [6: p.105].

The traditional solution here is to amplify the acoustic instrument. This reduces the live instrument to an electronic representation, and returns the music to the way the composer would have heard it in the studio—everything through loudspeakers—but in doing so creates a set of new issues. Through amplification, the instrument's rich radiating sound is reduced in richness. This sound reduction is very useful for the blending, as it moves the instrument from the room excitation mode into the sound projection mode typical of loudspeakers. Yet it can also be very problematic for the performer, as it alters the spatial and timbral characteristics of their instrument, leading to frustration and an impoverished performance environment. For instance, pianist Sarah Nicolls commented how amplification of the piano detracted from the usual intimate relationship that she enjoys with the acoustic instrument [14]. Sound reinforcement also introduces a cognitive dissonance for the listener, generated by the dislocation of visual and audio cues—between the visible position of the performer and the apparent position of the performer as implied by the audio cues, especially for off-axis listeners. This artefact has been previously noted by Simon Emmerson [6,7], and by Trueman et al in the Nbody Project [18].

Note that this set of problems does not apply to all types of amplification, as it is often successfully used as an effect or process in itself. They arise when amplification is used as sound reinforcement. We believe that, in the context of chamber music, sound reinforcement should be avoided as much as possible, as it has many effects on the experience of performers and listeners.

2.3. Problem #3: Foldback

Problem #3 is a by-product of amplification and loudspeaker positioning (Problem #2), as they often introduce the necessity for foldback monitoring, a solution described by Simon Emmerson as "inaccurate, distracting and interfering" [6: p.33].

When the performer is on a plane with or behind the loudspeakers, they are placed significantly far from the sweet-spot: their experience of the electronics is often dulled and distant sounding, with a poor or incoherent spatialisation and mix. To alleviate this problem, the standard approach is a foldback monitoring system that provides the performer with a sub-mix of whatever they need to hear projected at them through their own dedicated loudspeaker. This is reductive by nature as it must often present the performer with their own sound as well, at a volume level sufficiently high to cut through the level of the actual ambient sound. The performer's sense of the their own sound, as well as the overall sonic image, is therefore even more distorted.

The performers must also relinquish control of their place in the overall music to a remote mixer. Foldback can impair the performers' ability to hear both themselves and other elements of the music, leading to alienation and frustration on their part. As chamber music performers are accustomed to adjusting the balance of the ensemble with what they actually hear on stage, being uncomfortable with their sound and/or the ensemble sound will impair their focus and enjoyment of the music, no matter how dedicated they are to the genre. For the chamber musician, the compromises can be intolerable, as described by violinist Mari Kimura:

> *I feel quite helpless as a performer playing with tape in concert situations, especially in terms of ensemble and sound quality* [10: p.71].

What TItB proposes is to bring the qualities of acoustic chamber music to mixed music. This should result in a local blending of sound that is a richer sonic experience for the audience, and allows the performers greater control of both the local and global sound: a point also proposed by Emmerson [6: p.96].

3. PROPOSALS

The above presentation of the three problems may appear to be a caricature of the mixed music experience, but even when carefully managed by the concert team, these problems are ever-present at varying levels of seriousness, and all contribute to a reduced experience for composer, performer and audience. TItB proposes some ideas that may assist composers and performers in going further to improve the concert experience for all, as we assume that the live performance is the apotheosis of this genre. The working hypothesis of TItB has two main proposals:

1. A flexible and dynamic approach, at commissioning time, to loudspeaker type and their placement on stage. It must respect the textural demands of the music in order to militate against the problems described above.
2. Because these suggestions alter the format in which the music is projected in relation to the standard composition studio sound system, we

propose the use at composition time of impulse responses of the loudspeakers in position in the live space. This allows composers to work with the reality of the sound in the hall as a reference, rather than always within the perfection of the studio.

We do not view these proposals as a panacea, nor are we suggesting that composers should abandon the wonderful spatial precision that the studio can provide. Instead, we are dealing with the pragmatic concerns of mixed music that demand a different way of working for different outlets, be it a studio recording or a chamber-music concert. To maximise the effect of these proposals, they need to be acted upon at the earliest stage of composition: the planning and conception of the work.

3.1. Proposal #1

The first proposal of TItB involves reconsidering the type and the position of loudspeakers used in mixed music setups. This presents a solution to all three of the problems listed above by reducing the disparity between the modes of sound production used by instruments and loudspeakers respectively, and by placing the loudspeakers so that the performer can hear themselves within the whole mix, as they would in acoustic chamber music.

Using full-range radiating loudspeakers directly on stage—such as the Bose L1 'Personal PA' or the Bellecour Sensations—confers some advantages in music where the blend is important as they are a closer match to the radiating characteristics of the instruments. The use of "localised sound diffusion devices modeled on […] radiating properties" was also suggested by Misdariis et al in their 2001 paper, "Radiation Control on Multi-Loudspeaker Device: La Timée" [13] but we consider that using commercially available systems allows a greater portability [17].

Moreover, TItB has used configurations where the loudspeakers are placed near-behind the musicians, at volumes sufficiently low that the acoustic instrument need not be amplified; this absence of sound-reinforcement removes the problem of acousmatic dislocation. By placing loudspeakers nearer to the performers we also create the opportunity for a ensemble-type localisation of the sound, and allow the players greater 'control intimacy', as proposed by Simon Emmerson in his writings on the concepts of local and field. [6,7] This also removes the need for the problematic foldback.

3.2. Proposal #2

The second proposal reflects the need for the composer to work with the acoustic reality of a live environment rather than only with the un-portable perfection of studio's virtual stereo image. This reality of in-situ acoustic distortion is made even worse by our Proposal #1, as the use of non-conventional loudspeaker configurations will affect the possibilities of the sonic image. Even at the sweet-spot, Küpper's suggested 70 virtual points of the stereo field [11] will not work, as we do not respect the conditions of the stereo standard: being at the tip of an isosceles triangle with two loudspeakers.

To that end we have proposed that the composer worked their mix through virtual loudspeakers as if they were in the hall. To do so, we have made stereo (binaural, XY, ORTF and AB) and multichannel (3/2.0) IRs of specific loudspeakers at specific points in the hall, both on- and off-axis. We propose to then route each loudspeaker bus of the mix in a convolution reverb to recreate how it would sound in the hall. This allows us to use the loudspeakers as different point sources at composition time, to test on- and off-axis portability of the mix, and lessens the temptation to rely on the illusion of the stereo plane.

This reality-check approach is quite similar to the practice of studio popular music producers and engineers who test the robustness of their mixes by playing them through a cheap set of home stereo loudspeakers.

4. CASE STUDIES

In order to test the TItB hypotheses, we engaged in a series of case studies. The first came about through the combined fortune of the same piece of mixed music being played twice in one month in the same venue and by the same performer, allowing us to observe the differences between a standard mixed music loudspeaker setup, and our proposed one. The second case study was a set of mixed music etudes, composed specifically to test various concerns raised by the project. The final case study is full-fledged commission of mixed music written under TItB proposals, to assess their performance in the reality of a genuine composition project.

In each of these case studies we had listeners sitting on-axis and off-axis to observe and document their impressions. For this paper we will refer to the listeners by their position and use the same nomenclature for each study.

- Listener-A: on-axis and near to stage
- Listener-B: off-axis and near to stage
- Listener-C: on axis and far from stage
- Listener-D: off-axis and far from stage

Moreover, all the supporting material (score extracts, audio recordings, and impulse responses) is available online at:

`http://eprints.hud.ac.uk/4081/`

4.1. Case Study #1: Same Piece, Two Setups

Our first case study involves two performances by Anton Lukoszevieze of Matthew Adkins' *Between Lines* [1] for cello and fixed media. Adkins describes the piece as "intimate" and aimed to create a "meta-instrument" from the cello and the electronics [2]. Obviously the perception of blend between the two is paramount here, and as such the piece makes an excellent study for TItB.

The first performance featured a standard public address setup (PA): two axial-firing loudspeakers positioned in front of the amplified performer and at the widest edges of the stage; the performer had a foldback monitor. The second took place in the same hall a month later, and used two Bose L1mk2 radiating loudspeakers placed close to, and behind, the performer.

In the conventional PA setup, the perception of blend on-axis was acceptable but the cello often seemed quieter and smaller than the electronics, probably due to the differences in modes of sound production and sound reinforcement. Off-axis, the blending issue became much more problematic and often the electronics overpowered the instrument. Moreover, the acousmatic dislocation was terrible as, due to the precedence effect, the source position appeared to be from the nearest loudspeaker.

In the second concert, using the radiating loudspeakers, the music seemed much better served, as the composer's testimony shows:

> *Rather than replicate a studio setup with both electronics and cello coming through the loudspeakers, [this] presentation of the work seemed to accept the natural acoustic of the live instrument and present this and the electronics within the acoustics of the building. To my mind this was a much more successful concert presentation of the work* [2].

The blend was achieved through carefully considered loudspeaker positioning. They were placed a short distance behind each side of the performer; forming a narrow triangle. This meant that there was no true conventional stereo image, but the spatialisation in the tape part was still evident to the listener, although narrower. As the composer intended, the close blending between the instrument and the electronics blurred the identity of both, allowing a delicate ambiguity: some listeners in fact assumed that there had been live processing of the cello, and some others, sound reinforcement.

Composer Bryn Harrison commented that hearing *Between Lines* in the TItB set-up had "opened up new possibilities of mixed music" for him. Harrison, a composer who has not worked with electronics in the past, explained that in previous experiences of mixed music concerts he had always found the electronic part to be lacking in "depth of presence", and that in Adkins' work under TItB conditions "the electronic sound was not compromised"; that the electronic sound had become part of the acoustic environment [8]. The performer too was more satisfied with this setup, saying that he felt "more inside the sound" [12].

4.2. Case Study #2: Composition Studies in Blend

The second case study was a set of composition etudes written specifically to test certain aspects of blending the instrumental and electronic parts. The loudspeakers used were all placed on stage with the musicians. There were three Bose L1 for centre and near left/right, and two Meyer UPJs for wide left/right: the five loudspeakers formed a slightly curved plane just behind the performers. These etudes were composed for clarinet, cello and piano, each as solo instruments, and then as a trio. We chose these instruments as they present three different modes of sound production.

These initial case studies were successful in that most of our expectations were met and they allowed us to review the theoretical aspects in light of experimental observations, leading to case studies 2.5 and 3. The performers all described a greater sense of being "inside the sound" and likened the experience to playing with a chamber group: clarinettist Heather Roche states that she "responds to the electronic part as if it was another player", as opposed to her previous experiences of working with live electronics where she felt "overwhelmed" that she had "nothing to react to" [16].

We were able to get a proof of concept with regard to improving blending of instruments and electronics for all listener's positions. The sweet-spot problem was greatly reduced through not amplifying the instrument, and through careful loudspeaker placement. Audio-example-1 shows the clarinet playing the Blending Study (score available online); where consecutive individual lines are stacked into a harmony using delay lines. Listener-D described this example as "very convincing, especially as it becomes more dense in the later sections". This study also confirmed the hypothesis that blending would be especially effective in the denser sections, in reference to Bregman's work on auditory scene analysis [3]. The effect of greater density can also be seen even more effectively in audio-example-2 where the full trio plays the Blending Study, especially towards the end of the example where the full trio plays with many layers of itself.

However, we encountered a few issues in this case study. We found that blending was limited in some ways by not amplifying the instruments. However, our test group thought that the overall sound quality, improved by having a more natural, performer-biased approach to acoustic balance, was worth the compromise.

Another problem was a general issue with EQ, as the Bose radiating loudspeakers have a less linear response than our test projecting loudspeakers. This was audible especially when using live sampling, although at times, when there were no visual cues, it was impossible to tell which sound was the live instrument and which was the electronics. This is showcased in audio-example-3, a simple delay-based test. Though this problem was partially addressed through quick EQ correction at the mixer, further exploration into calibration of the microphone-loudspeaker audio-chain will be needed in order to achieve a more transparent reproduction.

There was also an issue with live sampling of the piano, as using a single microphone for recording led to a narrowing of the piano radiance, and the resultant electronic reproduction lacked richness: audio-example-4 demonstrates this clearly, especially between 0'47" and 1'00" where there is a phrase with piano and electronics followed by a phrase with electronics alone. Stereo microphone capture and stereo radiating loudspeakers would alleviate this.

The last issue brought to light was that the electronics sounded slightly distant relative to the live instrument: the solution to this is presented in section 4.3 below.

4.3. Case Study #2.5: Further Studies in Blend

In reflecting on case study #2, we found that there were some aspects that we had not thoroughly examined and some that we wished to re-examine. Specifically, we wished to compare the responses of different brands and type of loudspeakers, and to further explore their effect for on-axis and off-axis listeners.

This time we would test only one composition, and with three different loudspeaker setups: Meyer UPJs, Genelec 1032As, and Bose L1s. Three of each loudspeaker were positioned in an LRC configuration behind the performer, on a gentle curve. In all the files of the audio-example-5 folder, the same musical example (section 2 from Case Study 2.5) is used to compare the three different sets of loudspeakers, both live and as mock-ups through IR, and both on- and off-axis.

The general conclusions from this experiment were that the loudspeaker choice greatly affected the overall blend quality, with compromises to be made between the quality of sound reproduction and the ability to fill the space convincingly. As would be expected, the Genelec studio monitors had very high quality sound reproduction for the on-axis listeners. But as soon as the listener was off-axis, they acted as point sources and failed to fill the space between themselves. This highly focused sound projection is definitely an asset in the studio, to minimise the early reflections at the sweet-spot, but in the concert hall, it has a thinning effect for everyone outside the axis.

The Bose loudspeakers produced the most natural lower mid-range response and filled the space well, acting much less like point sources and providing convincing blend. However, they had less clarity of spatialisation, and their colouration made the blend with acoustic instrument uneven. In more declamatory passages, the instrument stood out too much but in slow sustained passages, the blend was very convincing, especially to more distant listeners: listener-D noted that at several points in the slow section he could not distinguish between the live clarinet and the processed sound without looking to see if the performer was playing.

The Meyer UPJs were most convincing as they struck a balance between point-source clarity and filling the space with a convincing blended sound: their design for live sound, with wider radiating patterns than studio monitors, could explain why they managed to fill the space despite them not being radiating loudspeakers as such. Their almost linear response was noted as more pleasant than the Bose by listeners in all positions.

We finished the day by testing the piece in a standard mixed music setup with a stereo pair of Meyer UPJs placed wide and in front of the performer, and with the performer being amplified. This allowed us to directly compare our approach with the standard setup. In this configuration the piece sounded very different and slightly unnatural, largely because it no longer had the chamber music quality that the other configurations achieved. The performer noted that she felt more uncomfortable and less sure of her sound in terms of balance, dynamics and tone. [16] There was also considerable audio/visual dislocation even for an on-axis listener: the clarinet sound shifted erratically between appearing to originate from the instrument (centre-stage) or one of the loudspeakers (wide left/right). These observations are all consistent with the problems described above in section 2.

4.4. Case Study #3: a Commission for Sarah Nicolls

The third case study is a commission from Sarah Nicolls to one of the authors: *Un clou, son marteau et le béton* uses a combination of fixed media and live processing. The composition of this work has cast more light on the project's second proposal as it was a real-life composition exercise, not just focused studies, thus providing a more involved perspective on the dichotomy between the intended studio-composed spatialisation and its in-hall perception.

Applying TItB Proposal #2 at studio-composition time meant that the studio/live dichotomy was reduced by the use of the virtual hall in the composition studio. Early collaborative sessions with the performer allowed testing of some musical gestures in blending, successes here allowed the composer to be more daring in the studio.

This study allowed us to build on our experiences with the previous two studies. We learned from Case Study #2 that there could be a point-source effect that added a slightly distant quality to the captured sound. In the mock-ups for Case Study #3, we compensated for this by using another IR for the instrument, to place the virtual instrument three feet in front of the electronics in the mix.

What was found to be a more important issue was that in composing through the IRs all of the time, the pleasure of studio composition was eroded. Much of this pleasure comes from the precision and control possible in that controlled environment: in general, this music is composed for live performance, but this should not preclude or limit subtleties in the studio composed space which, although they would be lost in the concert hall, would be very effective in personal listening conditions. This therefore implies a 'studio version', for an eventual album release, and a 'live version', where the mix is blunter, thus more robust and better able to withstand the vagaries of the concert environment.

To achieve this, we had to amend Proposal #2. The use of the virtual space at composition time—through the use of the IRs in convolution reverbs—should be considered as a reality check rather than an omnipresent filter through which the composer should work at all time. This subtler proposal allows the composer to assess the portability of their music between the studio and the concert hall, yet leaves the studio composition pleasure intact. It should be noted at this point—after amending our initial working hypothesis for Proposal #2—that this proposed protocol does not require that the composer have at their disposal custom-made IRs specific to every concert halls anymore. We have noticed that a reverb that approximates the characteristics of different position in a given hall, such as Altiverb's source positioning algorithm, gives the composer enough of a reality-check to enlighten them as to what will or will not work under live conditions. As IRs for many different halls are already available, composers should be able to find one sufficiently similar to the acoustic of their proposed concert space, and we are happy to add to this with our own repository.

The concert performance of *Un clou, son marteau et le béton* shows that the refined Proposal #2 was successful, as almost all of the blending that worked in the mock-up was equally effective in the live concert; see audio-example-6. Listener-B noted that the piano and electronics sounded as a single sound mass, except for those few places where explicit spatialisation effects were desired by the composer (score sections C and E), and in these places the effect had been successful. Listener-A (seated on-axis and three rows from the front) described the sound as having a definite chamber music quality where the electronics and piano sounded as two equal instruments.

The only prominent exception to this involved sections of the piece, such as section D, where sound was projected from loudspeakers placed under the piano, firing upwards at the piano soundboard. This blended less well live than in the mock-up: see audio-example-7. We believe that this is due to the different acoustic quality of such a placement, which was not taken into account in the mock-up IRs: an IR of a different colour should have been used to simulate the loudspeaker under piano.

In addition to this, the live concert re-emphasised that great care had to be taken with microphone choice and placement. In the concert, there was more disparity between the timbre of the acoustic piano's bass tones and its live-sampled counterpart than during the early composition tests, as the microphone chosen for sound capture (cardioid) was not the one tested in rehearsal (omnidirectional). The proximity effect of the former had not been taken into account, and because the piano is not amplified we need the loudspeaker rendition of the piano sampling to sound as much like the real piano as possible, thus microphone choice and placement are of the utmost importance. Again, EQing could have helped, but rehearsal with the actual concert equipment would have highlighted the issue at the source.

5. CONCLUSIONS

So far in TItB we have been largely successful in achieving our goals of addressing some pragmatic concerns of mixed music. The most pressing point for us has been to place the emphasis on mixed music as a live art, and not simply a studio art that is occasionally brought into the concert hall. To this end, we have proposed that composers conceive of their works as destined for the live environment, and live performers, from the beginning and thus stave off potential live issues from the outset. We have suggested changes to the arrangement and type of loudspeakers that takes into consideration the reality of the performers' and listeners' real-life experience. In addition, we have also suggested using IRs of these loudspeakers in position in the hall as tools to show the composer how their work may sound in the live environment and how to improve it before the usually disappointing premiere. The most positive result for us is that, when mixed music is composed under TItB proposals, the composers have focused on what is gained from the mock-up at the premiere, more than what is lost from their studio version. This reinforces our central point that mixed music must be composed with the concert hall in mind.

We are delighted that we have received positive feedback from all sections of the listening sphere. Composers, performers and most importantly, audiences, have responded enthusiastically to the music presented in the case studies above.

Looking to the future of TItB, we have commissioned four other experienced mixed music composers to write for string quartet with our specific setup of loudspeaker type and placement, and with the constraint that no sound reinforcement is to be used. For this project we will use a 4/2/1.0 setup, not dissimilar to 7.0 surround sound, and we will provide the composers with IRs (binaural, ORTF and 5.0, both on-axis and off-axis) against which they may check their mock-up mixes. To counter the less than linear response of the radiating loudspeakers, we will also explore calibration filters, probably using FIR, as it seem to become more prevalent: amongst others, Misdariis et al used this with *La Timée*, by applying a transfer function to their sound capture [13].

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