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Tuning to Trust: System Calibration as Creative Enabler

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ABSTRACT

This paper presents a mixed-music composition methodology emerging from the author's latest practice-based research in the field over the last five years. The calibration of the interactive performance systems has enabled trust in reproducible sound quality for both the composer and the performers, enhancing the portability and adaptability of the works, and permitting increasingly daring creative experiments without compromising the rehearsal and concert experiences. A set of general, transferable responsibilities and solutions are presented and assessed against clear design criteria in the author's latest pieces.

1. INTRODUCTION

Sound system calibration has been rising in popularity as an important consideration in the last 30 years. From the basic 31-band public address sound system (PA) EQ-ing to the now omnipresent dual-channel FFT analyser, and with major improvements of PA loudspeaker design over the last few decades, there has been quite a significant improvement in reproduction quality of music in live settings[1]. That is, when the concert producers could afford the expertise and the equipment, which is the case mostly for large-venue, high-budget music gigs.

In the scenes with more modest means, many studio enthusiasts and audiophiles have explored room correction [2] [3] along the same line, with a first generation of convincing commercial plugins for this purpose being developed a little more than 10 years ago [4], and since then have proliferated [5]. However only in recent years have major reference loudspeaker manufacturers included assisted calibration procedures in some of their products, in effect including DSP in their design to capture and correct errors in the monitoring environment [6] [7].

The advantages are indisputable to the author: tighter, clearer, more transparent sound reproduction allows for a better listening experience for all… who can afford such devices. Usually designed for stereo systems, these system's channel count is usually modest, affording at times as far as commercial multichannel formats up to 7.1, but rarely higher, and their cost is prohibitive. If these devices were ever used in multichannel chamber mixed music, this has not been documented and is probably limited to the very few research centres with permanent facilities, dedicated production teams and large budgets.

At least this was the case until the HIRT [8] became available, which has successfully provided the tools for tackling such issues to the masses. The author has used these tools systematically to great success, even when acoustic instruments share the stage with loudspeakers. Whilst doing so, new compositional possibilities emerged fortuitously: the systematisation of the calibration process allowed an increased trust in the portability of the performance systems, which in turn enabled more and more daring and nuanced uses of real-time processing.

This paper will succinctly share and link the consecutive experiences that built the author's considerations as they emerged, and their surprising conclusions and discoveries, to then share specific examples of fine-tuned methodologies and transferable good practices for this field, followed by further explorations to be undertaken.

2. A NEW TECHNE TO SUBVERT

First, a caveat: the author strongly believes in reflective practice as research, with constructive, challenging, creative cross-pollination of aesthetic considerations and technical constraints. The techne of mixed music (amplification, loudspeaker placement, live processing, creative coding, rehearsing, etc.) is considered as a similar body of knowledge to orchestration, and we can observe that their respective history follows a similar path: once a separate technique and set of constraints and conservative recipes, its integration as an intrinsic part of the composition process has allowed to use and abuse such constraints to widen the musical palette. For instance, the use of orchestral forces to personal effect in late Romantic music illustrates the gains of a synthetic approach to a techne and an art. Mixed music has reached that tipping point in its history a few decades ago, as previously argued in [9] and [10], but the essential practice of the 'musical assistants' and other oft-exploited technical aides were never documented in an 'orchestration' book [11]. One still has to learn by observing one peers' practices.

[1] The preface to the excellent 'Sound Systems Design and Optimization' by Bob McCarthy [1] gives a first-person account of this journey.

Through touring in suboptimal settings [12] and running electronic music festivals for two decades, amongst which the ICMC 2011, the author has seen many practices, yet very few good ones. Proposing ways of bringing these suboptimal conditions in the studio [13] was a good start to consider the live context during the compositional methodology, but where the breakthrough documented in this paper happened was through the extensive use of loudspeaker corrections [14] in the hall: suddenly there was a reliable fixed routine that allowed reproducible sound quality, swiftly and affordably. Whilst some music strives in the variability of sound reproduction, from different acoustic contexts and/or technical setup, the aim of improved consistency is in line with most studio composer's ideals of portability of mixing and orchestration decisions, and such reliability has been the driving ambition for the last decades of research in PA system correction.

The most unexpected conclusion along the way was that such reproducibility enabled trust, which in turn enabled better music: trust from the composer at the portability of sound design decisions; trust from the performers that the music would respond in similar ways to their actions. This allowed for a better chamber mixed music experience for all, allowing more subtlety, more rehearsal time, and more freedom: the composer could afford more daring explorations of potential processing in subsequent works; the performer could explore more daring gestures to push against the system, instead of the usual anxious hope that nothing will crash before the end of the concert.

Examples of such positive settings are thin on the ground[2]: our field is indeed in need of a foundational 'orchestration book'. In the meantime, the following text presents methods that have allowed such tuning practices to develop trust in the result by the composer and the performers towards enhanced musical experiences for all.

3. METHODOLOGY

3.1 Overall procedure and design criteria

3.1.1 An overview of the procedure

The overarching idea behind this procedure is to develop the performance patch[3] 'in the box' through the help of inter-application routing, using JackAudio [17]. This process is based on the making of a mock-up of the instrumental part, both audio and control (i.e. a MIDI pedal used by the performer to give cues) and is usually done in a DAW. Ideally, it is assembled from material recorded in workshops with the performers, with proximity microphones for processing, as well as room microphones as aide-memoire of the overall in-situ acoustic balance. These recordings are often edited within the composition process in the studio, and augmented with sampler-based instruments to provide an approximation of missing gestures; the whole process yields a very rough approximation of the instrumental part. Whilst this procedure might not be applicable to all types of music, it has also been used by the author in music relying on improvisation, using alternative takes to test different interactions.

This procedure of building the performance patch in the studio, with a clear calibration procedure, which will then be used every time the piece is performed in different halls, serves two agendas: 1) to rehearse the process of entering a new performance environment (*mise en salle*); 2) to refine the many parameters of the system. This preset calibration procedure consists of objective and subjective adjustments that will take a predictably short amount of time to yield consistent results in various venues and instruments. It will materialise in a clear set of instructions, presented as a bullet point list, as an integral part of the score, alongside the technical rider and the stage plot.

3.1.2 Design Criteria

These stated ambitions betray some design preferences: the composition process is considered as research, with the performance rituals and tools as its intrinsic parts. This has consequences on the system's design criteria: they must be clear and open, yet allowing subversion and challenges. One can remember the history of the vocoder, or of the 'clean' powerful valve guitar amplifiers, to see how artists can take what is considered as design flaws (i.e. the non-linearities) to new expressive levels.

From a musical perspective, the author favours an approach to chamber mixed music that stays clear from amplification-as-compensation, and instead prefers loudspeakers behind the performers in a setting that resembles a soloist in front of the orchestra. Indeed, all the musical relationships found in a concerto are of inspiration[4]. Both explicit causality [20] and mysterious interactions are exploited for their expressive values. Sonic fusion and extension rely on consistent sound quality, whilst hocketing or discursive opposition rely on powerful, clearly segregated antagonistic gestalts – the design and positioning of the loudspeaker setup is therefore an intrinsic part of the orchestration. This quest for innovative interactions and sonic possibilities brings new challenges to the hall.

From a human perspective, the author favours an equal collaborative relationship with the performers, as allies in

[2] With the exception of Hans Tutschku [15], the author has not come across such truly integrated approach of the composition and the pragmatic concerns of its performance in any of his 200+ encounters with such works. For a reference on how the opposite is the norm, see the excellent fieldwork by Sebastian Berweck [16].

[3] The word 'patch' is used throughout to represent a real-time music program. The proposed methodology is obviously not limited to any creative coding environment.

[4] A compelling analysis of such relation has been produced by Dominic Thibault for a piece by the author [18], by Maya Prynda for works by Stockhausen [19].

the musicking of a piece. This has strong implications on the loudspeaker placement, and on the confidence on the setup's consistency: if a powerful loudspeaker is at one's head, spiralling feedback is unacceptable. This must be true in all settings, on tour as well as in the practice room.

All these biases have strong implications on the design values for the performance system: it must be stable, reliable, low maintenance, with consistent setup time and results. The normalization of the calibration process enables this, which is quite common practice in high-flying popular music touring: if these artists can plan the load-in, setup, calibration, sound-check, performance and wrap-up of a stadium gig within a day, we have much to learn from their preproduction work ethics and practice.

3.2 Routing setup

The routing in the studio seeks to emulate the reality of the concert, and therefore should include all potential feedback problems, with some room colouration too:

1) The DAW send buss to the patch should include, as well as the direct closed-mic source, its room reverberation to a lesser degree. Depending on mic placement and type, this was found to be in the realm of 20dB below the nominal dry level.
2) The same send buss to the patch should also include a feedback from the patch's output, simulating in-hall loudspeaker feedback. These levels will again depend on the mic placement and type, but also its distance to the loudspeaker, and should be sent accordingly. In the case of multichannel speakers on stage behind the performer, a level between 18 and 24 dB lower than dry level seem to represent this reality accurately.
3) The patch return to the DAW should be sent to the same (ideally multichannel) room reverberation applied to the instrument, and will therefore also be sent back to itself. If loudspeakers are used in creative positions, in-hall impulse responses should be use to emulate this setup, or at least should be simulated, as argued in [13].

The author is well aware of the setup specific impact on the feedback level, and therefore recommends actual setup testing for such values very early in the composition process, as it is part of the orchestration constraints to compose within and challenge. The values above are taken from scenarios using proximity high-end cardioid microphones with loudspeaker levels matching the acoustic instrumental source (see section 3.3 below).

A word of caution on implementing such electronic feedback networks: beware that some DAW sends are instantiated at unity gain. If your patch is active as you send it back to itself, the consequences on your listening system (and your eardrums) could be unrecoverable.

Once the studio setup is emulating the concert setup with the routing described above, the calibration procedures can be designed and tested in the controlled environment of the studio, for consistent later uses in the hall.

3.3 Calibration procedure, part 1: setting up the i/o

The system is first calibrated to a normalised colour and level both at output and input. This is done in a series of four calibration patches that incrementally update files that will be loaded by the performance patch later. Patch #1 corrects the loudspeakers; #2 adjusts the overall output volume; #3 sets the input gains and corrects its colour; #4 tests the input to output gain structure. Each of these steps are explained further in the following paragraphs.

Firstly, a patch will send a burst of pink noise at -3 LU[5] to each loudspeaker, to troubleshoot routing issues and make level adjustments by ear. Then, the patch runs a series of impulse response measurements (IRs), ideally taken from a few points in the seating area, as documented and discussed in [8]. A few versions of this patch have been made to allow for different numbers of measurement microphones. Then, the IRs are truncated, averaged, smoothed, inversed and normalised, creating correction files in the form of finite impulse response filters (FIRs), with level and colour matching between loudspeakers.

The second calibration patch loads the FIRs and allows balancing the overall volume of the corrected loudspeaker system after the soundcard's output. It does so by sending a slow pink noise wave at -3LU in the loudspeaker directly behind the performer, who is requested to play the loudest part of the piece, so that the gain of the electronics in the PA system can be matched to the acoustic sound of the instrument. This patch also allows listening to the pink noise bursts in each loudspeaker to confirm that they are better matched. Since loudspeaker corrections can be quite drastic, the overall gain adjustment needs to be done in the analogue domain, post-computer, hence the importance of a quality powerful PA system.

The third patch sets the input gain for the instrument mic, and allows running an optional correction to be carried out as proposed in [22], this time compensating for the spectral imbalance of the instrument capture with hyper-proximity mic placement. It does so by comparing the closed-capture mic input with a reference mic positioned at the optimal recording position, designing again a FIR filter of the inverse of the problem. This stage is optional but improves significantly the parity of processing sound when fusion with the acoustic source is desired. The FIR is also band-limited to the useable range of the instrument to optimise further the input and reject feedback from the subwoofer.

The fourth calibration patch is where all of the above is finally put together in the ultimate unity test: with all the correction FIRs loaded, the performer is requested to play

[5] Calibrated according to the EBU R 128 standard of 0LU to -23LUFS. See [21] for the specifics on the standard.

a passage covering the full range of the instrument, that is recorded and then played back in the loudspeaker directly behind them. Adjustments to the instrument gain post-corrections are made directly in the patch, since the acoustic input is kept the same yet the mic correction FIR can produce significant gain changes. This level is saved in the file to be later loaded by the performance patch.

Note that in the suboptimal case of not being able to run either loudspeaker nor mic correction patches, unity FIRs are provided. In this unfortunate case, only the second patch (level calibration) and the fourth one (input-level calibration) are absolutely required. In effect, when running the patches from the DAW, these unity FIRs are the ones that are used, since there is no coloration induced by an in-hall setup (i.e. loudspeakers and mic in a room).

3.4 Calibration procedure, part 2: piece-specific musical settings

A fifth and final calibration patch is designed and must be run, since all essential piece-specific thresholds, stress points and levels are set and tested within, then saved.

Firstly, the control pedal input is set and tested. The many implementation of such device imply that different type of MIDI messages could be received, so the patch allows such assignation in the calibration routine without having to modify the code of the performance patch.

Then, an emergency reverberation send is set. Most chamber musicians are used to hearing themselves in a live acoustic space, and at times chamber mixed music concerts are in challengingly dry spaces. A reverb is then available in the loudspeakers behind the performers, and its level is set here to enhance the performer's comfort.

Many of the author's pieces use crude attack detection to enhance the interactivity of micro-gestures in the real-time processing, and/or to proceed to a subsequent section when it is relevant to remove the burden of pressing a pedal from the performer (i.e. in a delicate and suspended musical moment). Such attack detection can be tricky to set, so the thresholds are set through this stage: an absolute threshold to dismiss anything under mezzo-piano, then an adaptive threshold test with specific passages of the piece, with built-in error tolerance.

Another favourite process of the author is coloured and compressed hyper-amplification, with and without controlled feedback, for instance in [23] and [24]. The effect chain is a bandpass filter, going in an expander, going in a form of distortion, going into a compressor. This process is one of the most explicit example of what was made possible by the trust provided by such a calibration procedure: telling a performer that a loudspeaker capable of producing 128dB SPL will be creating feedback one meter from their head, takes a lot of confidence from both parties in the reliability of the system. Moreover, to compose such feedback in order for it to integrate the musical material at different, nuanced, reproducible levels is also quite demanding on the consistency of the system. To do so with limited setup time for the contemporary touring artist is a trial by fire. This feat is achieved through a simple procedure, starting from overly high thresholds for the expander, and overly low thresholds for the compressor. Depending on the piece, the number of settings to test changes, but usually consists of: 1) with the expander bypassed and high gain in the distortion, the user slowly raises the compression threshold until a desired feedback level is achieved; 1a) a higher threshold could also be set here to get a louder feedback level in other parts of the piece; 2) with a much lower gain in the distortion, still with the expander bypassed, the user adjusts the compression level just below feedback. This is usually still very amplified for soft sounds (i.e. key clicks, bow hair, breathing) but let the louder acoustic sources mask their compressed version; 3) then the user would set the expander threshold as a safety, just below the lowest sound to be amplified. With these thresholds set, the palette 's extremes are defined, and post-processing volumes can be adjusted downwards from that reference within the cue list of the composition as required by the musical context.

A third strand of real-time processing that the author favours is descriptor-based granulation of real-time audio capture, after exploring such processes with CataRT [25], and later streamlined with the help of real-time audio descriptor objects [26]. This type of granular processing allows a composer to go beyond the binary process switching, towards improved orchestration nuances: one can articulate the effect in terms of grain features as well as the usual playback parameters, i.e. producing clouds of quiet noisy bursts, or seamlessly expanding in time only the pitched material, or even using the analysis stream to drive real-time audio mosaicking of another corpus – an altogether new level of finesse of musical expressions is available from that approach. However, such processing being feature-dependant requires a consistent input between versions and instruments to produce the desired results every time. The disciplined calibration procedure has allowed the audio input to the patch to fulfil these conditions. Other methods such as machine learning could have been used to train the system for each setting, but such training is usually much more time consuming than the proposed methodology, and therefore would be done at the expense of musical rehearsing time.

3.5 Performance Patch Features

3.5.1 Best Practice

Obviously, the main performance patch loads all the settings saved in the calibration procedure. It also has all the essential features of a professional performing patch, in effect a compilation by the author of all the best practice observed over the years in others patches, and emerging from needs and errors. These are listed here:

1) the instructions needed to run the piece;
2) a version number for the patch;

3) a single, centralised audio input and output, and control input from the cue pedal;
4) a visual feedback for the cue pedal, the input and output levels, and the attack detector activity;
5) a set of small, quiet test burst buttons to test the system output routing discretely;
6) an option for stereo fold-back of multichannel audio output, for rehearsing over headphones;
7) a large panic button in case of emergencies;
8) a master volume that can be incremented by keyboard action;
9) a few keyboard shortcuts, to help recovering the cueing of the piece should things go wrong:
    a. to give a normal cue (following the constraints built in the performing patch, i.e. not allowing a pedal before a certain moment);
    b. to force-forward to the next cue (overriding the piece constraints);
    c. to mute the cueing pedal input (should a double trigger occur – this can happen when the performer remembers a forgotten pedal/cue, yet the composer manually already gave the said cue to compensate) – note that the patch is flashing red whilst the mute is enabled, to make sure it is not forgotten on;
    d. to come back one cue should a double trigger happen;
10) a method to start the piece at specified rehearsal points in the piece.

In addition to all of the above, the following few personal add-ons have been confirmed as useful.

3.5.2 Access to critical calibrated values

The temptation of such a thorough and reliable calibration procedure has been to leave all the settings hidden in the depth of the patch, to avoid the dreaded interface cluttering for the handler of the electronic part, whomever that might be. They were indeed hidden at first, but the surprising change in dryness of a public-filled hall motivated the emergency reverb level to be brought to the fore of the performing patch; on another occasion, a performer's enthusiasm made it relevant to bring up the attack thresholds to allow in-performance tweaking – not that the pieces would collapsed without these, but this access allowed to further improve performance conditions.

3.5.3 Stemming

Once the system becomes consistent, one is able to consider more refined adjustments. For instance, the music of the author is often multi-layered in dense counterpoints of multiple juxtaposed processes and sound-files. Relative levels between these layers could always be tweaked. The patch therefore has a hidden layer of stem mixing, where each real-time process, and audio file playback, gets assigned a special remote control grouped in musical roles. Should the composer not be present for the performance, the piece will run these at unity gain. Should someone with in-depth knowledge of the music wants to adjust relative levels, i.e. cutting the over-amplification by 1dB, or boosting an impactful fixed-media transitions by 2dB, one can do so easily. These preferences are saved independently of the main calibration patch and are completely optional, but give the composer something to play with in the frequent eventuality of the patch running by itself with reproducible results. This stemming allows a level of finesse beyond what is usually possible in live performance, starting to reach the quality level that is now the norm in stadium gigs with thorough PA calibration.

4. CONCLUSIONS AND FURTHER WORK

4.1 Advantages

The advantages have been highlighted throughout the paper: a rigorous calibration methodology gives trust in the result for all parties involved, allowing controlled risks, enhancing portability and adaptability of live performance almost at par with studio settings.

The disciplined tuning of fixed duration and reproducible results allows sound-check and rehearsal time to focus on musical questions rather than be spent on troubleshooting. The acquired performer's confidence allows for team work in the reliability short calibration process, then to take all the remaining time to do what chamber musicians usually do in pre-concert situation: getting used to the room acoustic, to the stage, to the lighting, etc.

4.2 Drawbacks

The main obvious drawback is the coding discipline this procedure requires. It means that once the piece is 'finished' as a score and its mock-up, there is still much to do. However, from the author's experience, some sort of time-consuming troubleshooting will inevitably have to happen at one point, so it is better to be done in the privacy of the composing studio than in the hall, with everyone waiting for the system to work – or the performance is not simply cancelled due to time constraints [16].

It also means that starting to 'compose' in the studio is not the beginning either. There should ideally be an iterative testing of ideas and patches in a room, with the actual setup, to see the constraints of the system, and to push them further with more ideas. This could be once more compared with orchestration, where there is nothing better than to be able to test sonic and notational hypotheses with the ensemble: only experience can tell the difference between a good idea, and an actually good sounding one.

4.3 Future Work

This approach is still in progress, and hopefully more people will develop the discipline of integrating thorough calibration as part of their orchestration practice. As for the author, the next step is to implement room gain compensation [27] and partial correction, as advocated in the emerging room correction literature.

Another element to change is the normalised pink noise reference in the calibration patch 2, for a studio recording of the actual instrument; indeed, the broadband nature of the pink noise gives a slightly skewed perception of its loudness. A recording of the band-limited source would give a more accurate point of comparison of loudness.

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