CHAPTER 6

Recording

Let us now transcend the action of producing and perceiving a tone to how we document and mediate it through technology. For the twenty-first century lutenist, technology is ever present. When we play at a concert, someone places a microphone before us; we record music that we try to get published by a label; we make home recordings that we share through online networks such as Sound Cloud, YouTube or Facebook (the list could go on). To this day, much has been written on the recording process, but there are still considerable holes to fill within academia. Handbooks treating the recording process and mixing[1] often provide thorough understanding of technological processes, but they usually neglect the artistic effects of technological decisions. Also, since the end of the twentieth century, recorded music has been a preferred text to analyse in popular music studies,[2] but Early Music is noticeably absent in these contexts. More recently, the recording process itself has been accepted as an academic discipline,[3] and some also take into account how more detailed levels of technology shape creativity and aesthetics.[4] However, the classical genres are still underrepresented in academic literature when it comes to recording technology,

1 Such as Gibson, D., *The Art of Mixing: A Visual Guide to Recording, Engineering and Production* (USA: Artist Pro Publishing, 2005); and Miles Huber, D., and Runstein, R.E., *Modern Recording Techniques* (USA: Focal Press, 2010).

2 To name only a few: Hawkins, *Pop Score*; Lacasse, S., 'Intertextuality as a Tool for the Analysis of Popular Music: Gérard Genette and the Recorded Palimpsest,' *Practicing Popular Music: 12th Biennial IASPM International Conference Montreal 2003 Proceedings*, edited by Alex Gyde and Geoff Stahl (Montreal: IASPM, 2003): 494–503; Moore, *Song Means*; and Zagorski-Thomas, S., 'The Stadium in your Bedroom: Functional Staging, Authenticity and the Audience-Led Aesthetic in Record Production,' Popular Music 29, (02; 2010): 251–266.

3 See for instance Frith, S., and Zagorski-Thomas, S. (Eds.), *The Art of Record Production: An Introductory Reader for a New Academic Field* (UK: Ashgate, 2012).

4 Such as Collins, K. (Ed.), *From Pac-Man to Pop Music: Interactive Audio in Games and New Media* (UK: Ashgate, 2008).

and even more so the lute. By discussing the dialectical relationship between the lute and recording technology in the twenty-first century at a deeper level, I will address the transformative processes from which recorded lute sound evolves. (I speak now of the more general 'lute sound' rather than tone production, because we are addressing both its production and representation.) My motivation stems from a hypothesis that a performer can no longer consider their authenticity (whether authenticating his or her own persona, the work, the genre or the audience) as detached from, or independent of, the production process. This is especially true at a time when musicians have been given the possibility of performing, even existing, in multiple countries at the same time through various streaming and downloading agencies. In fact, the self-expressing tone production, including its historical, contemporary and physical building blocks, now enters a new level of significance. This is because we no longer act in the same room or time where that very act is received and also possibly perceived. Our possibilities of positioning ourselves socially, as discussed in Chapter 5, are suddenly theoretical. We don't know who the perceiver is, where they are or even when they are active. We cannot judge them by their appearance to adjust our impression management, nor can we know beforehand if our recording appeals to critics or 'fans.' We are naked, so to speak, and can only present ourselves and our self-expressiveness in a one-way communication without getting instant feedback from our audience.

The lute makes a particularly interesting case here because it has such a feeble, crisp and weak tone, making it quite troublesome to record well. The dynamic range is restricted in such a way that the clear tone and the noise produced upon playing (such as breathing, the changing of hand positions, and noise from the chair while moving around) are difficult to separate; sometimes in quieter passages the noise can overpower the clear tone and attract more attention. The strong, quick attack that comes directly upon plucking a string and the much weaker and quickly dying tone that follows present other problematic issues; for instance, when setting proper recording levels that are strong enough to produce a good sound without having the signal exceeding its maximum level. Moist or dry environments can affect the thin woodwork of the instrument in a manner that alters the tuning and tone quality of the instrument to a greater degree than other instruments associated with the Early Music genre (see Chapter 4). In a post on the *Unquiet Thoughts from Mignarda* blog on 27 October 2010 we find a description of the problems surrounding a lute recording that is quite revealing:

> […] Recording the lute […] can be a musician's worst nightmare, and lutenists can be the bane of an engineer's existence. Since the lute is so quiet, a recording engineer's tendency is to place a microphone close to the instrument so as to cancel out extraneous noise that can filter in, even in the controlled environment of a recording studio. There is typically a protracted negotiation between the lutenist and the engineer that involves a great deal of experimentation with microphone placement and, likewise, a great deal of whingeing on the part of the musician. The engineer wants the mic closer, the musician doesn't like the intimidating, nervous-making thing so close, nor the presence of string noise and breathing in the recorded result. Money is spent and no one is happy. The best solution is to record in a very live, resonant space that is relatively quiet and allows both musician and engineer to relax and capture a pleasing natural sound with the mic at a comfortable distance.
>
> Recording in old churches with their conducive atmosphere, high ceilings, hard surfaces and spacious resonance – the preferred venue for lute recordings – can be nearly impossible because of noise from building mechanical systems, traffic and routine neighborhood activity. […] If ventilation systems aren't running, the space is probably either cold and damp or hot and stuffy, affecting the sound of the instrument, tuning stability of the strings and concentration of the lutenist. […] Then there is the sometimes bizarre, unfocused sound resulting from the lutenist's refusal to allow the microphone to be placed so close that finger noise or breathing might possibly be detected. What is heard is more of the room echo and less of the real instrument and the musician's interpretation. This is not happenstance, it is a choice on the part of the lutenist and producer. […] But the manufactured perfection listeners have come to expect in recordings of lute music is not the same as what one actually encounters attending a live concert, with human beings reacting to music being performed by other human beings.[5]

5 This post can be read in full at: Unquiet Thoughts, 'Is Lute Best Heard Live or on Recordings,' *Unquiet Thoughts from Mignarda*, 27 October 2010. Retrieved 16 July 2014. URL: http://mignarda.wordpress.com/2010/10/27/concert-versus-recordings/.

Recording and engineering mentalities

On the technological journey from sound waves to electric currents to binary digital information, back to electric currents and back again to sound waves,[6] it is clear that sound, after being produced, lives a complex life before reaching our ears. During the long evolution and debate of the authentic, and later historically-informed, performance in Baroque music, we have seen numerous recordings being produced all over the world. The discussions are often focused towards the musician and their instrument, but the technological production and its aesthetical compound are often neglected within the field of 'Classical music.'[7] In recent years, we have seen how music reviewers also include comments or even grades on sound quality, yet, what is often neglected is how the sonic design of the recording relates to historical data in terms of 'sound' and not only performance. As Lelio Camilleri points out, although much more obvious in popular music productions, 'the studio has become a compositional tool in which musical ideas are formed into sounding matter.'[8]

One considerable difference in the recording of 'Popular' music *versus* 'Classical' is the sonic and spatial mentality behind the production. In popular music productions, close microphone placement, acoustic separation and use of multiple microphones have become a natural part of both the sonic and spatial design. Frequency modulation, panning, amplitude and effects are used in order to create a superficial sonic sphere appropriate to the product itself, rather than thinking of its live re-enactment before or after the recording session (of course, there are exceptions to this statement). On the other hand, Classical music always seems to seek 'natural' sounds and it is rather unusual to hear recordings truly elaborate with sonic matter. Simon Zagorski-Thomas (2010) suggests that 'the fact that these musical forms pre-date recording mean that there is greater resistance through the recording process,' but there is perhaps

6 Or more materialistically: From sound to microphone, through cables, into the recording machine, through an AD/DA converter, into the DAW (Digital audio workstation), transformed to a 'master' of some sort (physical or virtual), and finally into the industrial press machine.

7 Generally, I am careful in using terminology such as 'classical', 'rock', etc. due to their wide adoption and the spectrums of assumptions accompanying them. I have nonetheless decided to use such terminology in this essay for the sake of clarity.

8 Camilleri, L., 'Shaping Sounds, Shaping Spaces,' *Popular Music 29, 2* (2010): 199–211, 199.

more to it than that.[9] Thanks to Herbert von Karajan's will to explore and embrace the new recording technology from the end of the 1930s, there is no doubt that Classical music has joined the technological sphere. But when focusing on spatial and frequency exploration and modulation, the 'resistance' mentioned by Zagorski-Thomas proves more evident. Developments in the 'Classical' genre seem, up until today, to have focused more on high fidelity and the perfection of sound, rather than exploring *new* sounds and spatial placement (such as guitars all panned to the right and the accordion all to the left). When discussing high fidelity (hi-fi), Zagorski-Thomas brings into focus some requirements for good sound quality. The frequency range should be broad enough to retain all aspects of a sound, making the reproduced sound identical to its source (free of distortion and noise, and with loudness and dynamic range[10] being comparable to the original source).[11] Further on, he mentions two additional stipulations, maintaining spatial naturalness and life-like reverberation, which are often neglected as 'attempts to reproduce the full dynamic range of a concert hall in a small listening room would not create a very pleasing effect.'[12] Recent recordings have, however, proven to be more interested in elaborating on these points. We can often see a division between the sizes of the ensembles recorded, where orchestras are often sonically presented in a concert hall with the reverberation that follows, and ensembles are more widely panned and are perceived to be placed more closely to the listener. This is, of course, a natural phenomenon due to the physical size of an orchestra *versus* the chamber ensemble. An ensemble is more

9 Zagorski-Thomas, *Stadium*, 263.

10 The *dynamic range*, in the context of this paper, is the range between the lowest and highest sounding volume (i.e. amplitude) of an instrument or recording equipment. For example, a piano has a wider dynamic range than a flute using standard playing techniques.

11 Paradoxically, after achieving best possible sound, the recording quality is reduced by half or sometimes even a fourth of its resolution to fit on a CD. Lislevand, R., *La belle homicide: manuscrit barbe* [CD], France: Naïve, 2003, was, as an example, recorded on a Nagra digital field recorder (24 bit/88.2 kHz resolution) which suggests that, in order to fit on a CD (with an industrially-standardized resolution of 16 bit/44.1 kHz), the resolution of the original file had to have been cut in half before it could be printed on a CD (Lislevand, *Homicide*, booklet: 27). Another oxymoron is the application of *dithering* in the mastering process, where one adds low levels of noise to the digital sound file in order to 'hide' digital miscoding and thus reduce the perceivable noise upon listening.

12 Zagorski-Thomas, *Stadium*, 261–262; Although those willing to embrace the digital plug-in world are given many options in restoration and creation using reverb effects.

likely able to appear in a smaller room, putting the listener closer to the instruments than a massive orchestra, and thus making the spatial distribution between the instruments more obvious. Turning towards film music we soon realize that the case is quite different. Whereas the 'Classical concert' recording tries to restore natural spatialisation, the modern film scores are recorded more 'hot,'[13] 'clean' (i.e. low levels of noise) and sonically detailed. Instruments are more three-dimensionally placed within the sonic frame (which does not need to reproduce reality), the perceived 'sonic headroom'[14] appears larger, and featured instruments are emphasized when needed. In this case, the sonic treatment in film music becomes interesting if we turn to Early Music ensemble recordings (as well as several contemporary art chamber music recordings). For example, in the recordings *Forqueray: Pieces de viole avec basse continuë* (1995)[15] and especially *Santiago de Murcia Codex* (2010),[16] we see how headroom, spatial use, and hot level resembles more closely the mentality of film scores than recordings of later period classical projects. Solo recordings of lutes present a different case again, as can be heard in many recordings where the lute is placed at a certain distance, preferably in a church with quite a lot of reverberation. One of the exceptions is found in Anthony Bailes' recording *Lute Music of the Netherlands* (2012),[17] presenting a much more detailed, 'roomy' quality in opposition to his earlier recordings, *Gaultier:*

13 Within all analogue recording equipment, sound is processed as electrical currents. Recording 'hot' signals is a popular metaphor of maintaining a high level of electrical currents within the equipment through the recording process, making the physical wire within the electronic circuits reach a higher temperature (hence the use of the word 'hot'). This terminology has come into use also when using digital equipment as a signifier of the same recording mentality (note, there are wires in digital equipment as well). Some positive outcomes of this mentality result in increased dynamic range and better signal-to-noise ratio (i.e. the distance in volume between the inherent noise of music recording equipment and the recorded sound. Put simply, the greater the distance between sound and noise, the less the noise is heard during playback).

14 The term 'headroom' can be interpreted in several ways. In this case I refer to 'headroom' as a metaphor of the perceived sonic space upon listening. This means, for example, that by modifying the frequency range, as well as reverberation, one can create an illusion of situating the recorded instruments in a more spacious room (especially on the perceived vertical axis).

15 Pandolfo, P., Balestracci, G., Lislevand, R., Egüez, E., and Morini, G., *Forqueray: Pieces de viole avec basse continuë* [CD], Spain: Glossa, 1995.

16 Ensemble Kapsberger and Lislevand, R., *Santiago de Murcia Codex* [CD], France: Naïve, 2000.

17 Bailes, A., *Lute Music of the Netherlands* [CD], Germany: Carpe Diem, 2012.

Apollon orateur (2009)[18] and *Une douceur violente* (2011),[19] where the strong church-like reverberation is clearly present.

Cosmetics and editing

One does not have to investigate much before realizing that there is a gap between what is produced on the recording *contra* 'live' on stage. Perhaps one important factor to consider in this context would be 'sonic memory.' As memory (in this case long-term memory) is triggered by repetition, it soon becomes evident that repeated listening to a recording makes the memory of it more consistent,[20] compared to a concert performance only heard once. Thus, it is understandable that a recording artist would wish to make that sonic sensory *autograph* flattering by editing the recording.[21] Also, when we cannot interact directly with the audience, we are also more interested in creating a good impression regardless of context and situation. Humans are, after all, human, and even the most accomplished musician sometimes wishes to be able to go back to a concert and do something a bit differently. In a concert this is, of course, not possible[22] but recording technology enables us to make those changes. Still, at a concert, small 'human alterations' or even mistakes are, to some extent, accepted but never so on a recording. As a microphone perceives more in a 'live' situation than our ears can, the musician becomes more self-conscious than perhaps they would have been in a concert. A small, unconscious body movement inaudible on a concert stage could certainly be audible on the recording, thus making the musician focus even more on controlling their movements; especially considering the possibility of turning up the volume, making the details and ambient noises even more

18 Bailes, A., *Gaultier: Apollon orateur* [CD], Belgium: Ramee, 2009.

19 Bailes, A., *Une douceur violente* [CD], Belgium: Ramee, 2011.

20 Especially since cerebral regions activated by listening also appear to be active while remembering music; see BBC 'Musical Minds: Imagining and Listening to Music (Excerpt),' [YouTube video] 2009. Retrieved 31 July 2012, URL: http://www.youtube.com/watch?v=_FkdDX--IaU.

21 Camilleri, *Shaping*, 200.

22 Although I did indeed participate in a concert once in Oslo where the piano soloist asked the audience if she could do her performance of a certain piece again as she did not believe she played it well enough the first time.

evident. Those otherwise inaudible sounds suddenly interact with, blend with or even compete with, our recorded tone production.

The Classical recording in a consumer context

Although recordings of Early Music perhaps wish to capture a 'natural' performance, placing the listener in the audience, they get edited and polished beyond naturalness. In addition to the mixing traditions previously mentioned and the performer's aesthetic agenda, this may perhaps have something to do with market criteria. Mixing engineer Dave Pensado comments (although in a different context than Classical music), 'Back when radio stations ruled the world, if you did a mix you only had to compete against other songs in the genre you were working in […], but now, in 2012, you have to compete against everything.'[23] Modern audiences, thanks to the Internet, are often not only attracted to one or two genres alone. The same person could have hip-hop, rock and Classical music on the very same playlist, which inevitably places Classical music next to other genres with completely different sonic approaches. Pensado further makes a comment (a mix of humour and reality, as is often the case in his videos) that he mixes rock as if it was hip-hop. I suggest this also applies to Classical music to some extent, as modern technological possibilities and trends form our expectations of good sound (e.g. emphasized bass register and noise-free sound). This is not to say that a Classical piece would be mixed in the same way as a song by Rihanna. Rather, when a listener places a Classical piece on their playlist next to a rock song, they do not expect to have to, for instance, increase the volume every time a piece by Bach comes up, or have their 'ears explode' every time the next piece starts and the volume has been turned up too loud. This brings us to another crucial aspect of today's recording, mixing and mastering reality — the compressor. A *compressor* reduces the overall dynamic range of a recording (making quiet sounds louder and

23 Pensados Place 'Into the Lair #42 - Working with Bass and Kick Drums.' YouTube video, 10'53", posted by 'Pensado's Place,' retrieved 10 August 2012, URL: http://www.youtube.com/watch?v=1OfSS3Py-Tk; I have omitted superfluous words like 'uhm', and repetitions of words while thinking of what to say in this quote.

loud sounds quieter), to allow the music to be played at higher volumes.[24] Compression is an integral part of most recordings today, independent of genre. It is used during production, post-production and in mastering. As it directly alters the natural dynamics of recorded music (making the dynamic range of the original performance narrower),[25] compressors become essential to consider. (Especially in terms of 'authenticity' and music mediation). What's more, given the compressor's function as a sort of automatic volume controller, by increasing the volume of quiet sounds it makes the inherent production noise more apparent.[26] The compressor does not only affect the dynamics of the performance alone, but also the dynamic relation between the musician and their sonic surroundings. By extension, dynamic compressors are not only employed during the production process alone, but also, for several reasons (such as making music audible in noisy environments), during all types of broadcasting. Television channels, radio stations and online distribution all add compressors to audio signals. (Even the satellites directing TV and radio signals affect sound quality through their encoding into MPEG-2, or MPEG-4 formats incorporating AAC data processing.[27] The music TV channel *Mezzo* is one of many channels streaming through such satellites.[28]) Obviously, a whole range of additional problems arises during music-streaming, but I will not treat these matters in detail here. One may rightfully argue that the CD is an obsolete and outdated recording medium, which is increasingly set aside by more modern technologies, such as streaming. In fact, initially, my idea was to include the more recent developments in stream-

24 This process differs from another important method called *normalisation* that raises the whole sound to a chosen level related to the sound file's highest sound (both loud and quiet sounds get louder). Normalisation can only be applied following a completed sound file (while compressors can be used in real time) and does not alter the dynamic range internally as the compressor does (as it makes everything louder) and will thus not be treated in detail in this paper.

25 In some dance genres the dynamic range gets compressed to a volume difference (between the loudest and the quietest sound) of between 2 or even 1 dB.

26 Thus, we understand better why it is important to maintain a proper signal-to-noise ratio (see Note 8 for explanation) during the recording process in order to minimize perceived noise at later stages of the production.

27 The AAC format follows the same principles as the famous MP3 format, using algorithms to extract all 'unnecessary' information (at least according to the algorithms) from the original sound file, making the new version take up less memory space and processing power.

28 Lyngsat, 'Mezzo,' *lyngsat.com*. Retrieved 6 September 2017, URL: http://www.lyngsat.com/tvchannels/fr/Mezzo.html.

ing on computers, smartphones and tablets, and the lute's appearances in gaming, film and 'second-life' virtual reality games as well, but the topic soon overwhelmed me, given the context of the present project. Not only because there are so many variations and possibilities to consider, but also because their technological use and repercussions vary from instance to instance. Furthermore, they present many hidden processes to which I have no access (both for practical and juridical reasons) which would make the consistency of the line of reasoning that I wish to present here difficult. What should be noted is that the technology used in modern streaming (and other types of uses) is based on the same principles as that of the CD. As such, CD technology has not only an archival, historical function, but also works to provide a pedagogical tool for understanding later technologies. That is, the underlying principles remain the same, upon which one must consider each separate distribution medium through their technological framework. (The latter is even further complicated by the fact that some people listen through smartphones, for instance, where both the streaming service, the phone itself and the earbuds all transform the sound in their own specific way. Clearly, it would be almost impossible to offer the reader clear options to optimize the fidelity of lute tone when played back on a tiny smartphone speaker, for instance, without knowing the specifications of all components involved. The CD, then, provides a common ground of standardised and disclosed processes which can later be transferred to other media services through dedicated reading on the relevant issues for a specific situation. Furthermore, as the main ambition in this chapter is not foremost a practical one — when we upload a sound file to YouTube, Distrokid or Spotify, for instance, we give up our hands-on influence on the result to their respective predetermined algorithms — but a theoretical one, to understand the biology of lute sound from a meta-perspective (which I return to in the final Conclusion), the now old-school technology of the CD recording will suffice. On this background, I will for the remainder of this chapter focus on the CD for the sake of clarity and efficiency, leaving discussions related to more recent technological developments to future projects which can treat the subject from more approachable perspectives.

Recording technology and authenticity

There is a tension, then, between technology, performance and scholarly contributions that we must not fail to consider. Alan Moore states in his book *Song Means: Analysing and Interpreting Recorded Popular Song* (2012) that there is no single notion of authenticity. By directing us to five key moments in history towards developing 'authenticity' as a concept, Moore reveals several aspects in which authenticity becomes an issue: 1) the collecting of folk tunes and putting them to new use in creating nationalistic music; 2) the friction between autonomy and function, between the musician's self-realisation and the audience's expectations; 3) friction between music responding to market needs and music attempting to *annex* one in the emergence of rock 'n' roll; 4) the tension between an artist's accepted persona and their received transgressive persona; and 5) the opposition between mind and body. Moore's discussions, of course, direct themselves to recordings of music from a period of time other than the Early Modern period, incorporating a rather different cultural context (not to mention a different source and empirical reality). The second case of friction above (between autonomy and function and between the musician's self-realisation and the audience's expectations) stands out, in our case, as most obviously related to our situation. He writes:

> On the one hand, an expression is valued because its production appears to rest on the integrity of the performer, an integrity that is read as secure, as in some sense comfortable. On the other hand, an expression is denigrated because that integrity appears, from the viewpoint of the critic, to have been compromised […] the commonest attribution to the term 'authentic' in relation to music refers to the maintenance of the origins of a performance practice.[29]

On hearing an Early Modern CD, we perceive the musician's presentation of historical music. We listen to their attempt to interpret the written material, channelled through their personal subjectivity and integrity. The critic, then, does not actually criticize the 'authenticity' of that performance solely based on the written material it interprets (be it literature or a musical score), but rather based on the performance of

29 Moore, *Song Means*, 262–263.

that material as interpreted by the musician. Also, the lute tablature used to denote the lute repertoire I am concerned with here, by its very nature, is even more open to interpretation than regular staff notation and leaves much of its realisation to the integrity of the performer. The question of 'authenticity' is thus strongly connected to the musician's own historical understanding of baroque music tradition and the performance as presented on the CD. Moore further directs our attention towards two schools of addressing 'authenticity.' On one hand, we find 'authenticity' as 'purity to practice,' and on the other, 'authenticity' as 'honesty to experience.' In extension of the latter, Stephen Felds argues (cited by Moore) that 'authenticity only emerges when it is counter to forces that are trying to screw it up, transform it, dominate it, mess with it.'[30] The two practices are perhaps more difficult to separate when speaking of historical music than in speaking of modern genres such as pop, rock and jazz. To provide an example from my personal experience, one of the most frequent debates I encounter when talking to fellow lute players is that of whether one should or *can* play lute music on the Classical guitar or not, as it is not 'authentic.'[31] The critic then assumes a judging role, claiming to possess the 'truth' of how music was appreciated and received in the seventeenth century and how it should be performed today. So, with Felds' comment in mind, 'authenticity' becomes a matter of right and wrong in order to protect one's own position, and this is, at least so I believe, a dangerous path to follow. It is crucial to be aware of the fact that those people fighting for this culture (in my case Early Modern lute music) live today, or at least in recent history. Our modern notion of 'authenticity,' then, is based on modern research projects — 'authenticity' becomes 'maintenance of the modern scholar's practice.' And this is why it is hard to separate the subjectivity of 'purity to practice' and 'honesty to experience,' at least in terms of scholarly works, as they solely build on a modern understanding. In light of this, we find Moore's perhaps most important argument:

30 Keil, C., and Feld, S., *Music Grooves: Essays and Dialogues* (Chicago: Chicago University Press, 2994), 296, cited in Moore, *Song Means*, 262.

31 Of course, I frequently meet with opposing opinions as well. I recently had the good fortune to perform Antonio Vivaldi's concert for two flutes in C major together with a famous flute player who, right before we entered the stage, amusingly said to me: 'Vivaldi is dead. We can do whatever we want.'

> [M]eaning is not embedded in the music listened to, but is discovered in the act of listening, and I can see no reason why attributions of authenticity that are, after all, an aspect of meaning, should fall into a different class. This means that any analysis that claims that a particular song, or a particular performance, is authentic must be regarded with suspicion. […] 'authenticity' is a matter of interpretation that is made and fought for from within a particular cultural and, thus, historicized position. Like all meanings, it is ascribed, not inscribed.[32]

'*Ascribed* authenticity,' then, questions the integrity of the subject with whom we relate, making the musician the actual focal point. It may be, as Moore puts it, more 'beneficial to ask who, rather than what, is being authenticated by that performance.'[33] If we let 'who' signify the 'recorded performer,' we can see how the technological aspects (such as recording equipment, aesthetical choices and market expectations of technological performance) provoke questions of how the modern Early Music performer is authenticated through the CD. Recall the many aspects altering the sound, not only by perceived frequency content, but also dynamically and spatially. The music we hear on the recording is something other than what we would hear sitting in the same room next to the musician (not only from an ecological point of view but also from a pure, cognitive-perceptual viewpoint). The 'live' musician is transformed into a medium that evidently did not exist in the Early Modern era and so becomes a construct of Other — a representation of a constructed musician. As a result, we may ask ourselves if the 'authentic Early Modern music CD' is in fact plausible or even possible?[34] The Early Modern music CD balances between, or becomes the *nexus* of, the different aspects of a CD production's construct. Thus, on one hand, we have the production team (performer, recording team, producers, manufacturers, etc.) and on the other, the scholarly dialogue with the past (empirical data, scholarly work and theorisation). So, within the sonic autograph of an Early Modern lute music record, we meet the need to carefully balance the performer's artistic intentions and the musicological foundation

32 Moore, *Song Means*, 265–266.

33 Moore, *Song Means*, 260–271.

34 We see how Zogorski-Thomas' hi-fi criterion, referred to earlier, of reproduced sound being identical to its source, suddenly must be regarded from other perspectives.

behind the music (whether empirical or subjective) with the listener's own expectations. Following previous discussions, it is not hard to argue that the truly interesting aspect in our case is perhaps not the aesthetical choices, authentication or sonic design *per se*, but the unforeseen and perhaps unexpected results of interaction between music, research and technology. Furthermore, what we see is how the perspectives presented in Chapter 5 are also valid here. The matter of positioning ourselves through tone production as self-expressive acts is now transferred to the technological realm. We can now speak of how the recorded music positions us within a social construct, how it is judged (e.g. attitudes regarding authenticity) and how we preserve our self and identity. I find this multifaceted perspective on recording fascinating, as it works against viewing the singular recording as merely an artistic product, but rather reframes the singular recording as part of a self-representation and self-formation. By constructing and designing the sound on a CD, for instance, we deliberately work with the re-representation of our tone production. It is then easily argued that recorded lute sound must be seen as an entity other than the original performance and performer, and that within an analysis it must be judged on its own merit (this has, in fact, already become the practice of most musicology, taking the recording as case).

The question of authenticating the performer, then, must be addressed at the intersection — the dialogue — between performer and recording, in the relation between the sum of technological production and, to borrow Philip Auslander's terminology, musical persona.[35] This is, however, somewhat troublesome. If recorded lute sound has become something other than lute sound itself, as a result of the processes behind its appearance, then how can one authenticate the other? Perhaps authentication of the artist is rather to be sought, where they approve of the final recording; it authenticates their vision of how they wanted it to sound — the vision is authenticated through the recording process. There are, of course, many other instances where authenticity can be ascribed and debated: How is a recording authenticated by its audience? How is it authenticated, to use

35 Auslander, P., 'Musical Personae,' *The Drama Review 50, 1* (2006): 100–119, http://www.posgrado.unam.mx/musica/lecturas/interpretacion/complementarias/perspectivaFenomenologica/Auslander_Musical%20Personae.pdf.

Serge Lacasse's terminology, at an *archiphonographic*[36] level?[37] How is it authenticated by the record label (for instance: 'this is how our productions should sound,' 'this is our sound')? Situations like these cannot be treated without incorporating cultural and social aspects in order to deal with them; as I am not concerned here with the cultural implications of recorded lute sound but rather its transformation and how it evolves, I will not go any further into these topics. It follows, then, that we cannot simply speak of the authenticity of a lute recording without bringing it into a cultural and social relation, presenting a set of parameters around which the discussion will evolve. Authenticity, then, can be seen, at least from the line of argument that I have pursued here, rather as a tool for cultural discourse than for authentication itself (audio forensics, of course, uses the term 'authenticity' differently, but their process is somewhat different from what I am trying to depict here).

Technological considerations — approaching a biological perspective

Jack Martin and Tom Jessell state in *Essentials of Neural Science and Behavior* (1995), that '[c]olors, sounds, smells, and tastes are mental constructions created in the brain by sensory processing. They do not exist, as such, outside the brain.'[38] Sound is only, in reality, physical movements of particles (see Chapter 4), and it is not until it passes the ascending neural auditory pathway (from the outer ear, through its cochlear transcoding, all the way through the brainstem to the higher processing of the cerebral cortex) that it becomes music. Consequently, it becomes evident that technology *per se* physically blends with the original instrument before reaching our perception as one unit. By following the sound of lute chronologically throughout the recording process, using

36 'In the first area, we find a single item, archiphonography, which is concerned with relationships occurring at the highest, most abstract level. Paraphrasing Genette, it consists in the entire set of general categories—types of discourse, performing styles, musical genres—from which emerges each singular phonogram;' Lacasse, *Intertextuality*, 496.

37 Lacasse, *Intertextuality*, 496–497.

38 Kendall, E., Schwartz, J., and Jessell, T., *Essentials of Neural Science and Behavior*, International edition (USA: McGraw-Hill, 1995), 370.

a computer to store the sound until it reaches the state of a physical CD following the Red Book standard, we can see how tone production and the recording process live a more complex life than a simple documentation process. In fact, we see how the original lute sound is deconstructed and remodelled through a process where every action and decision take part in shaping our tone production. Let us now review the process step by step.

Microphones

The first stage of lute sound transformation is through the microphone, a so-called transducer, where periodic pressure waves are converted into electric currents. When sound reaches the microphone, it makes the membrane inside of it move according to the pressure waves it perceives. Through electrostatic (condenser-type microphones) or electromagnetic (dynamic-type microphones) principles, an electric current is generated that reproduces the sound by alternating the electric current. Obviously, the design of the membrane plays an important part in the sound it produces. A dynamic microphone membrane is heavier to move than a condenser, making the response to the sound it perceives somewhat slower. Another factor to consider in terms of microphone design is its characteristics (i.e. at what angle from the centre of the microphone it perceives sound). Omnidirectional microphones perceive an equal amount of sound from all around, no matter the angle; cardioid microphones perceive most from in front, which decreases in proportion to the increase in the angle from the centre, receiving next to nothing from behind (depending on the particular microphone); bi-directional microphones perceive sound that reaches them from behind as well as from in front, but not from the sides.

These characteristics can be used close to the instrument for more direct sound, placed at a distance to record the acoustics of the room, or in pairs to record stereo. As soon as multiple microphones are in use, one risks phase problems such as comb filtering; this is especially important to consider when using pairs of microphones in stereo configurations, as they are often relatively closely spaced (see Chapter 4).

A microphone will also inevitably perceive the environment in which it is placed. The closer to the instrument, the more of the direct instrument sound is captured; the more distance from the instrument, the more the room is heard. Also, the closer the microphone is placed to the instrument, the more its timbre is altered, as all instruments project different frequencies in different directions. Additionally, the characteristics are crucial when setting the ratio between instrument and room sound. An omnidirectional microphone facing an instrument will capture more of the room than a cardioid microphone would in the same place. Further, the microphone actually perceives more noise from the environment than what we hear upon listening in the same situation, as our minds emphasise the sounds they find most interesting (and that is usually not noise, for instance, from lamps or the refrigerator). This means that the sound forwarded by a microphone is a distorted version of the internal balances of the auditory scene when compared to how we perceive the environment where the microphones are placed; however, when listening to the sound recorded through the microphones, we perceive the noise in the same way as the microphone picked it up. Of course, some of the noise we hear on a recording may stem from the equipment's self-noise; I will return to this matter very soon. It becomes clear, then, that microphones, and the way they are treated, are considerable contributors to recorded lute sound. If we were to admire a recording of lute music, finding the sound of the lute precious, we would perhaps ask ourselves: 'Wow, that sounds nice! Which lute is it?' but perhaps our enquiry would be more properly expressed by 'Wow, that sounds nice! What lute *and* technology have been used?' However, as it becomes clear that the construct of microphones (and other electric equipment) is indeed important to consider, I must also briefly address the electrical circuitry from which it emerges.

Into the circuit — join the resistance!

In the end, a mixing engineer (whether they are also the artist or the producer) works by modulating electric currents. The dB measured by LED's or a VU meter on their analogue mixer is not actually dB SPL (sound pressure level) but in fact dBv (voltage).[39] Also, all recording equipment has its own sound; Api, SSL and Neve, for instance, are textbook examples of this, as they are quite easy to separate aurally when compared. The specific colour that each of them possesses depends on their design (both internally and externally), which components have been used and how in the electric circuitry. While preparing this chapter, I made a journey of discovery by opening up a small four-channel mixer I have at home, to see what I found inside. (Due to legal considerations, I will not provide any exact information on the manufacturer's identity or product identifiers for each separate component; I will only refer to their type and function.) The channel strip (i.e. the pathway the recorded sound travels from input to master section) consists of input, a three-band equaliser (EQ), pan pot, auxiliary send and return pots.[40] Interestingly enough, what I found inside was just a number of resistors, capacitors, transistors, diodes and internal circuits (IC's).[41] A very brief mention of the function of each of these will suffice to unveil why it is interesting. A resistor provides resistance to the current that enters it, lowering the voltage; a capacitor charges and stores voltage, only to discharge it slowly when the power is cut off; a transistor can be seen as a specific kind of relay that can be used to amplify a signal; diodes ensure that electrons can flow in only one direction, often used to protect components; finally, IC's are actually circuits capsuled in a small plastic box. In my quest into my mixer I could identify two different IC's: An operational amplifier type that amplifies the signal considerably; and a voltage comparator type that compares two signals and passes on the strongest of the two. The interesting part, I think, is how our lute sound has now been reduced to energy storage, energy resistance and amplification. We can perhaps say that in speaking of acoustic sound, we focus on sound propagation, whereas in terms of electric circuitry it is more about

39 When speaking of electricity there are four parameters that we deal with: *voltage* (V), *amperage* (I), *resistance* (R) and *wattage* (W). Voltage refers to the force in which electricity is conducted; amperage is the current (i.e. flow) per second; resistance is the resistance the current meets when travelling through matter; and wattage is the labour produced by the others.

40 I will not go into detail on the function of these controls as it is not directly important for my line of argument; for more information on what mixers are and can do; see Huber and Runstein, *Recording*.

41 Obviously, the manufacturer does not wish people to see what is inside without breaking it, so I can only refer to what I saw through my investigation.

forming the signal; in other words, in dealing with acoustic signals we try to understand what happens and how we can deal with it, but in electric circuits we need to focus more on what we create, how it can be created and how that creation shapes the original sound.

It is not always easy to grasp directly how these basic functions can, for example, select frequencies (as we see in an EQ for instance), morphing them into a new sound. To give an example of how this can be done we can turn our focus towards a very simple EQ circuit that can be applied in speakers, recording equipment and playback hardware. At first, we can construct an easy low-pass filter (i.e. low frequencies are passed and higher frequencies are attenuated) by placing a resistor in series with a non-polarised capacitor; the capacitor builds up and stores voltage exponentially over time and a resistor reduces voltage. It is in the relation between these two that we can construct a cut-off frequency (see Equation 6.1):

$$f_c = \frac{1}{2\pi CR}$$

Equation 6.1. Equation for calculating the cut-off frequency in a simple RC circuit.

C is the capacitance in farads, R is the resistance in ohms (Ω) and f_c is the cut-off frequency. So, if we have a resistor of 10 kΩ (kiloohms) and a capacitor of 15 nF (nanofarads), we provoke a cut-off frequency of 1061 Hz. Now, if we were to reverse our circuit, placing the capacitor before the resistor, we would achieve a high-pass filter (i.e. passing high frequencies and not low), and by employing the same mathematical formula, we can calculate its cut-off frequency. By extension, if we want to create a band-pass filter (attenuating frequencies both higher than and lower than certain frequencies) we simply combine the two, making the current pass through a low-pass filter before a high-pass filter. Auditory circuitry, then, is simply about altering and moulding electric currents employing simple components in a specific sequence.

The fact that sound is now processed as electric currents presents us with some potential problems; electromagnetic and electrostatic energy may enter our circuits and produce noise that we did not intend to record in the first place. Also, each piece of equipment we use produces some level of self-noise (information in so-called 'specs' normally accompanies equipment to inform the buyer of these conditions for that specific product). Recording hot levels (i.e. recording at the highest possible volume without disturbing the signal, making the physical wire hotter) is one way to deal with self-noise. Increasing the volume when recording makes the recorded signal much louder than the noise — increasing the so-called signal-to-noise ratio (S/N ratio); the low amplitude noise can later be cut off, perhaps by using a gate (i.e. a tool where all sound below a certain dB level is silenced; of course, not without more or less affecting the frequency construct of the recorded sound). If the recorded signal is too low it blends with the self-noise and becomes next to impossible to separate without severely compromising the sound; so, we see that the S/N ratio is in fact important to consider. The question we must ask then is: how does increased amplitude upon recording affect the captured sound? Allow me once more to employ some basic physics. Newton's second law of motion (F = MA) teaches us that acceleration is proportional to the force that is applied to it.[42] Therefore we must differentiate between two instances: Firstly, when two identical sounds are played at the same time, the amplitude doubles accordingly (a 6 dB amplitude becomes 12 dB and a 1dB amplitude becomes 2 dB); secondly, when volume is turned up in a circuit, more voltage is presented to the entire signal, meaning that whether the amplitude of that signal is 2 dB or 50 dB, they both increase with the same force (2 dB + 6 dB = 8 dB; 50 dB + 6 dB = 56 dB). (All this can be traced back to the earlier-mentioned phase issues, such as comb filtering.) This is interesting if we consider that when music is being played through two closely-spaced speakers, both of these instances will occur; increasing the volume will induce an equal amount of voltage into the circuitry, but the identical parts (not to mention the non-identical parts) of the two sound streams (i.e. coming out from the left and right speaker) will behave differently, according to the basic principles stated above. From this we learn that a hot level will eventually influence the amplitude of the recorded signal;

42 The equation has, of course, been much refined, by Albert Einstein among others, since Newton first presented it, but this is beyond the realm of this chapter.

however, vast numbers of factors appear (more than we can investigate in this context) such as: where the speakers are placed within the room; what the acoustics of this particular room are; and what everything is made of (both speakers and room). The list goes on. To bring this exploration to a close, we can state that the level of voltage induced into the circuitry (at least at later stages in the process) affects lute sound. In conclusion, we see that electric circuits and the design of equipment (such as microphones, amplifiers and mixers) not only transfer sound from one instance to another, but also transform lute sound; yet, as we will see, this is only the first transformative process which lute sound encounters.

To bits and pieces — on PCM, Nyquist and jitter

So, the question then is, what happens when lute sound enters the digital domain? The keyword here is pulse code modulation (PCM).[43] PCM is a technique where one takes digital snapshots of a sound. An analogy from the movies can provide a quick introduction to the process: A film consists of thousands and thousands of still images; by fast-forwarding the film in front of a projector, we perceive the fast-going sequence of still images as moving pictures. The same (almost) applies to PCM: The film itself represents the time domain and each separate image represents quantisation. Since we are dealing with still images, it follows that we must divide time into segments of representation. In films we can speak of a frame rate of 24 frames per second (i.e. every second you are presented with 24 still images in succession). In audio, however, we speak of frequency rate. If the frequency rate is 44.1 kHz, it means that every second the ears are exposed to 44,100 still images of the sound. Each sonic still image consists of data describing the positive or negative amplitude at that moment out from a pre-set grid. To put it briefly, a 24-bit rate provides a denser grid (enabling each reading to be closer to the original sound) than an 8-bit rate (see Fig. 6.1 below).

43 To be more exact, *linear pulse code modulation* (LPCM).

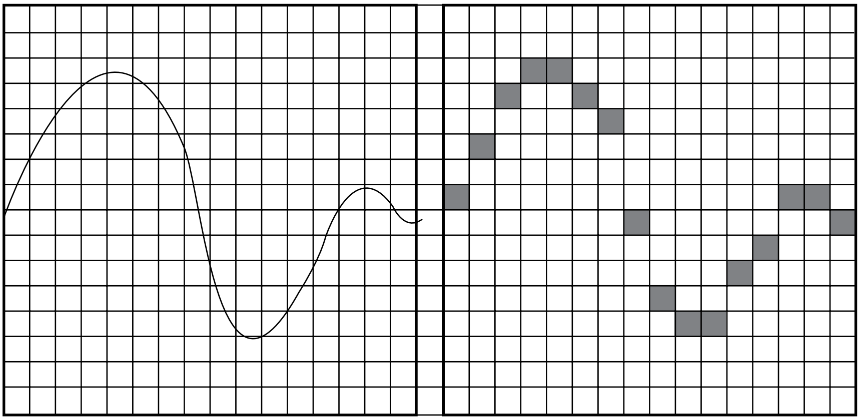

Figure 6.1. Illustration showing a sample rate and bit rate partitioning of a sound pressure wave.

So, when entering the digital domain, the sound segment gets partitioned horizontally (sample rate) and vertically (quantisation; bit rate) into binary code (i.e. 0s and 1s). Human hearing can perceive frequencies as high as roughly 20 kHz (i.e. 20,000 cycles per second), so we can understand that proper PCM coding is crucial for the design of lute sound. In order to cover the full range of human hearing we would perhaps believe it to be sufficient to divide the sound horizontally into 20,000 fragments per second to cover every cycle; however, each cycle consist of both positive and negative amplitude and therefore needs two readings per cycle (one for positive and one for negative). As a result, we must divide the sound segment into at least 40,000 segments per second to cover the full range of human hearing (see Figs. 6.2a and 6.2b below). The Nyquist Sampling Theorem states that '[i]f a function x(t) contains no frequencies higher than B hertz, it is completely determined by giving its ordinates at a series of points spaced 1/(2B) seconds apart.'[44] If the sampling frequency is less than two times the highest frequency of interest, one risks provoking aliasing errors, meaning that wrong readings create an unwanted phantom tone (see Fig. 6.2c below). This is the reason why high quality, modern digital audio software often offers much higher sample

44 'The Nyquist-Shannon Sampling Theorem.' Retrieved 6 September 2017, URL: http://www.princeton.edu/~achaney/tmve/wiki100k/docs/Nyquist–Shannon_sampling_theorem.html.

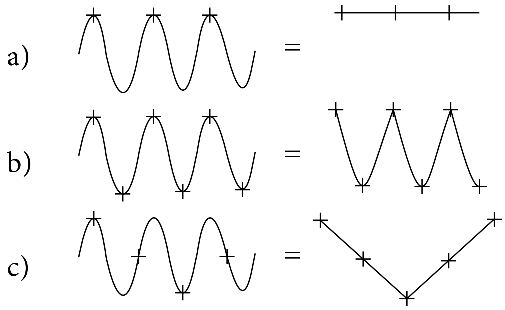

Figure 6.2. Example A and B illustrate the need for two readings per cycle. Example C illustrates aliasing.

rates than what is needed (such as 88.2 kHz or 92 kHz) — put crudely: The higher the sample rate, the less aliasing.[45] (Recall that there are higher frequencies at work than we can perceive, and those frequencies can cause aliasing.) One way in which developers have dealt with aliasing is by applying an anti-aliasing filter before the signal enters the sampling function. The logic is simple: cut away the undesired frequencies above the Nyquist limit before they are transformed into code.

Now, let us consider amplitude quantisation (measured in bit rate). Bit rate tells us the vertical density of the grid upon which an individual sample can be locked (as seen in Fig. 3); it basically informs us of how many 0s and 1s are being employed to describe each level of the grid; 3-bit offers eight levels (i.e. 000, 001, 010, 011 … 111), 16-bit subsequently offers 65,536 levels (0000000000000000, 0000000000000001, etc.) and 32-bit offers 4.3 billion levels. Again, the logic is easy: The denser the grid (i.e. the higher the bit rate), the closer the digitized audio resembles the signal it receives from electrical circuits. The only problem is, however, that no matter how high the bit rate is, it will still move stepwise from one level to another (see Fig. 6.3a below). Again, developers have provided a solution: dithering. Dithering implies that noise is added to the digital signal, making the signal bounce back and forth between neighbouring bit levels (see Fig. 6.3b below). Of course, this only makes the signal noisier, but if one subsequently averages the signal, one will even out the signal and make

45 National Instruments, 'Analog Sampling Basics.' Retrieved 13 July 2013, URL: http://www.ni.com/white-paper/3016/en/-toc3.

the bit levels smoother, resembling even more closely the original signal (see Fig. 6.3c below).[46] By following the simple (at least theoretically) steps of PCM coding presented up until now, working with as high bit and sample rate as the system allows, one produces better sound representation. (It must, however, be reduced to fit the CD's 16-bit and 44.1 kHz sample rate in the mastering process; I will return to this issue soon.)

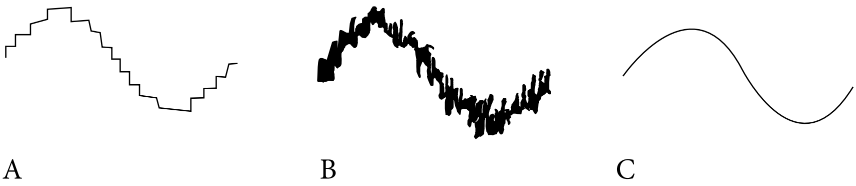

Figure 6.3. Example A illustrates an undithered signal. B illustrates dithering. Example C illustrates a dithered signal that has been averaged.

A final issue that we must address when discussing digital recording is jitter. An analogue to digital converter (ADC) or digital to analogue converter (DAC), for example, employs an internal clock to control when a signal is to be converted. When that internal clock signal does not correspond to the periodicity of the original signal, we get jitter. Jitter can affect both the time domain and the amplitude domain and result in noise, popping sounds, phase problems and altered frequency representation; it can be caused, for instance, by electromagnetic interference, as well as non-corresponding clocks between multiple equipment. To deal with this, many studios and software employ a master clock to control all other clocks; it can function both within the computer and control outboard hardware.

This second transformation of lute sound is perhaps even more clear than the previous one, as it deconstructs sound (or rather the electric representation of sound) into fragments that are described by numbers. For some recordings — more frequently in other genres than lute music — the story ends around here. The digital sound file is uploaded to free-to-use online services, such as YouTube and Sound Cloud, or sold through services such as iTunes and Amazon; broadcasting and

46 National Instruments, *Sampling Basics*.

different sorts of sound compression now become an issue, but, as stated earlier, I will only consider the physical CD. Before the music reaches the listeners, in this latter scenario, it must be attached to a physical format that can be distributed and sold; I will now look into that process — the third transformative process — to see how lute sound is affected by this technology.

Into the press

A CD is an optical disc that must follow IEC standard 60908 for Compact Disc Digital Audio (CDDA, often classified as Red Book CD). The part of the IEC standard for CDDAs (I will keep referring to them simply as CDs) that is interesting for us in this chapter is that a CD must incorporate a 44.1 kHz sample rate and a 16-bit rate standard. This means that high resolution projects (i.e. those with a higher sample and bit rate than those demanded by CDs) must then be converted into 44.1 kHz and 16-bit format (recall the Nyquist Sampling Theorem mentioned earlier; 44.1 kHz means that it can replicate pitches up to 22.05 kHz) using a converter that can be either hardware, stand-alone software or integrated into a sampling program (such as Cubase, Logic or ProTools). Before a master is forwarded to the manufacturer, error correction must be performed using a dedicated program for this task. This is to ensure that the CD will be read properly when it is duplicated by the manufacturer; the error rate cannot exceed 3%.[47] The finalised recording is then sent from the project team, either in physical or electronic form, to the manufacturer. Upon getting approval from industry professionals, a master CD is manufactured in glass and processed and shaped through various industrial machines; this will later be used as the template to manufacture the final product. The data is etched into a CD in approximately 0.5 mm (i.e. micrometre) wide pits that are tightly packed together, so compact that it would be possible to fit 60 CD 'grooves' into a single vinyl groove. Designed to be read by a 780 nm (i.e.

47 Owsinski, B., *The Audio Mastering Handbook*, 2nd ed. (USA: Thomson Course Technology PTR, 2008), 63, 65 and 165.

nanometre; near infrared) semiconductor laser, the CD is often coated in aluminium foil (or sometimes even gold) to enable the laser to reflect light onto the receiver more effectively. The pits are key to CD encoding: each pit edge is interpreted as 1 and each absence of pit edge is interpreted as 0. This is obviously a fragile technology so every data encoder in CD players also includes an error-correction function.[48]

The audio CD, then, delivers an audio data stream of 1.4112 Mbits per second (44,100 Hz × 16 bits × 2 channels [i.e. left and right stereo channel] = 1,411,200 bits/s); note that this is only the pure audio stream, not including the sub code and channel data (these contain information about index, track numbers, etc. that I will not concern myself with here). We have now, in this third transformation, reached a high level of abstraction, where the original lute sound has been transformed multiple times into chunks of bits (i.e. 0s and 1s) delivered at a rate of 1.4112 Mbits per second.

A brief note on recorded stereo space

The human auditory system (as represented by the outer and inner ear, the brainstem and the cortical structures associated with auditory information) localises sound by using three distinct methodologies. The first detects small differences in time between the two ears, called Interaural Time Difference (ITD); if a sound reaches the right ear slightly before the left, that sound is perceived as being located on the right side. The second method detects level differences between the two ears, i.e. variation in amplitude or sound-pressure level. This is called Interaural Level Difference (ILD). According to the 'duplex theory' it has been suggested that ITDs are used to localise low frequencies and ILDs are used to mentally place high frequencies. The third methodology detects variations in frequency content, or spectral cues, as caused by acoustic shadows provoked by the outer ear, or pinna, as well as the head. Each of these methods have their own designated pathway through the auditory system. Other

48 Huber and Runstein, *Recording*, 577–579; Owsinski, *Audio Mastering*, 60–73.

contributing factors that help to localise sound are, for instance, sight and sensory detection. If one hears a sound in close proximity but one cannot see it, it probably comes from behind. Similarly, if one stands in front of a loudspeaker with one's eyes closed, one will feel the sound pressure generated by the speakers on one' s body.[49] If a person is placed in a room together with a single sound source generating some sort of sound, we can speak of both direct sound and reflected sound reaching the ears at different times. But the situation will be quite different if we listen to recorded sound through headphones. Instead of being exposed to one signal from which we extract ITDs, ILDs and differences in frequency content, we hear two individual sources of sound that are independent of one another. If we only hear sound on the right side it is because there is no sound on the left. This is because a stereo track is not one sound source but two individual sound streams played at the same time in the respective ear. These sound streams can have different characteristics: one side may have reverberation signifying a great hall, while the other may sound like a small wooden chamber. In reality, we would hear the sound source as interacting with only one particular acoustic environment. If we play a stereo file through two loudspeakers instead of a pair of headphones, we would find ourselves in a similar situation, although it will be less obvious than through the headphones.

These situations clearly exemplify that space perception in real life is something other than it appears in music production. What appears to be an authentic space in which we perceive a source of sound may, in fact, be constructed out of several digital reverberators from competing manufacturers that all contribute to the sound production. As an example, American mixing-engineer Dave Pensado illustrates in a YouTube video how he uses three different types of digital reverbs on a single voice recording, that are panned, i.e. placed at different locations within the one sonic space.[50]

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49 Schnupp, *Auditory Neuroscience*, 177–221.

50 Pensado's Place, 'Get Great Vocal Reverbs Using Three Mono Sources - Into The Lair #84 (Pensado's Place),' YouTube video, 5'03", posted by 'Pensado's Place'. Retrieved 31 March 2014, URL: https://www.youtube.com/watch?v=wFg_lAw1ROc.

To give an even more technical example, consider a standard, uncompressed stereo WAV file format (44.1 kHz, 16-bit, linear PCM). In the part of the file where the actual sound data is stored, we find each sample presented chronologically (i.e. Sample 1, Sample 2, Sample 3, etc.). It is interesting to note that each sample consists of four bytes, where the first two are the sampled sound on the left side and the last two are the sound on the right side (see Fig. 6.4 below). This is called *stereo interleaved*. It is one single stream of data, 1s and 0s, that, through cyclic patterns, distributes information about the sound at a specific moment in time to every other left and right speaker. This is done at such a speed, of course, that it is not perceptible; however, there is some form of dialogue, almost poetry, inherent in this technology. It is so detached from human perception that it is truly artificial, but at the same time it is performed at such a speed that we perceive it as natural. In fact, the audio file specification of 44.1 kHz mentioned above actually means that we hear 44,100 samples chronologically played each second, each consisting of left/right designated bytes.

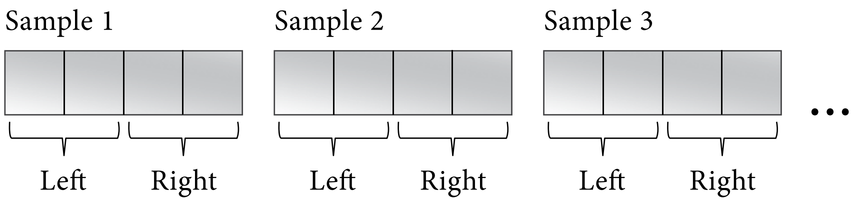

Figure 6.4. The organisation of stereo data in interleaved stereo files.

Lute sound as transformative process

Let us review our findings. First, we may say that lute sound moves from concrete to abstract and back again. In real life, sound consists of propagating periodic pressure waves, in which the particles of matter contract and expand. In an electric circuit, the electrons behave quite differently. They do not expand and contract in the same manner as pressure waves; rather it is the voltage that forwards the sound information by altering its amplitude. (As such, it is only now that Fourier spectrums start to resemble reality more than just being a presentational system.) At a third stage, this current enters its third phase, being the digital realm. Through a two-dimensional process (partitioned first horizontally and then described vertically), sound is being kept, processed, and communicated as 0s and 1s. Sound is now approaching its most abstract state. Following this, at the CD manufacturer, the digital sound is joined by additional data (such as channel data and sub codes) and physically coded into the disk. When the CD is put into a music player of some sort, this entire process is performed in reverse, only to reach our ears once more as sound pressure waves.

Secondly, in this pathway there are numerous possibilities for not only tone modelling, but also the appearance of direct errors in sound representation. At the microphone level, the transient response may misinterpret some high frequencies approaching it, depending on how slowly it reacts, as well as occurrences of self-noise provided by the circuit within the microphone. Self-noise is present throughout the analogue parts of the recording chain, but the electrical currents may also be subject to electromagnetic and electrostatic noise from outside the recording equipment. This includes wrongly-matched polarities (i.e. positive and negative conductors) within the equipment setup and grounding problems; at the digital level, jitter becomes a real issue as well as proper coding, decoding, and conversion; finally, moving towards the industrial press, data processing errors are often at work (this is, of course, part of the job for both manufacturers, producers and mastering engineers to minimise). These are just some of the possible errors in sound representation that we may encounter.

What, then, can we make of this? First of all, different stages of the transformation process present us with various considerations and approaches — what is problematic in one instance is not so in the next. Secondly, all stages of this modelling of lute sound consist of complex, intertextual considerations that incorporate not only maths, physics and technology as we have seen, but also aesthetics, representation and tradition. Behind the sound of the lute, as it is being heard during playback, lies numerous decisions, both intentional (by decision making during the entire recording process) and unintentional (the inner workings of technology that one has to deal with). We can present this process schematically:

vision (desired effect) —> available material (from instrument to equipment) —>
knowledge of how to utilise that material —>
the inner workings of all the equipment involved
(determined by manufacturer and tradition) —>
creative production and problem solving —>
dealing with unforeseen effects (such as code failure, jitter, electric noise) —>
verification, manufacturing and duplication —>
playback ≠ vision, but = finished, fixated sound

Recorded lute sound, then, appears as a dialogue between instrument, electricity and digital code — a dialogue that aims to reproduce sound true to its original, but which inevitably provides its own contributions to lute sound. The most obvious example of this is the ADC and the DAC, that break the signal into somewhat accurate pieces, only to rebuild the signal from these fragments rather than restoring it to its original. One may easily argue, and perhaps rightfully so, depending on the system employed, that the incoherencies between original and processed signal are not audible to the human ear; but the fact remains that the audio leaving the electric circuitry, or digital code, is something other than the sound originally produced by the lute. On this basis, I argue that it would be erroneous to draw a direct parallel between sound being recorded and sound being heard through a stereo, without taking into account the multifaceted process in-between. Although I have focused on the recorded CD, this same argument can be applied to other instances of music reproduction and sound reinforcement, such as live performances. When incorporating microphones in a live performance, some of the direct sound from the lute is heard while some is heard from the speakers (lute concerts rarely reach the same volume levels as rock stadium concerts). The musician, then, does not only need to consider the sound produced by their plucking of the strings on stage but also what version of their sound comes out of the speakers, blending with the acoustic timbre and reaching the audience. In the twenty-first century, then, a musician must acknowledge this dialogue between technology (whenever and however present) and instrument, in order to ensure a performance that is in line with the musician's intent.

According to my line of argument, recorded lute sound is the sum of the processes involved in its formation; it consists of multiple instances, all contributing a specific transformation. If we consider recorded lute sound as an isolated event, we can follow the evolution from generated sound into electric current; from electric current partitioned into approximated fragments described digitally; transferred from the pure digital realm into physical realisation of code imprinted on optical discs; restored into electric currents from these fragments through interpretation of digital data; reaching a stage of sound once more. This is again why I propose a term like biology in the title of this book, as recorded lute sound is something that evolves over time, not necessarily a fixed description of a present state. Also, these technological transformations are an active part of an aesthetic process, just as each individual part of an organism plays a significant role in what we perceive as that organism. Although I have taken the CD as my case, I think that whatever the format used for preserving a recording (or whenever technology is present in a performance), we must take into account in our evaluations (as scholars, performers or producers) the internal processes that constitute the whole — the biology of lute sound — rather than skipping ahead of technology and only thinking of what the musician performed, where it was recorded and how the recording sounds. We must stay critical to the entire process, both the parts that are deliberate (playing, microphones, mixing) and those that inevitably follow the process whether we like it or not (circuitry, digitalisation, errors).

Returning to the hypothesis mentioned at the beginning of this chapter, that performers can no longer consider their authenticity as detached from, or independent of, the production process, we see how the recording process presents numerous aspects to consider also in a cultural context. Lute recordings, or any other recordings for that matter, function as signifiers that are perceived by listeners, and from those signifiers they read a cultural debate: 'This is a recording from the 1970s or 2000s,' 'this sounds professional or amateur,' or perhaps, 'this sounds like an authentic or inauthentic Baroque recording.' By being aware of the biology of a recording, the performer may be permitted to gain further control of the recording as a signifier and, thus, also better communicate the initial vision. Additionally, the scholar may be more prepared not only to differentiate between performer or performance and technology, but also to address the gradual development from one to the other, or perhaps better formulated, the dialogue between them. On a recording, recording technology takes the role as a hidden instrument, or perhaps the filter through which we perceive the music. Is it really fair, for instance, to judge a musician's tone in a recording if the microphone used to capture him or her was not, in fact, the most suitable? Perhaps what we hear is not the tone of the instrument but rather a misinterpretation made by recording equipment. Similarly, a bad tone can be improved on by adjusting frequencies and dynamics, making the instrumentalist sound better than they might do alone without any microphone. Clearly, this has become practice in much of the vocal music of more recent times, where having a microphone has become part of the vocal technique, and the singer sings in a fashion that demands a microphone in order to be heard. In such cases, recording technology has in fact become part of a musician's aesthetics. This can also be seen in the often-complex composition of technologies incorporated by the electric guitarist, used for the purpose of finding that unique sound. For the Early Music performer, then, embracing technology during the stages of planning and recording can enable more coherent and successful communicative results than a mentality that musicians should do their thing while the technicians do theirs.