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LIVE

POPULAR ELECTRONIC MUSIC

'PERFORMABLE RECORDINGS'

by

CHRISTOS MORALIS

A thesis submitted for the degree of Doctor of Music

Supervisors:

Professor Simon Zagorski-Thomas

Associate Professor Andrew Bourbon

Professor David Osbon

LONDON COLLEGE OF MUSIC

DATE: JANUARY 2019

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ACKNOWLEDGMENTS

First, I would like to express my sincere gratitude to my supervisor, Professor Simon Zagorski-Thomas, for the continuous support of my DMus study and related research, and for his patience, motivation and immense knowledge. His guidance helped me during all my research and while I was writing this thesis.

Besides my supervisor, I would like to thank the rest of my thesis committee, Associate Professor Andrew Bourbon and Professor David Osbon, not only for their insightful comments and encouragement, but also for the hard question that prompted me to widen my research from various perspectives.

I thank my fellow musicians and research participants, Kostis, Luigi and Panos, for the stimulating discussions, for the sleepless nights we spent working together, and for all the fun we have had in the last four years.

I am also grateful to my partner, Alexandra, for the inspiring conversations and the emotional support throughout these years.

Last but not least, I would like to thank my family and especially my mother, Sofia, for supporting me spiritually while I was writing this thesis and throughout my life in general.

Thanks for all your encouragement!

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‘PERFORMABLE RECORDINGS’

ABSTRACT

This research focuses on Electronic Dance Music (EDM), or popular electronic music, specifically the way a band's live performance can have the same sonic attributes as a studio production by using production techniques and performance practices that work with these contemporary mediatized live performances. For the purposes of this research, an EDM live act has been formed using vocals, conventional instruments such as electric guitar and keyboards, and other more sophisticated electronic devices such as midi controllers and electronic drums. The emerging phenomenon of new types of bands or performers who try to bring the studio sound to the stage has created a gap between 'human' and 'non-human' that requires performers in this musical style to work with technology in new ways.

This thesis builds on research into authenticity and its relation to aspects of liveness in these types of live performances. More specifically, it builds upon research on Moore's tripartition of authenticities and the two forms of authenticity that are most salient in this process of 'musicking'. These are the first and the third person as described in Moore's (2002) model. The first-person authenticity relates to the extent to which the participants feel that the performers engage in authentic human expression through their performance. The third-person authenticity relates to the participants' assessment of what constitutes an authentic sonic example of a musical tradition or genre – in this case EDM. In addition to what it should sound like, third-person authenticity is also concerned with what are the appropriate 'tools' that should be used and factors such as the coherence between aural and visual, employment of skill, performativity and the constant awareness of a 'standard of achievement'.

The aim is to create a musical process in which all the participants feel that the band is performing authentically while being sonically faithful to the genre or tradition. The key is to combine machine accuracy with some aspects of human expressive performance in a way that maintains the integrity of the popular electronic musical style. Following on from the multiple theories that underpin this research, various methodologies have been followed. Qualitative and quantitative research methods have been followed, through interviews, video observations and audio data analysis.

Having said that, a real-time production and performance process has been developed called 'Performable Recordings'; that is, '*a type of music production that enables the artist to perform a musical piece live, using, in real time, the mixing and post-production processes that create the aesthetics of a studio-produced version*.'

This model intends to promote and support performers' emotional expression and the creativity that comes from spontaneity, musicianship, face-to-face performance and freedom of movement, which over the past years have been minimized or eliminated by contemporary production processes and performance practices. Furthermore, it creates opportunities for performers and musicians to get involved on stage with a broader range of modern musical styles and genres.

Keywords: electronic music, popular, live performance, studio production, real-time, liveness, musical descriptors, authenticity

CHAPTER 1

1. INTRODUCTION

The main artefact of this DMus submission is comprised of three creative works:

- The Produced Recordings, which are in the form of stereo audio files and are created through the mediatized process as suggested in this thesis.
- The Live Performance Examples of audio files without this mediatized process. Since this project is not about transferring existing productions on to this model, the live performance examples do not include machine timing, consistent (almost uniform) dynamics or highly 'artificial' production values as suggested in this research. These examples can also be found throughout this thesis and in the submitted folders alongside this thesis.
- The 'Performable Recordings' model in the form of Ableton Live Projects that contain all the information and settings discussed in this thesis. Video examples of live performances are also submitted alongside this thesis.

1.1 Statement of the problem

Over the past years, owing to the evolution of technology, music producers follow specific studio production techniques that alter the musical descriptors or combine performances that humans cannot possibly create or perform without any technology involved. In EDM, or popular electronic music, this extraordinary sound has defined the character of this genre.

These production techniques, along with the post-production or mastering process as it has been applied over the past two decades, have altered the perception of how an instrument or a voice really sounds, compelling performers to mime and lip sync, or to try to reproduce the sound of a CD in real-time and in live situations, in order to sound contemporary and be competitive in the music industry.

Over the past years, the big piles of hardware used in studios have turned into software installed in the small electronic chips of portable devices. Therefore, the extensive use of laptops and other electronic equipment is not only a music producer's privilege but lately also one for musicians and performers. New bands are born into the culture of electronic music that combine live vocals, musical instruments, laptops and other electronic devices in their live performances.

These contemporary live popular electronic mediatized performances often do not meet the sonic attributes of a studio-produced song, altering the performers' perception of how these actually sound and forcing them to act differently from studio to stage, with issues emerging around their performing techniques. It is essential to bear in mind the phenomenon that performers try to mimic with their voice the sonic characteristics of a studio-processed voice, changing their natural timbre and, most of the time, damaging their vocal cords. Also, musicians cannot perform accurately because of the discrepancy between sound and gestures caused by the extensive mediation technology used to match the sound of the studio production. Performers use fixed pre-recorded tracks or sound samples to bring the studio sound on stage but that results in non-spontaneous live and humanly performed sound.

From what has been said so far, the research problem focuses on the emerging phenomenon of a new type of band that tries to bring the sound of studio production on stage. They combine electronic devices such as laptops and conventional instruments to reproduce live the sonic characteristics of a genre that is mostly based on non-real-time production processes. In some other cases, they use pre-recorded material or karaoke playbacks, stripping out the human spontaneity from the live performance. The main reasons are the lack of expertise in the combination of techniques from studio production and the live production into one process, in real-time, as well as the less-explored contemporary performance practices that would allow them to perform expressively and to produce their tone accurately through this extensive use of mediation technology.

Furthermore, the discrepancy between the visual and the aural often challenges the audience's – and even the performers' – perception of what is real or not in this kind of performances. Also, the use of electronic instruments, devices and laptops, and the ease of this use, often calls into question the meaning of 'live', due to the opacity of the performers' activity or the lack of freedom of emotional expression during their performance.

Having said that, the gap that this research project is seeking to bridge is between the sound attributes of the studio production and the live sound, through the combination of machine accuracy with some aspects of human expressive performance in a way that allows bands to perform authentically while maintaining the integrity of the popular electronic musical style.

This research project is not about compensating for performers who are not of a high enough calibre, but about some aspects of musical styles which rely on machine-accurate timing, pitch accuracy and consistency in dynamics. This is not about researching ways to perform 'better' but combining the human with the non-human by requiring performers to work with technology in new ways.

1.2 Aim and objectives

Aim

The aim of this research is to develop production approaches and performance practice techniques that enable the combination, in real-time, of the sonic characteristics and aesthetics of a contemporary studio-produced song of popular electronic music with the live, human performance. However, the most crucial aspect of this is to create a musical process in which all the participants feel that the band is performing authentically while being sonically faithful to the genre or tradition.

Objectives

- **To balance effectively between the live human performance and the sonic attributes of electronic music.**

- **To combine the studio production and the live sound production in real time.**

The main element that affects the sound is the human performance; in addition, the sound the instrument makes affects the human performance of it. For this reason, on the one hand it is necessary to investigate and develop production approaches that will become an extension of these performance practices, and on the other hand it is essential to preserve performers' perception of what 'live' means. The key is the combination of machine accuracy with some aspects of human expressive performance in a way that maintains the integrity of the popular electronic musical style.

The combination of studio production and live performance in practical terms means a combination of real-time editing, mixing and mastering processes in the live sound process.

1.3 Significance of the study

Since most research in this area is concerned with recording techniques or electro-acoustic performances, this thesis will contribute to the less explored area of mediatized contemporary live performances by bridging the gap between the studio and the live sound production.

As this is a DMus rather than a Ph.D., it is primarily concerned with my own practice, but this production and performance practice techniques model will also enable musicians and singers working in these musical styles to perform their songs on stage with studio quality and the related aesthetic approach. Furthermore, combining studio-quality sound, in real time, with the live performance, will enable bands and artists to preserve and develop their trademark sound. This model, due to its portable nature and high-quality sound, could be applied to every type of live performance, from small pubs to big festivals and from radio and TV broadcasts to internet-streamed music. This will give researchers the opportunity to further investigate this production and performance model in different live situations.

By developing new production and performance practices, this research will contribute to the academic study of music synthesis and arrangement, sound designing, live performance, and studio and live production. Furthermore, this research will also contribute to the discourse around the meaning of 'live' in these types of contemporary mediatized performances.

CHAPTER 2

2. REVIEW OF RELATED LITERATURE AND PRACTICE

2.1 The theoretical background

As Collins and Rincón mention (2007), '*It is perhaps a general human habit to view the technological and the organic as opposites. It is certainly the case that the phrase "live electronic music" strikes many a music fan as oxymoronic*.' The theoretical background that underpins this research, along with the studies in studio and live production, focuses on the concept of liveness, as this is the key to understanding the nature and the philosophical aspect of this type of contemporary mediatized performances using Moore's tripartition of authenticities.

The *'performance ecosystem',* as described by Tom Davis (2011) and initially by John Bowers (2003) in Improvised Machines, is a term used to understand and conceptualize the environment of live performances. It is not a topographic reference, but rather refers to the relationship between performer, instruments and environment. As described by Simon Waters in Davis (2011), the performance ecosystem '*problematizes the "self-evident" boundaries between performer, instrument, and environment, recognizing the often-interpenetrating agency of each component of the performance*.' Through the functions of this system, the meaning of 'live' can be understood, and hence the Performable Recordings model can be built.

2.2 Liveness

Thorton (1995) writes: *'Live music does not exist without its recorded other. In other words, the concept of liveness in music was unknown until there was something not live – recordings – with which to compare it.'* Having said that, we should examine the case of the electronic music band The Bays. This group has deliberately never created recorded versions of their songs. Therefore, the meaning of 'live' developed out of a generalized concept of the recorded other and does not always come from a specific collation of the recorded and non-recorded versions of a track. Indeed, the term has outgrown this original narrow definition.

Auslander (2011), however, acknowledges that the concept of 'liveness' is used in various situations that do not meet this basic condition – for example, live broadcast, recorded live, online liveness, group liveness, digital liveness – suggesting that 'live' can occur between humans and technology without being spatially or temporally co-present. Auslander also suggests that '*any distinctions need to derive from careful consideration of how the relationship between the live and mediatized is articulated in particular cases, not from a set of assumptions that constructs live.'* (Transmediate, 2011).

According to Bown, Bell, and Parkinson (2006), '*1. Liveness can be based on the prior perception of performer activity or decision making. 2. Liveness and mediatization can co-occur. Live laptop music involves the performance of the mediatized. Mediatization may, in fact, amplify perceptions of liveness. From this viewpoint, audiences call something "live".'*
In this study, the concept of 'liveness' derives from a recognition of the performers' activity, also arguing that the co-existence of recorded and non-recorded audio may amplify the perception of 'liveness'.

Auslander also remarks that *'liveness is not in the thing but our engagement with the thing and our willingness to bring it into full presence*' (Transmediate, 2011). Therefore, the perception of liveness is related to the performers' occupation with their instruments or voice. According to Bahn, Hahn, and Trueman (2001), *'The instrument conducts touch, amplifies it and sonifies physical gesture. In return, the body responds to the "feel" of the instrument and its resulting sound.'* Therefore, to understand the meaning of 'liveness' in electronic music performances, we should examine it as a conception and the way this is perceived rather than as a quality or attribute.

2.3 Authenticity

Based on Moore's (2002) tripartition of authenticities, first-person authenticity relates to the individual's personal integrity of expression, second-person authenticity relates to the connection or empathy the audience feels for the performance, and third-person authenticity relates to how true to a particular culture or tradition the performance is perceived as being. In addition, we could see third-person authenticity as a two-part process: of being faithful to the genre or tradition, which is about making sounds that are true to the genre, as perceived by both audience and artists; and the second prong of being faithful to the recorded 'original' version. This is similar to the way that third-person authenticity in classical music can be seen as being true to the score or deliberately seeking out the 'original' unedited score.

According to Zagorski-Thomas (2014, p.47), '*Our perceptual system is built around the recognition of patterns of connectivity between stimulus and action, but this is a multi-modal system, and any incongruence between different modes affords a recognition that something is "wrong".'* The sound stimulus triggers our perception of what is real or not, and nowadays, although the performers can distinguish a 'fake' sound, thus distinguishing what is real and unreal, the question of the meaning of 'live' remains. To understand better the meaning of 'liveness' in electronic music, we should consider the reasons that humans in general seek 'real' or authentic performances.

The Oxford English Dictionary (2015) defines 'authentic' as *'known to be real and genuine and not a copy, true and accurate, made to be exactly the same as the original.'* However, the way we interpret and understand something is based on our culture. As Moore (2002) describes authenticity, *'Authenticity is a matter of interpretation which is made and fought for from within a cultural and, thus, historicized position.'* Furthermore, Keil and Feld argue (1994, p.296), found in Moore's paper (2002), '*authenticity only emerges when it is counter to forces that are trying to screw it up, transform it, dominate it, mess with it*.'

In electronic music, sometimes performers feel the need to try to prove themselves to the audience as 'real' performers. According to Johnson (2010), 'live' in a performance is 'the lack of a second chance.' DJ Whopper and Ricardo Villalobos (Wunderground, 2014), try to make mistakes in their performances to prove that they play 'live'. The spontaneity and imperfectness of human nature and, hence, of human performance are indicators of 'liveness'. Furthermore, the same thing also happens in the studio. According to Frost (2007), *'Some artists are troubled by the moral issues raised by editing, so they turn instead to "live" recordings in the belief that they represent a true and honest account of a real performance.'* Again, opposed to the extensive use of editing techniques that they feel make a performance 'fake', in contemporary rock music production, the objective of these artists is to sound authentic, through unedited performances. This can be interpreted as an attempt, to be honest, or true. The desire for this form of first-person authenticity, therefore, in music comes from the notion of being cheated or deceived. However, in the forms of electronic popular music that this project is dealing with there is also a third-person authenticity – of creating the right machine-like feel for the musical style.

Moore (2002) lists the most-used value terms in the discussion on the topic of music are *'authentic, real, honest, truthful, with integrity, actual, genuine, essential, sincere'*. Therefore, one basic aspect of 'liveness' in music is based on first-person authenticity, relating the correlation of the sound produced to the performers' activity and the coherence of visual to aural. All music strives for some form of authenticity, but different forms of music do it in different ways.

2.4 Emotional expression

According to Christophilou, I. D., (1985), '*Music is the art and the science that deals with the sounds and aims to express, with appropriate combinations of sounds, human ideas, and emotions.'* Consequently, when we listen to music, subconsciously we expect to understand human ideas and feel emotions. Therefore, if human ideas and emotions are not communicated, we could say that something is 'wrong'. If something is 'wrong', then 'authenticity' and, hence, 'liveness' are questioned. According to Marshall et al. (2012), '*This initial study has shown a link between the positive emotional response of an audience and the liveness of a performance. Additionally, a link was also found between a less live performance and a negative emotional response from the audience*.' In this research study, it is shown that emotional response is linked to the perception of 'liveness'.

Emotions are expressed through complicated variations in the sound in music. This variation in the sound is categorized into 'musical descriptors'. These descriptors, as presented by Jens Maden in his thesis (2011), are *'Pitch, Ambitus, Register, Harmonics, Harmony, Tonality, Brightness, Timbre, Loudness, Roughness, Tone attack/voice onset, Tempo/Speech rate, Articulation/pauses, Rhythm/meter/mode, Jitter/vibrato.'* However, we can categorize them into four main musical descriptors: dynamics, pitch, timing and timbre. These musical descriptors are often dramatically affected in electronic music, with less or sometimes no variation. Under these criteria, we could say that electronic music is emotionless music – although most creators of electronic music would undoubtedly dispute this.

Chordia (news.discovery.com, 2011), adds, *'People have this deep feeling that music should be authentic. And I think the reason why it's so important for music to be authentic is because it's so powerful emotionally… The more basic the emotions involved, the less listeners want to feel like that someone is simply pushing a button. They want to believe the music they love is an authentic human expression.'*

2.5 Definition of terms

2.5.1 Live performance

As Lalioti (2012) suggests, *'Liveness, an unmediated situation that can put us in the presence of other breathing human beings, is traditionally considered to be the uniqueness of performance*.' Having said that, this project seeks the momentary expressive variations of a live performance tied with Moore's (2002) first-person authenticity. This can be evaluated and seen also through Carlson's (2004) three concepts for evaluating a performance:

- *Appreciation of performers' employment of skill*
- *Engaging with 'repeated and socially sanctioned modes of behaviour' (2004,4) – a concept referred to by Judith Butler (1990a, 1990b, 1993) and others as performativity (or entrainment)*
- *A constant awareness of a 'standard of achievement' against which each performance is evaluated (2004,4)*

(Sanden, P. 2013)

The extent to which live performance has involved the manipulation and distortion of the original 'unmediated' performances has changed over time and varies between musical cultures, styles and traditions.

On a very basic level, microphones and amplifiers are affecting the amplitude and timbre of a performance. The fact that live sound reinforcement often now involves dynamic compression and pitch correction is a further incursion into the 'integrity' of the initial performance. Instruments such as samplers can also be seen as breaking the direct line of causality between the original sound and the activity that produces it in performance.

As Lalioti (2012) continues, '*Electronic technology used in musical performances thus puts the issue of performers', sounds' or instruments' materiality on a new basis and thus live performances are no longer considered to be specifically human activities.'* In this notion, the presentation of a musical piece with the use of electronic or other devices (DJing) could be considered as non-live music performance activity, and as a representation of a live music performance or studio-produced music.

All of these phenomena can be seen either as continuations of a creative tradition of using technology in music that began with using two sticks instead of clapping or singing through a tube to change and amplify the sound, or as technologies that de-skill or strip the emotion out of the performance and thus undermine its authenticity. Neither is 'true': they are judgments that individuals ascribe to particular forms of activity.

Coming back to the 'uniqueness of the performance' as mentioned earlier, another aspect of the live performance is sharing experience and involvement. According to Bahn, Hahn, and Trueman (2001), '*the social context of musical performance is built on shared sensibilities and embodied practices. Seeger observes: All human communicatory systems produce concrete visual, auditory and/or tactile products that in their own respective forms of transmitting the energy used in their production are models of the act of production on the parts of their producers. (Seeger, 1977: 23).'* The important point that the Bahn, Hahn and Trueman quote reinforces is the shared aspect of Moore's definition of authenticity and in particular that the perceived authenticity at any given moment is negotiated by all participants such as performers, entrepreneurs, venues and audiences. In addition, Davis (2011), suggests that '*music as practice is an active consideration of music formation such that the listeners are given an active role in the process of music creation*.' This ties with Small's (1998) concept of *'musicking'* where the author suggests this term as communal activity even when it involves technology.

From this point of view, the live performance in this project is looking for the first-person authenticity related to the question of liveness and creative control in the following ways: on a very basic level it relates to the relationship between the gesture made by the performer and the sound that is produced, but it also relates, less importantly in this instance, to the freedom the musician feels to vary their performance. The performers, and hopefully the audience, should feel that they are in control of the performance, that they have enough agency.

Another aspect of the live performance is the improvisation. As far as variation and improvisation are concerned, the domain of performance where creativity is most conspicuously present is improvisation. The ethnomusicologist John Baily writes that improvisation *'implies intentionality, setting out to create something new in each performance, "composition in real time" as it is sometimes described.' (Baily, 1999: 208)*. As Clarke E. suggests, (2005), *'Novelty and uniqueness, which Reber (above) takes as defining attributes of creativity, are central to that powerful Romantic notion of creativity which still dominates our culture – creativity portrayed as the mysterious appearance of the radically new, apparently from nowhere.'*

Although the approach is to find new ways to perform through technology, the creative aspect should be maintained. This could be, for example, a drum fill between specific groove patterns, or improvisation on the hi-hats. When it comes to synthesizers or guitar, this can be related to different timings and variations in the dynamics and other musical descriptors. The lead synth or guitar solo parts could vary musically, composing a new melody every time. The third-person authenticity relates to the idea of the music sounding right to the performers but also to the audience. Since this project is not about transferring existing productions on to this model, the basic aesthetic is driven by machine timing, consistent (almost uniform) dynamics and highly 'artificial' production values: values highly processed in ways that maintain certain features of a sound but inhibit or reduce others because they are not musically necessary and create a less simple, messier sound.

In the notion of Zagorski-Thomas's (2014) 'Sonic Cartoons', a low-pass filtering or equalization of the vocals creates more clarity in the mix by removing some of the natural low-mid – which isn't necessary for hearing the melodic shape, lyrics and timbre – and by slightly masking some of the other instrumental sounds.

The machine timing and consistent volume are also partly about creating a 'sonic cartoon' of simplicity and clarity, especially for making entrainment (i.e. dancing) easier. However, they are also cultural markers of modernity, dance culture and the excitement of an incessant high-level energy. The third-person authenticity is also found in ideas about how each of these specific songs should sound and, therefore, where the performers have scope for first-person authenticity in their performance. However, there are certain musical descriptors that cannot be changed because they are right for the culture of EDM and these can be different from song to song.

2.5.2 Studio production

In contemporary music production, and particularly in electronic music, the musical context is created mainly by a combination of edited live performances and computer-based music, 'machine performances'. Furthermore, the mixing and editing techniques applied in contemporary studio production of popular music and electronic music create non-humanly performed sonic attributes with characteristics such as extreme consistency in timing, timbre, dynamics and pitch. However, according to Zagorski-Thomas (2010), *'players seem to be trying to sound more like machines, and on the other hand, programs creating computer-based music were often aiming to make the machines sound more like people*.' Fuelled by this, although the electronic music is defined by its artificial sonic characteristics, nowadays producers try to give a more naturally performed aesthetic to many genres of electronic dance music such as House, Dubstep, Deep House and Drum 'n' Bass.

In this research, the term 'studio production' refers to the aesthetics of a professionally edited, mixed and mastering processed audio to meet the quality standards of a contemporarily released song.

2.5.3 Performable Recordings

The Performable Recordings model is '*a type of music production that enables the artist to perform a musical piece live, using, in real-time, the mixing and post-production processes that create the aesthetics of a studio-produced version*.'

In other words, the Performable Recordings model will combine the editing, the mixing and the mastering process, in real-time, fed by the live performance of the band. For this reason, it is necessary to combine the mixing and mastering process with the overall timbre of the instrument or the singer's voice by which a better understanding can be developed of the necessary production and performance techniques that need to be followed.

2.6 Practice review

Contemporary bands often combine the latest available technology to bring studio aesthetics to the stage. Their live performances may include powerful computers that can perform challenging and demanding audio processes in real time, midi controllers and other sophisticated devices, as well as traditional instruments that may have more advanced technology, such as real-time audio-to-midi converters.

One of these bands is the Pinn Panelle. This band has a singer that also plays guitar, a bassist, a drummer and a keyboard player. The electric guitarist and the electric bass players combine their instruments with wireless and wired midi controllers to manipulate their sound, in real time, and at least potentially to control other hardware. The drummer uses a hybrid drum kit: an acoustic drum kit combined with midi triggers and midi drum pads. Lastly, the keyboard player uses keyboards to reproduce sampled and synthesized sounds. Also, the guitarist has a smaller keyboard and performs sampled vocal phrases to recreate the aesthetics of a processed vocal part similar to those found in Dubstep music. Pinn Panelle, in the example of their live cover 'Skrillex – Scary Monsters and Nice Sprites' (Pinn Panelle, 2011), combines, in real time, pre-produced audio samples and live performances. The result is the typical sound of an electronic band that performs live, lacking the timing consistency found in electronic music. The overall mix audio balance, although it is very professionally done, is not quite of the high audio fidelity found in a studio production.

Similar to this is the Submotion Orchestra, who perform to a click track, and all effects are synced with the overall tempo track. Although their sound and the way they have mixed their instruments sounds more like a studio production, they lack the pitch and timing accuracy integral to the culture of recorded electronic music. Furthermore, the drummer, although he mainly uses electronic drum samples, which help him to get closer to the electronic music aesthetics and culture, his performance is left natural and unprocessed, similar to a live performance on an acoustic drum kit.

Another example is the music artist Shawn Wasabi, who uses midi pads to trigger, in real time, different audio samples that all together construct a song. In this case, this performing and production approach creates aesthetics that are similar to DJing. By triggering already mixed studio-produced samples, he can only manipulate further the sound of these existing recordings by applying various effects such as reverb, delay, modulation, pitch shift or time stretch. Similar to this is the artist Afishal, who triggers audio samples but in this case with the use of drum pads.

The band Destroid use a large number of sound samples in their live sets. As with Swan Wasabi and Afishal, this band use midi pads to trigger their samples. In addition, the lack of traditional instruments such as guitars or vocals helps define the machine accuracy found in this style of music but lacks the variation found in a live human performance.

There are also artists who combine pre-recorded sound samples and audio loops with more naturally live performed instruments or vocals. One of these artists is the duo Darkside. In this case, the Ableton performer triggers audio samples, or loops, and acts as a DJ applying various effects and manipulating their sound further. The guitarist performs along with extensive use of different sound effects on his guitar to blend better with the aesthetics of electronic music. This model of combining sequenced or pre-recorded material with live performance is commonplace but mostly doesn’t provide the aesthetic of pitch and timing correction in the live setting.

Apart from bands that combine technology with traditional instruments, there are music artists and singers who use backing tracks in their live performances. Sometimes this is done to enrich their live sound or to include instruments or sounds that cannot be performed live with the same sound attributes as those of the studio-produced ones. This sometimes involves performers miming to the backing tracks because of the needs of the show. For example, Beyoncé’s (2013) performance at the Super Bowl 2013 halftime show combined live instruments with backing tracks of studio vocals and the brass section from the song ‘Crazy in Love’. The band Coldplay (2016) at the Pepsi Super Bowl 50 halftime show used a recorded string section on the song ‘Viva la Vida’. Furthermore, artists such as Britney Spears or Justin Bieber might perform on stage with full playback or half playback (karaoke) because of the consistency they want to keep in their voices while they perform demanding dancing moves.

All these approaches to some extent involve stripping out the human spontaneity and expressivity to re-create the studio sound aesthetics. Furthermore, even if there is real-time audio processing, there are still some crucial aspects of a studio production missing, like the timing consistency. For example, Daniel Green (2012), the FOH/studio engineer and producer for Coldplay, during their show at Hollywood Bowl, explains the real-time processing audio plugins from waves on the Digico SD7 mixing desk. This brings a lot of the studio techniques on stage, but the live instruments lack the kind of timing consistency found in popular and popular electronic studio productions because of the natural live human performance.

The novelty of the practice presented in this thesis is based on the ability that the performers have, performing live, without any pre-recorded material, and at the same time balancing human expression, improvisation and the sound attributes that define the nature of this genre. It’s a balance between feeling that they perform authentically and authenticity in the sound of popular electronic music.

CHAPTER 3

3. METHODOLOGY

3.1 Types of research

The research methodology followed is based on the combination of qualitative and quantitative research methods. According to the multiple theories that underpin this research, multiple methodologies have been followed.

3.1.1 Correlational

Through a correlational approach, the variance tolerance of the sonic attributes regarding the studio production aesthetics of electronic music and performers' emotional expression has been balanced.

3.1.2 Quasi-experimental

The combination of electronic instruments and traditional instruments found in EDM live acts, along with the real-time tweaking and triggering of effects and sounds, needs a group of people capable of performing this kind of music live. This project involves a band comprising a singer, guitarist, drummer, keyboard player and DJ, the last having the role of triggering and tweaking sounds with the use of electronic devices such as samplers and midi controllers.

Although this kind of band is appropriate for the validity of the experiments and is capable of performing at a professional level in a wide range of electronic musical styles, this research does not meet all the conditions of a true experimental design. It has been carried out with specific musicians and, in line with the nature of a DMus, is focused on the researcher's own practice rather than on solving generalized problems.

3.1.3 Descriptive

Video observations taken from the performances in the studio environment and laboratory environment, along with formal and informal interviews with the performers and self-observation and experience gained by also participating in these experiments as a performer, enabled me to identify the factors that affect the performers' sound perception. Also, these videos helped to develop techniques that work with performances and hence serve the production process model.

3.1.4 Responsive evaluation

The data collection and thematic analysis of interviews and videos and sound analysis through sound software helped identify issues and suggest changes in the production process of the Performable Recordings model and the performing techniques throughout my research.

3.2 Process plan

The Performable Recordings process model is shown in Fig. 1:

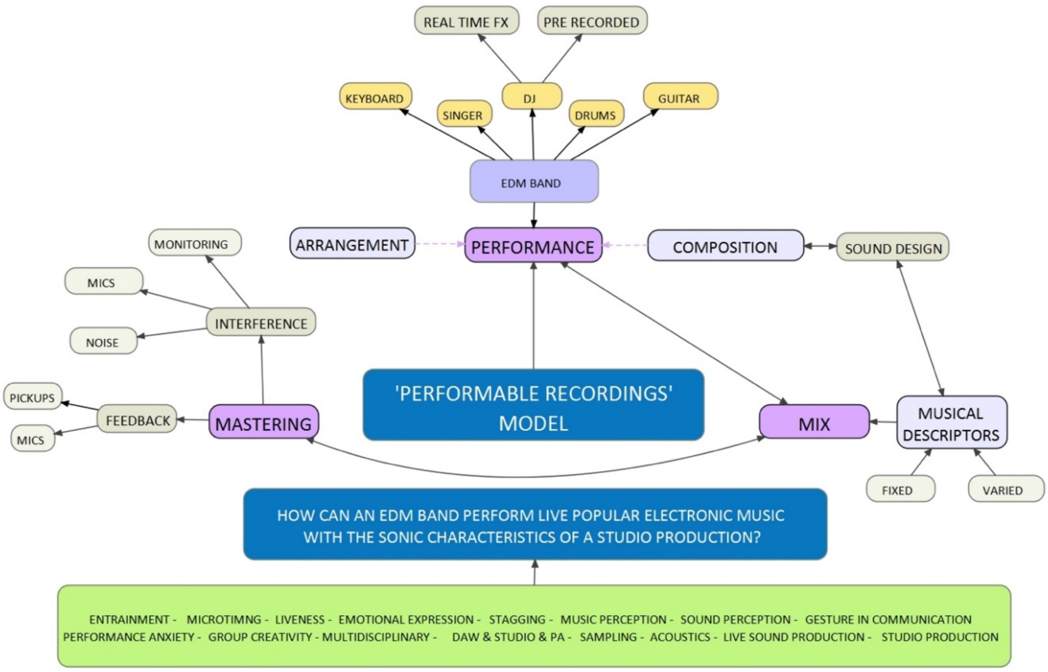


Figure 1: MORALIS, C. (2015) 'Performable Recordings' [photograph] (Designed with VUE software)

3.3 Performable Recordings

The Performable Recordings model has been based on the following four stages as shown in Fig. 2:

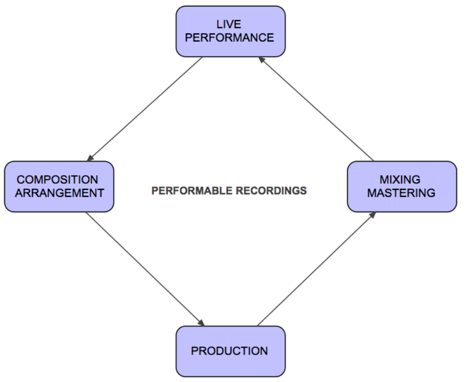


Figure 2: MORALIS, C. (2016) 'Performable Recordings Process' [photograph] (Designed with VUE software)

3.3.1 Composition / Arrangement

As Christophilou (1985) suggests, music is the combination of sounds that constitute the expression of my ideas and emotions. Consequently, appropriate music composition and arrangement can help the mixing process and, in turn, a good mixing process can help to express the idea of the song.

3.3.2 Production

During this stage, the research has been focused upon audio treatment as a process of correcting the input signals as well on the importance of appropriate sound design and combination. According to Concato (YouTube, 2014), *'...When people chose the right sound, the groove is right, the programming is right, you push the fader and is already great. It mixes itself.'* With the appropriate sound designing, where applicable, the mixing process has been minimized, reducing the processing of the sound, and preserving the musical and expressional context of the performance.

3.3.3 Mixing / Mastering

The mixing process has been done in the digital domain using internal or external DSP effects to control the sonic attributes of the production accurately. The objectives of the mixing and mastering process are accuracy and the ease of recall of their settings. The mastering process is as transparent as possible to preserve the sonic characteristics of the performance. However, studio mastering processes must meet the demands of contemporary productions, and this also relates to live sound production.

3.3.4 Live Performance

Interviews and discussions have been conducted with the band members to understand the gestural response with their instruments, the entrainment process, and the sound perception. This was required to effectively create the Performable Recordings model by securing their convenience and confidence during their performance.

CHAPTER 4

4. EXISTING TECHNIQUES

EDM is a genre that over the past decade, and especially the most recent years, has been characterized by the excessive application of sound effects, with the use of digital audio workstations that allow multiple instances of these effects, as well as by the extreme loudness levels achieved in the mastering process. For the purposes of sound designing and timbre definition, these processes may include excessive use of audio compression, saturation, modulation, distortion, equalization and limiting either applied on an already produced sample as a further sound designing process or on recorded or synthesized audio waveforms.

In addition, recent production techniques focus more on producing impressive quality sound rather than on preserving the realism of the performance. With the evolution of technology, music producers can create music that is, according to Zagorski-Thomas (2014) '*inhumanly*' performed. The sound produced, as explained earlier, is often impossible to perform live because the layering of sounds with multiple recordings on top of others and the excessive use of non-real-time effects and editing approaches, altering the timbre, dynamics, timing and pitch of the performances, is the trademark of the contemporary style of productions.

According to Pretolesi (2015), '*there are certain records that if you were to turn off the plugins, you would lose some of the body and the emotion/soul. So, I think the mix engineer who can maintain the integrity of the song but also add an element that brings out the emotion in the song, will have a chance at this career.'* Pretolesi points out the phenomenon that producers using excessive sound effects may cause the performer's initial message to be altered or eliminated during this production process.

The Performable Recordings model is based on the combination of the digital audio workstation 'Ableton Live' with laptops, sound cards and digital signal processing (DSP) cards, in real-time, to deliver the sound characteristics of a studio production and the natural-feeling sonic response to the performers' activity.

For the purposes of this research, it was necessary to focus on recently introduced technologies such as the adaptive tonal linearization, pitch-tracking equalization, the phase interaction mixing process, matching equalization, real-time midi quantization and real-time envelope shaping. The purpose of these technologies used in this research, whether during the production of the sounds or the live performance, was to deliver the sonic attributes of 'mastered' audio eliminating the necessity for a post-production process on the audio waveform.

4.1 Pitch-tracking equalization

Pitch-tracking equalization is a technology that tracks the pitch of monophonic audio signals and can move the band frequencies relative to what is being played, *'making it possible for the first time to naturally control the fundamental frequencies or harmonics of a track'* (Soundradix, 2015). This technology dramatically improves the natural harmonic balance of the audio signal without affecting its natural timbre.

4.2 Adaptive tonal contour linearization

This digital effect automatically '*detects and removes resonances, excessive equalization, roll-offs, and comb filtering, linearizing the frequency response of a signal automatically'* as well as *'perform[ing] mastering grade adaptive, free-form, and graphic equalization'* (Zynaptiq, 2015). The purpose of using this technology is to fix the audio signal by removing all the unnecessary information that could affect the overall mixing process. Furthermore, this plugin is based on the equal-loudness contours theory as it is explained by Fletcher and Munson (1933).

4.3 Phase interaction mixing process

This technology dynamically rotates the phase between the mixed audio signals to match the maximum correlation of the phase between different sounds. This will help to minimize the equalization and compressing process by keeping the timbre of the sound as natural as possible. Specifically, it *'minimizes overlapping frequency cancelations between instruments within the mix, improves mono compatibility, and brings back the depth and focus lost when out-of-phase frequencies in the mix cancel each other out'* (Soundradix, 2015).

4.4 Matching equalization

The matching EQ utilizes sonic fingerprinting to help preserve the initial timbre of acoustic instruments (including the electric guitar) since factors such as variation in the temperature or the extensive use of the strings may cause changes in the timbre of the instrument.

4.5 Real-time midi quantization

The Max for Live patch demonstrates real-time midi quantization by conforming incoming MIDI notes to the 'grid' in user-defined intervals. The patch works as an intermediate layer between incoming note events and the destination, which could be anything. It is, in essence, a MIDI transformer (much like an *arpeggiator*).

The patch takes incoming MIDI note data from Ableton Live, separates it into pitch and velocity and then stores them in a *list*. This *list* (a two-dimensional array), is read out in reverse (**LIFO: L**ast **In F**irst **O**ut) at regular clock intervals (as specified by the user) synced to the Ableton Live host clock.

Since the notes are always read out at the clock pulse, each Note ON (and OFF in this case) is on the musical grid. Obviously, in real-time performance, the patch quantizes to the next clock interval and can therefore only correct notes that are played earlier than intended. Fig. 3 shows how the process works.

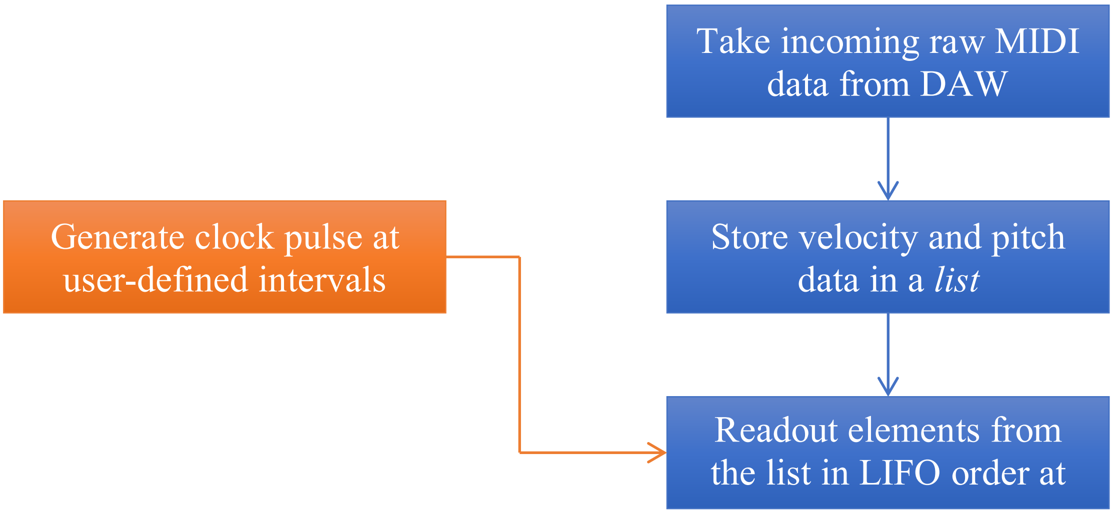


Figure 3: DAS, S. (2015) 'Real-time Midi Quantization' [photograph]

4.6 Real-time envelope shaping

This technology processes incoming audio as well as generating a MIDI message stream for controlling other instruments, allowing users to sculpt custom LFO curves and shapes.

CHAPTER 5

5. TIMBRAL CONSISTENCY

This chapter discusses the innovative aspects of the creative practice for individual instruments in the setup; specifically around the timbral consistency, an important aspect of the studio production of electronic music. The typical contemporary production process chain, excluding the composition, arrangement, and rehearsal process, is as shown in Fig. 4 (MPG, 2016):

Figure 4: MORALIS, C. (2016) 'Music Production Process' [photograph] (Designed with VUE software)

However, regarding the Performable Recordings model, since it is a real-time procedure, 'the lack of a second chance' as explained by Johnson (2010) is what defines the nature of this process. Therefore, the overdubbing process will be excluded from this chain leaving the performance, editing, mixing and mastering – four procedures that happen together in real time.

To preserve the perception of 'liveness', one of the important factors is to balance gesture and sound response. According to Bahn, Hahn, and Trueman (2001), physicality, feedback, and gesture – the reintegration of the body in electronic music – are all key to maintaining and extending musical/social traditions within a technological context. As Zagorski-Thomas (2014, p.65) suggests, *'timbre is a function of the nature of the object making the sound as well as the nature of the type of activity*.' Since the creation of the samples is a procedure that is based on the producer's aesthetics, the explanation of the production process does not go into full detail regarding the settings and parameters of the plugins used, but rather focuses on the innovative aspects of this production approach. However, it is necessary to explain main production processes and instruments' setups since they play a significant role in the overall mixing process and the overall loudness levels of the songs.

5.1 Sequential layering

The approach followed in this research for creating the drum and percussive sound samples, was inspired by the concept followed by 'Synth Kick' by Sonic Academy (2016) as well as by 'Big Kick' by Credland Audio (2016). The idea in both samplers is that there are two layers, with often a punchy sample at the beginning of the sound and a second synthesized or sampled sound that defines the body and the tail of the overall sound. See Figs. 5 and 6.

Figure 5: SONIC ACADEMY. (2016) 'Synth Kick' [photograph]

Figure 6: CREDLAND AUDIO. (2016) 'Big Kick' [photograph]

However, the limitation in both samplers, even though the digital audio workstation can automate these, is the effective real-time response and varying the sound according to the drummer's performance. For this reason, the drum samples have been produced separately and then contained within the Battery Sampler by Native Instruments, where parameters such as attack, decay and pitch can be modulated effectively according to the drummer's dynamics. For the creation of the samples, a multi-stage effect application process has been applied to control the sound effectively in order to reach the highest possible loudness levels. Every stage in the mixing/creation process is treated as a mastering procedure in order to achieve a sound that will not need any post-production.

In order for the sound samples to work better in the real-time mixing and mastering process, the production approach followed is based on a sound sample construction of two different parts. The different and innovative aspect is that these two layers are not on top of each other but follow each other and are layered sequentially. More precisely, first sounds the 'character', which defines the timbre and the expression, and then comes the 'pitch', which defines the note or tone of the sample. See Fig. 7.

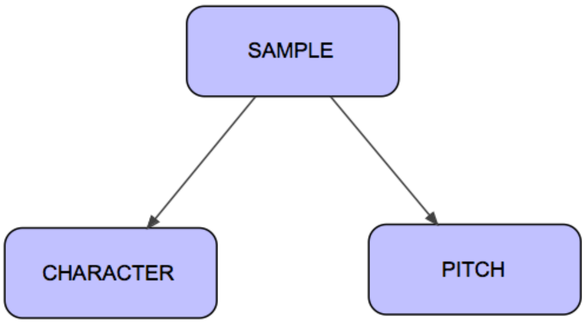


Figure 7: *MORALIS, C. (2016) 'Sample Production' [photograph] (Designed with VUE software)*

Furthermore, as seen in Fig.8, in order to create the character (the first part of the sample) different layers of sounds have been used. This helps to achieve a rich and unique sound. The separation of the sound sample into two different parts also helps to control the audio sample effectively and more precisely, in many different ways, rather than applying one audio process to the overall sample.

For a more detailed explanation of the production of the main drum character sound samples, see Appendix 2.

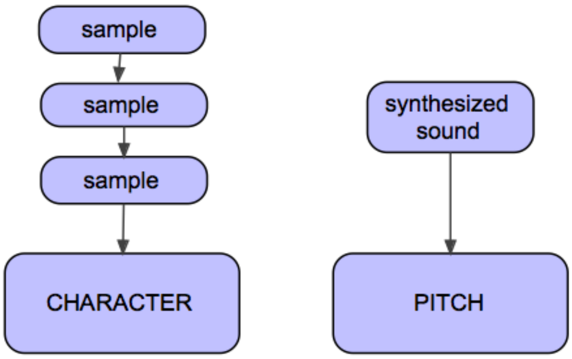


Figure 8: *MORALIS, C. (2016) 'Sample Representation' [photograph] (Designed with VUE software)*

Apart from the different sound layers that give a unique character to the drum sample, producers also tend to tune the sample according to the song's key. In the case of the drum kick, as Drumcode (2016) explains, producers tend to tune it according to the key of the song or a 5th or 7th lower than the bass instrument. In addition, any note from E1 through to B1, and especially A1, responds really well to a club's audio system. As Drumcode suggests (2016) '*Most club sound systems are tuned to give the most power and punch at 55 Hz, which directly corresponds, in pitch, to A1*.'

The role of the second part of the sample, as mentioned earlier, is to define the note. Inspired by the recently introduced pitch tracking equalization technology, the pitch sample that follows the character sample has been produced for one whole octave covering all seven notes. The main reason for following this approach is to improve the harmonic compatibility of the kick and bass.

Since in EDM the sub frequency range of the kick is maintained, the different harmonic focus of the kick, in regard to the note of the bass, should be avoided. Taking a step further, rather than tuning the kick to the scale of the song only, the kick, snare, and toms are tuned according to the chord played during the song. The next example shows the dramatic improvement of the harmonic blend between kick and bass during the 'pitch follow' approach of the kick according to the song's chords.

- **FIXED PITCH KICK:** Audio Example 1
- **VARIED PITCH KICK:** Audio Example 2

For a more detailed explanation of the main drum pitch sound samples, see Appendix 3.

- KICK

Fig. 9 shows the kick's pitch sound samples and their respective frequencies:

NOTE	MAIN Hz	HARMONIC Hz
A1	54	108
A#1	57	114
B1	60	120
C2	66	132
C#2	70	140
D2	73	146
D#2	77	154
E2	81	162
F2	85	170
F#1	47	94
G1	49	98
G#1	52	104

Figure 9: MORALIS, C. (2016) 'All Kicks Hz Table'

- **ALL KICKS:** Audio Example 3
- **ALL KICKS - TOP:** Audio Example 4
- **ALL KICKS SUBS:** Audio Example 5

- **SNARE**

Fig. 10 shows the snare's pitch sound samples and their respective frequencies:

NOTE	MAIN Hz
G3	200
G#3	205
A3	216
A#3	227
B3	242

Figure 10: MORALIS, C. (2016) 'All Snares Hz Table'

- **ALL SNARES:** Audio Example 6
- **ALL SNARES - TOP:** Audio Example 7
- **ALL SNARES SUBS:** Audio Example 8

- **TOMS**

Fig. 11 shows the pitch sound samples of the toms, and their respective frequencies:

NOTE	MAIN
D2	73
D#2	77
E2	81
F2	90
F#2	94
G2	99
G#2	104
A2	110
A#2	115
B2	121
C3	134
C#3	137
D3	144

Figure 11: MORALIS, C. (2016) 'All Toms Hz Table'

- **ALL TOMS:** Audio Example 9

- **ALL TOMS - TOP:** Audio Example 10

- **ALL TOMS SUBS:** Audio Example 11

- **BONGOS**

Fig. 12 shows the pongos' pitch sound samples and their respective frequencies:

NOTE	MAIN
D2	146
D#2	154
E2	162
F2	180
F#2	188
G2	198
G#2	208
A2	220
A#2	130
B2	242
C3	268
C#3	274

Figure 12: MORALIS, C. Bongos Notes (2016) 'All Bongos Hz Table'

- **CYMBALS**

For the creation of the cymbals, as shown in Fig. 13, a more traditional layer method has been used. To balance between gesture and sound response, the approach is to blend natural cymbal sounds with the electronic style sounds. However, since it is necessary to minimize the mixing process during the performance, different steps have been followed to blend the audio signals together (as shown below). Furthermore, to improve the aesthetics of the cymbals as well as their mono compatibility, monophonic electronic-style cymbals have been added to the cymbals sound. The addition of the electronic samples reflects the necessity for parallel compression in order to improve the overall presence. For a more detailed explanation of the cymbal sound samples, see Appendix 4.

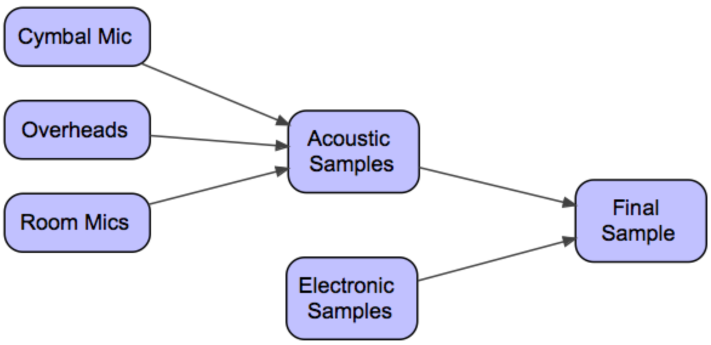


Figure 13: MORALIS, C. (2016) 'Cymbals Creation' [photograph] (Designed with VUE software)

The Electronic Drum Kit sample pack is loaded into a sampler for further manipulation of its sounds. More specifically, the Battery 4 sampler by Native Instruments has been used for the purposes of this research. Modulated parameters such as attack, decay and sample pitch are controlled internally while velocity, pitch, and sample selection are controlled outside the sampler by Ableton Live's integrated midi effects and devices. Fig. 14 shows the modulated parameters:

MODULATED PARAMETERS:

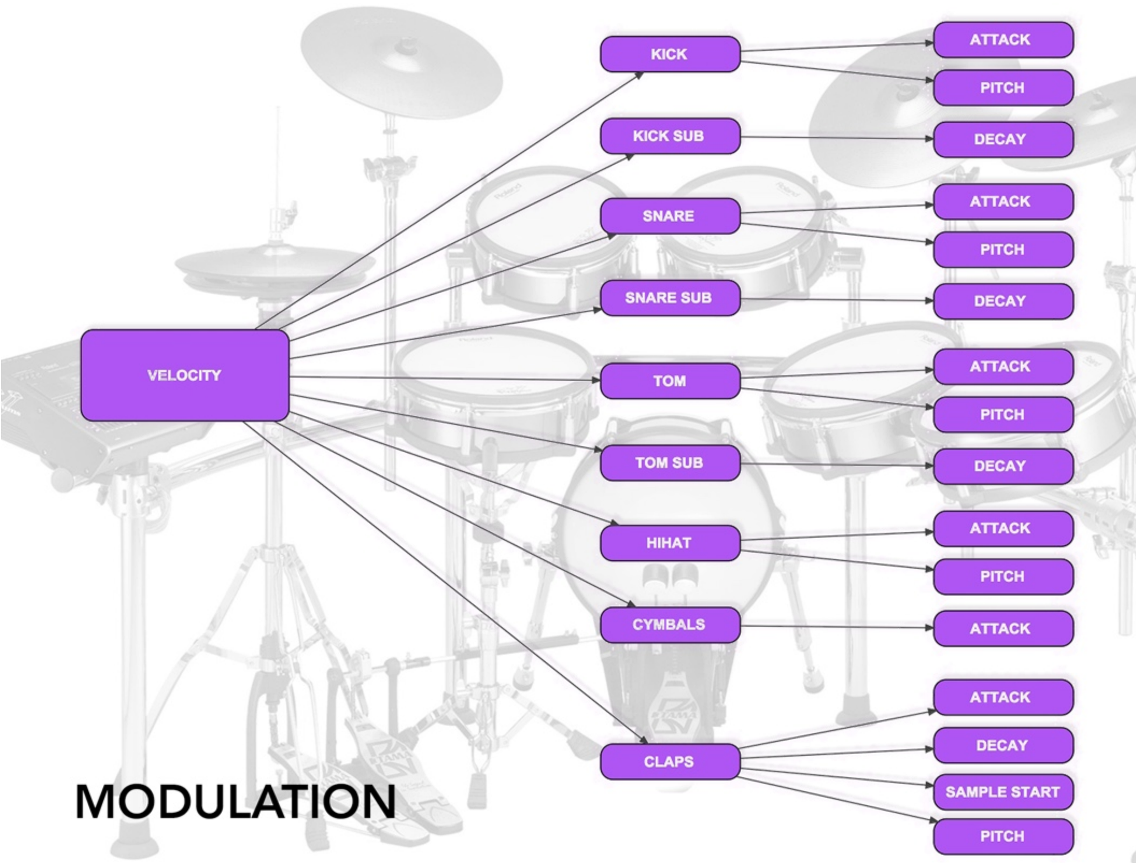


Figure 14: MORALIS, C. (2016) 'Modulation Parameters' [photograph] (Designed with VUE and Photoshop software)

By using modulation based on the drummer's dynamics, the drum samples can be further manipulated, enriching the expressivity of the drummer's performance. See Fig. 15.

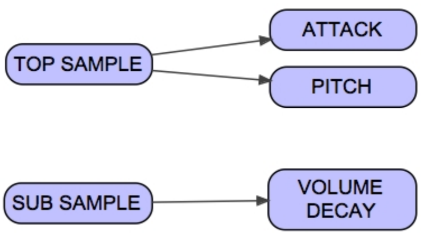


Figure 15: MORALIS, C. (2016) 'Drum Modulation' [photograph] (Designed with VUE software)

The modulated parameters are the volume, the pitch, and the attack of the sample. At the lowest velocity, the volume changes 28% of its initial level; the pitch changes four semitones (minus), the attack affects the first five milliseconds of the sample. Since the pitch affects a sound sample without any note information, it also changes the timbre, giving a different sound by shifting down its frequencies. Another critical parameter is the number of voices used. For the kick sample, it has been set to one to prevent multiple kick samples overlapping each other and creating additional constructive gain. However, the dynamic envelope curve and the way it affects the modulated parameter is based on the way the drummer performs and needs to be programmed according to his performance. Using the modulation in such a way, regarding both samples, the overall kick sample softens its sound and loses bass, emulating an acoustic kit's natural response. The same procedure has been followed for the snare and toms. However, the number of voices allowed to play together has been set to two to allow the quantizer to create snare and tom rolls.

Regarding cymbals, a slightly different approach has been used. The foot pedal contains two samples that play together with different velocity curves in order to enrich the timbre of this sample. See Fig. 16.

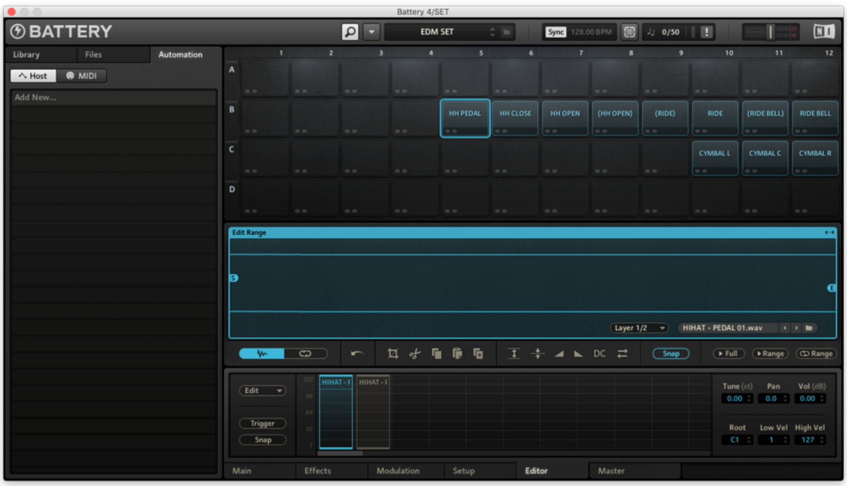


Figure 16: MORALIS, C. (2016) 'Foot Pedal Setup' [Screen Shot]

The closed hi-hat samples are distributed equally over the 127 velocities while another sample is triggered at the same time following its own velocity curve. This enriches the timbre of the hi-hat while the extra hi-hat acts similarly to parallel compression, meaning the attack and presence of the hi-hat sample are maintained over the 127 velocities. See Fig. 17.

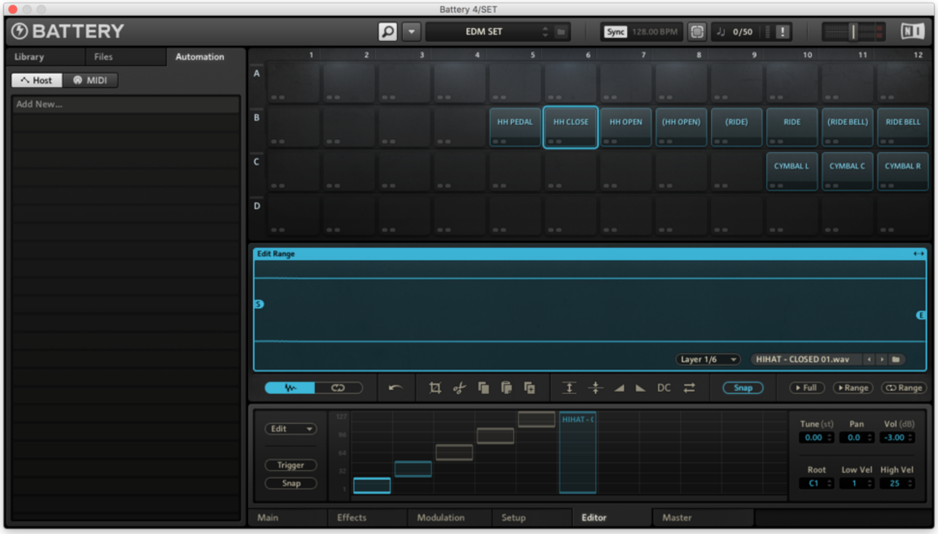


Figure 17: MORALIS, C. (2016) 'Closed Hi-Hat Setup' [Screen Shot]

The open hi-hat is fed by two different samples that play in round robin order. The open hi-hat's sound has five different samples, to emulate the drummer's hits from soft to hard hits in the 127-velocity range. Furthermore, an extra sample, with fixed timbre and pronounced attack but with varied volume, has been set up to maintain consistency in the hi-hat's presence. By setting up the open hi-hat in this way, the timbral variety has been enriched while their presence over the 127 velocities is maintained accordingly. Regarding the Ride sample, another layer with a more pronounced metallic sound has been used at the highest 32 velocities. This amplifies the metal timbre of the cymbal when it is hit hard, emulating the natural response of the metal. The same approach has been followed for the Ride Bell sample.

Regarding the crash cymbals, the five produced layers are distributed equally over the 127 velocities. The number of voices used on all cymbals is 25 to allow the drummer to perform rolls and swells. See Fig. 18.

Figure 18: MORALIS, C. (2016) 'Cymbals Setup' [Screen Shot]

The modulated parameters on all cymbals are the attack and the decay emulating the soft- and the hard-hits timbres as well as the different durations of the cymbals caused by the performer's dynamics.

- **SIDECHAIN COMPRESSION**

In EDM, sidechain compression, applied to the snare or the cymbals and usually triggered by the kick, is another common technique that improves the mixing process and the balance of the different elements of the rhythmic section. This technique can also produce a musical context when it is applied to effects, such as the reverb, or to instruments. However, in this instance, it has been used to minimize the constructive gain.

The creation of two different sounds, top and sub, allows a different sidechaining process on the same sample. For example, as shown in Fig. 19, when the kick and the snare are hit at the same time, the kick compresses the snare's sub-sample while at the same time the snare compresses the kick's top sample.

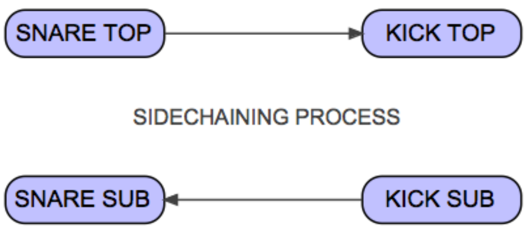


Figure 19: MORALIS, C. (2016) 'Sidechaining a Sample' [photograph] (Designed with VUE software)

Fig. 20 shows how the samples are mixed together in a 'four on the floor' groove pattern:

Figure 20: MORALIS, C. (2016) '4 on the floor SC' [photograph] (Designed with VUE software)

Furthermore, using this technique, the kit's samples can be prioritized according to the producer's taste or the song's arrangement. In the diagram above, the snare is prominent on the second and fourth hit, while the bass coming from the kick's sub-sample is not affected.

Fig. 21 shows the sidechaining process applied to the drums' samples:

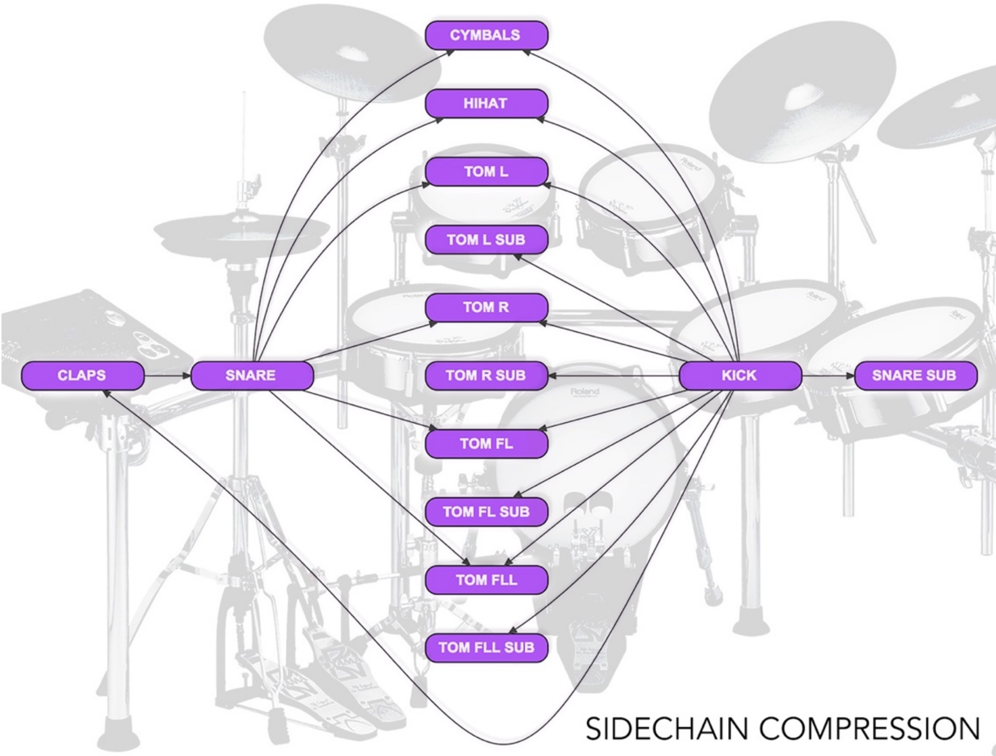
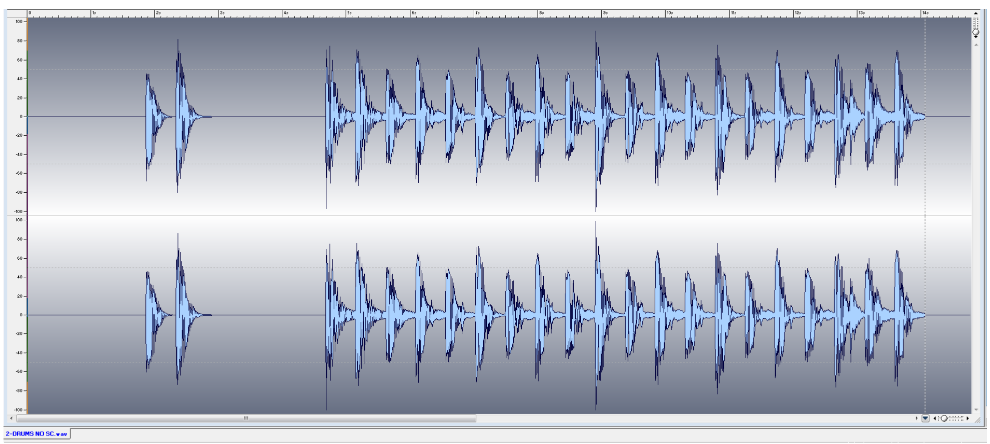


Figure 21: *MORALIS, C. (2016) 'SC Comp – Drums' [photograph] (Designed with VUE and Photoshop software)*

The sidechaining approach is very subtle and serves the mixing process as another level of controlling the overall loudness rather than having a musical approach. For the sidechaining process, a volume envelope shaper has been used: the 'Kickstart' by Sonic Academy. This envelope shaper allows the producer to shape in precise detail the volume of the sound according to the mixing procedure that he wants to follow. For a more detailed explanation of the sidechain volume curves, see Appendix 5.

Fig. 22 shows a waveform comparison of the drums playing without and with the sidechaining process.

NO SC:



SC:

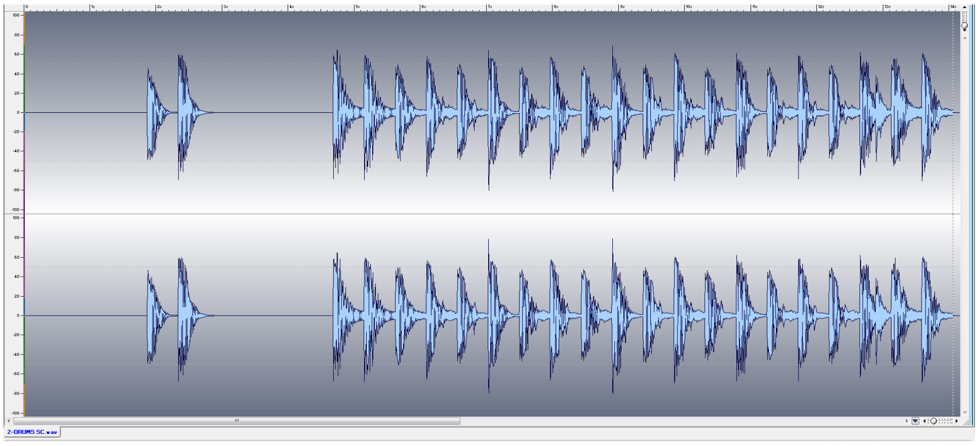


Figure 22: MORALIS, C. (2016) 'SC Drum Comparison' [Screen Shot]

As can be seen above, it is clear that the peak level is controlled much better without any dramatic audible differences between no sidechaining and sidechaining. Below are the two audio examples:

- **DRUMS – NO SC:** Audio Example 12
- **DRUMS – SC:** Audio Example 13

Fig. 23 shows the routing of the audio channels to the main stem group channel:

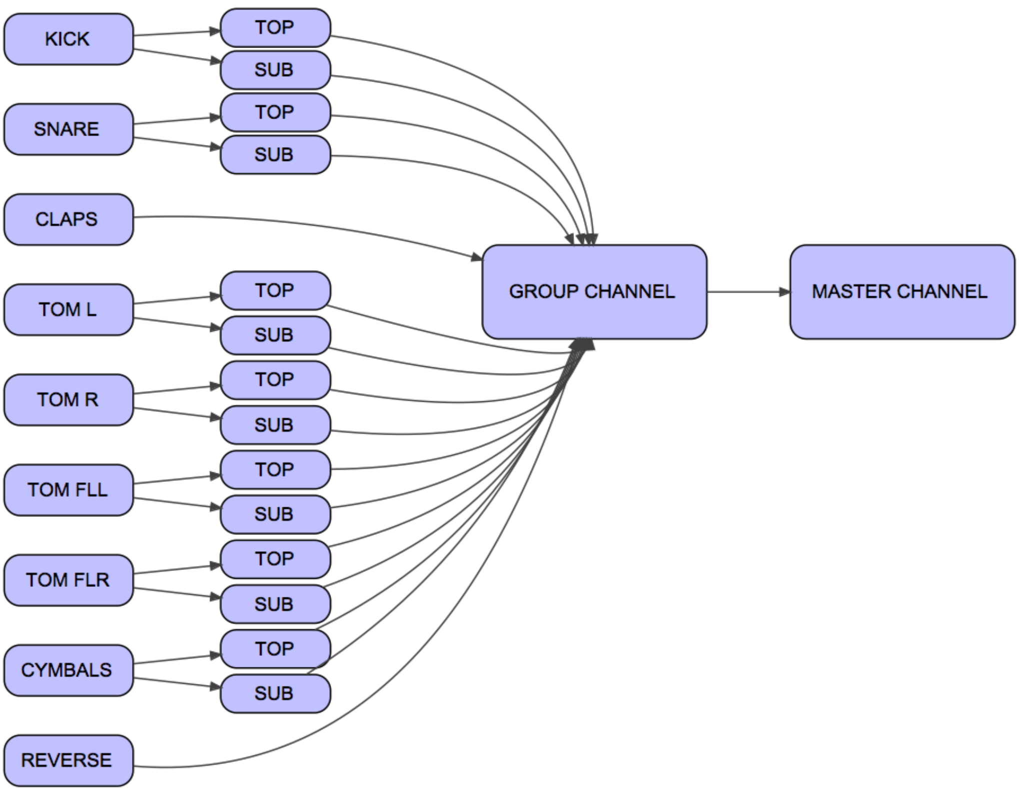


Figure 23: MORALIS, C. (2016) 'Drum Routing' [photograph] (Designed with VUE software)

- **AUTOMATION**

Dynamics

The range of the modulated parameters of the drum samples is controlled as an automated procedure according to the song's arrangement. The reason for partially controlling the expressivity of the performance is to help the drummer maintain the sound needed on the specific parts of the arrangement while allowing for varying his sound in the specified range. The midi effect 'Dynamics', by Ableton Live, controls the velocity curve and range. The drive parameter pushes the midi velocity either on the upper or lower values while 'Out High' defines the highest velocity. The compand acts either as a compressor or an expander according to its position. This midi effect is applied to the incoming midi in order to control the overall performance and not only a specific sample, maintaining in this way a more natural performance.

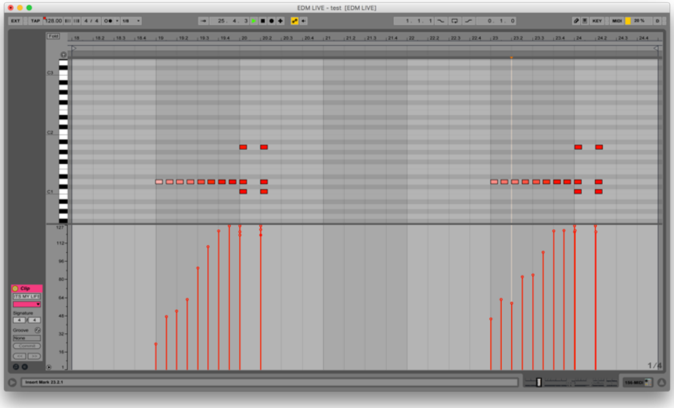
Figure 24: MORALIS, C. (2016) 'Midi Dynamics' [Screen Shot]

Fig. 25 shows an example of this automated midi effect parameter as it is applied to the specific song:

Figure 25: MORALIS, C. (2016) 'Midi Dynamics Example 1' [Screen Shot]

Fig. 26 shows an example of how the midi notes are affected by this midi effect:

Before:



After:

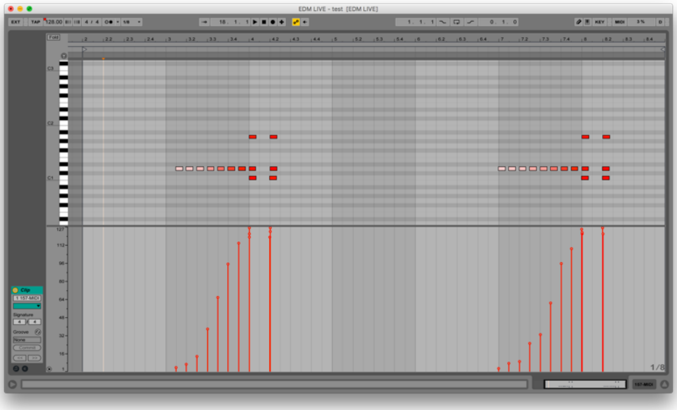


Figure 26: MORALIS, C. (2016) 'Midi Dynamics Example 2' [Screen Shot]

Pitch

Since the approach, as previously explained, is the kick pitch to follow the song's chords, a pitch selector has been used and automated as shown in Figs. 27-29:

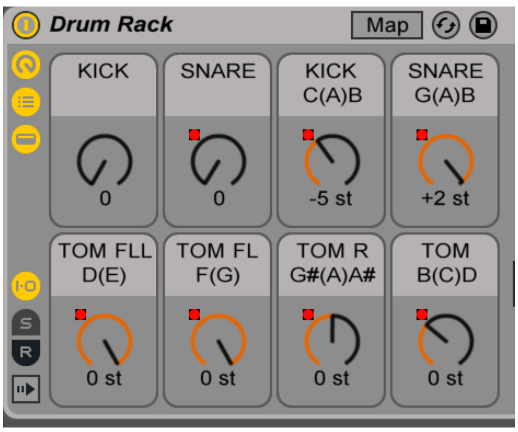


Figure 28: MORALIS, C. (2016) 'Pitch Selector' [Screen Shot]

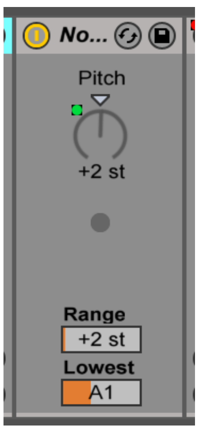


Figure 27: MORALIS, C. (2016) 'Snare Pitch Shifter' [Screen Shot]

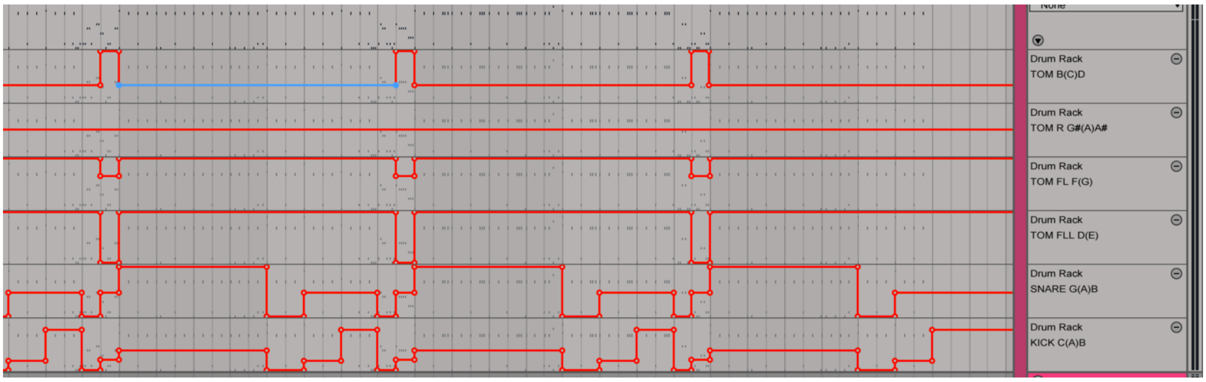


Figure 29: MORALIS, C. (2016) 'Automated Pitch Selection' [Screen Shot]

Apart from the small pitch variances caused by the drummer's performance, the pitch shifting technique is a trademark of the EDM genre. To create this pitch shifting effect, as a musical approach on the drum samples, only the subpart of the sample has been selected, while the top sample has been left unaffected. The reason for affecting only the subsample is that this part defines the pitch of the sound while the top layer helps the sample to cut through the mix.

This parameter is controlled by the 'Tune' knob, which is the sampler's internal pitch shifter, as shown in Fig. 30:

Figure 30: MORALIS, C. (2016) 'Battery – Pitch Shifter' [Screen Shot]

5.2 Hocketing Layering

According to Bhatara, Tirovolas, Duan, Levy, and Levitin (2011), *'average listeners are able to detect subtle variations in the expressive performance of piano pieces. Musicians demonstrate a greater sensitivity to these performance variations than non-musicians ... showed that listeners are attuned to such subtle cues as changes in timing and amplitude … both musicians and non-musicians can detect the difference between levels of expressivity when the two dimensions of timing and amplitude are decoupled and manipulated separately.'* In electronic music, the repeated phrases and sounds make listeners focus on other elements in the mix. This could help to amplify the excitement and the engagement with the song. As Zagorski-Thomas (2014, p.53) suggests, *'the ability to listen to the same performance many times allows the attention to focus on the minutiae of timbre, pitch and phrasing, and these lie at the heart of this performance and timbre-led aesthetic.'* When there is much repetition in what are considered the 'traditional' areas that stimulate interest in listeners – melody and harmony – then, because our brains are geared up to find difference, change and variation to be interesting, we notice changes in other parameters more. Of course, it is also true that listeners become more expert over time at noticing the particular small details that relate to their preferred style of music.

Apart from the production techniques, in order to improve further the mixing process, it is necessary to focus also on the arrangement of the song. 'Hocketing' is a textural layering technique used to add dynamics, interest and pace to an arrangement. According to Britannica.com (2016) *'Hocket, also spelled Hoquet, Hoquetus, Hoket, Hocquet or Ochetus, in medieval polyphonic (multipart) music, the device of alternating between parts, single notes, or groups of notes. The result is a more or less continuous flow with one voice resting while the other voice sounds.'* This technique helps to speed up the perceived rhythm tempo by alternating between multiple bass sounds. The main reason for using this technique on the bass synths is to avoid using many short and fast notes in the lower frequencies that may cause a loss of the perceived power of the bass.

Going more in-depth in the culture of popular electronic music, the familiar synths used in EDM are a mixture of different waveforms and effects. In this project, the 'Omnisphere' synthesizer by Spectrasonics has been mainly used for the creation of the sounds. Apart from the sound designing abilities, this synth permits the user to assign the sounds according to his/her convenience on the keyboard while setting different audio routings.

As previously explained, 'Hocketing' is the main approach for producing and later arranging the keyboard's sounds. To maintain a low end with high energy while being able to give a musicality to this sound, the bass instrument is made from two different instruments. Following, in a sense, the same production process as that of the drums, character top sample and pitch subsample. These two different instruments acting as one allow the final mixing process to reach loud volume levels without distorting or creating phasing issues within the song. See Fig.31.

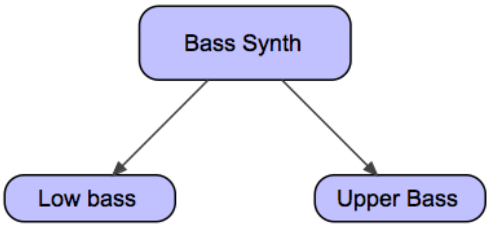


Figure 31: MORALIS, C. (2016) 'Bass Synth' [photograph] (Designed with VUE software)

For the creation of the 'low bass' instrument that will play mainly the frequencies below 300Hz, different waveforms have been used in order to enrich its sound. Since 'Omnisphere' provides the producer with a choice of waveforms, from just a simple saw or sine to the more complex, the need for further sound effects is minimized and sometimes eliminated. The combination of multiple waveforms will help to produce sounds that will sound rich and will blend properly into the mix. In the example of 'It's my Life', for the creation of the 'sub' or 'low' bass, two sound layers have been used with different waveforms. The first layer is a 'June Octo 1' waveform, which is a complex saw waveform, and layer two is a 'Juno 60 Sub Pulse' waveform, a square complex waveform. See Fig. 32.

Layer 1 Layer 2

Figure 32: MORALIS, C. (2016) 'Sub Bass Waveforms' – Score' [Screen Shot]

Micro tuning variance is another technique that is used to create depth and space between the synthesized sounds. This can be achieved either by slightly tuning up or down a sound or using low-frequency oscillators (LFOs) to control the pitch shifting of the instrument's frequencies. However, other effects such as flanger, chorus and other modulated effects can also help in creating space and depth in the mix. Furthermore, the first waveform has been enriched with a unison effect provided by the synthesizer. See Fig. 33.

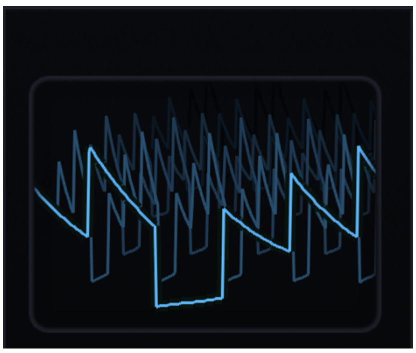


Figure 33: MORALIS, C. (2016) 'Sub Bass – Layer 1 Unison – Score' [Screen Shot]

Below are demonstrated the sounds of the ‘low bass’ instrument:

- **LAYER 1:** Audio Example 14
- **LAYER 2:** Audio Example 15
- **LOW BASS:** Audio Example 16

As shown in Fig. 34, to give the signature EDM ‘pumping’ effect to this instrument, sidechaining compression set to one quarter note and synced to the bpm has been applied. Furthermore, a limiter has been applied to control and maintain the low bass volume.

Figure 34: MORALIS, C. (2016) ‘Low Bass SC’ [Screen Shot]

- **LOW BASS (SC+LIMITER):** Audio Example 17

For the creation of the second (upper) bass that will give extra movement to the sound, a ‘reedy’ waveform has been used for the first layer and a sine waveform for the second layer. See Fig 35.

Layer 1 Layer 2

Figure 35: MORALIS, C. (2016) ‘Upper Bass Waveforms’ – Score’ [Screen Shot]

This upper bass acts as the main timbre of the bass, and in order to provide rhythmic content to its performance, a synchronized LFO has been used. This LFO acts like an arpeggiator with a sixteenth-note rate. See Fig. 36.

Figure 36: MORALIS, C. (2016) ‘Upper Bass Sync LFO’ – Score’ [Screen Shot]

Furthermore, for more complex melodies, an arpeggiator has been used on the upper bass. As shown in Fig. 37 for example, the bass in the song 'Beat It' has an arpeggiator attached with the following rhythmic pattern shaped by these velocities.

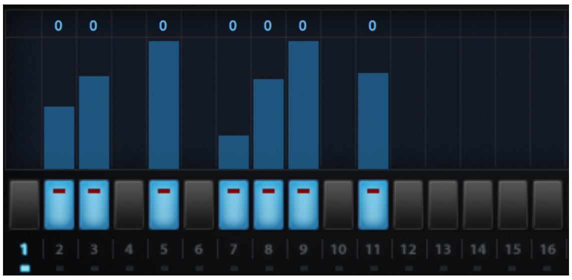


Figure 37: MORALIS, C. (2016) 'Bass Arpeggiation Example' – Score' [Screen Shot]

- **LAYER 1:** Audio Example 18
- **LAYER 2:** Audio Example 19
- **UPPER BASS:** Audio Example 20

In the case of the 'upper bass', two consecutive sidechain compressors – or better volume envelope shapers, as shown in Fig. 38 – are affecting its pumping feeling while a limiter controls the instrument's volume.

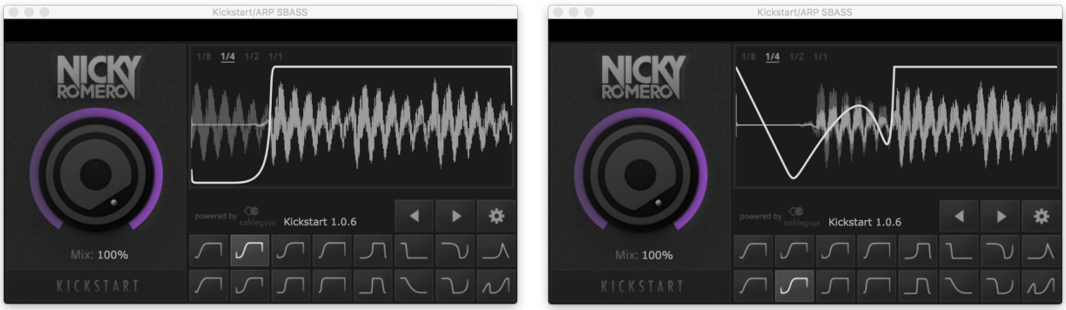
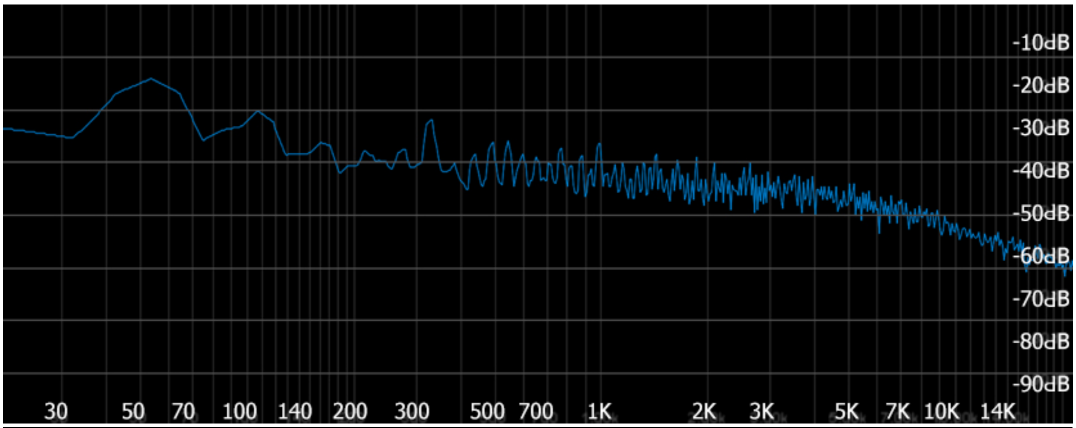


Figure 38: MORALIS, C. (2016) 'Upper Bass SC' [Screen Shot]

- **UPPER BASS (SC+LIMITER):** Audio Example 21

The upper bass is covering the whole frequency spectrum while the lower bass is focused more on the mid and low frequencies. However, both instruments constitute one perceived bass instrument. The frequency range in which each bass instrument occurs is shown in Fig. 39.

Upper Bass:



Low Bass:

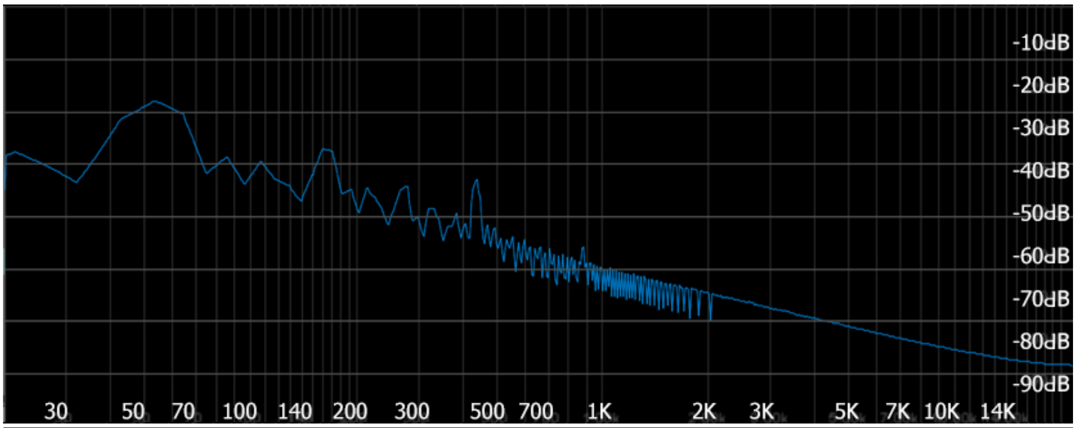


Figure 39: MORALIS, C. (2016) 'Low Bass Waveforms' – Score' [Screen Shot]

Because of the latency added by the processing time, the pitch tracking equalization, phase interaction mixing process and adaptive tonal contour linearization cannot be applied in live performance. Therefore, the approach to improve the mixing process will be based upon the sound designing of the waveforms by adjusting the symmetry, the synchronization and, especially, their shape. An example of this is shown In Fig. 40.

Figure 40: MORALIS, C. (2016) 'Waveform shaping parameters' [Screen Shot]

In Fig. 41 is shown the mixing routing process of all the synthesized or sampled instruments that are being used by the keyboard player.

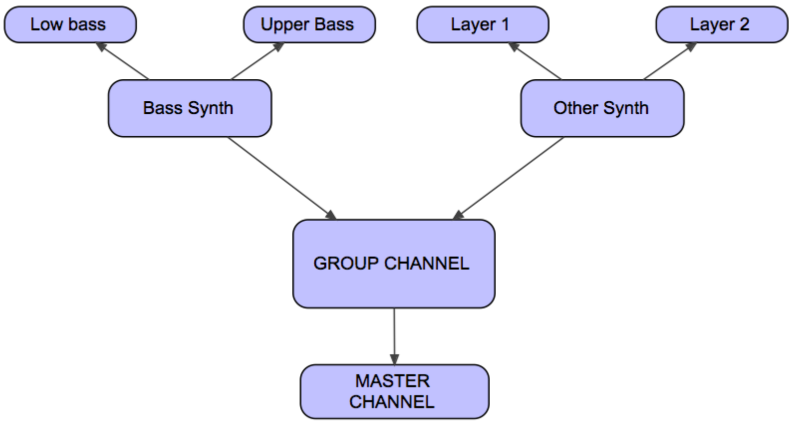


Figure 41: MORALIS, C. (2016) 'Synth Mixing Process' [photograph] (Designed with VUE software)

Regarding the keyboard setup, it is necessary to mention the factors that may affect the Performable Recordings model in the case of keyboards. The number of different keyboards that can be used for the performance through this production model depends on the ability of the performer. However, to minimize the performance errors, the performer is separated from the sound selection process. All sounds used in every song are loaded and enabled automatically, leaving the performer to focus on the musical content and not on the technical. With software packages such as Mainstage, this is standard practice across all musical genres that use multiple keyboard sounds in a single set. Also, most of the sound effects, such as filters, delays and reverbs, are also automated.

Having that said, another important element that affects the keyboard performance is the sound's placement on the keyboard. Since in this project there is no bass player, the keyboard player also acts as the bass player, triggering with his left hand the mostly arpeggiated bass synthesizer.

Fig. 42 shows a typical live setup combining different sounds on the same keyboard, allowing the performer to have immediate access to all the sounds used in the song.

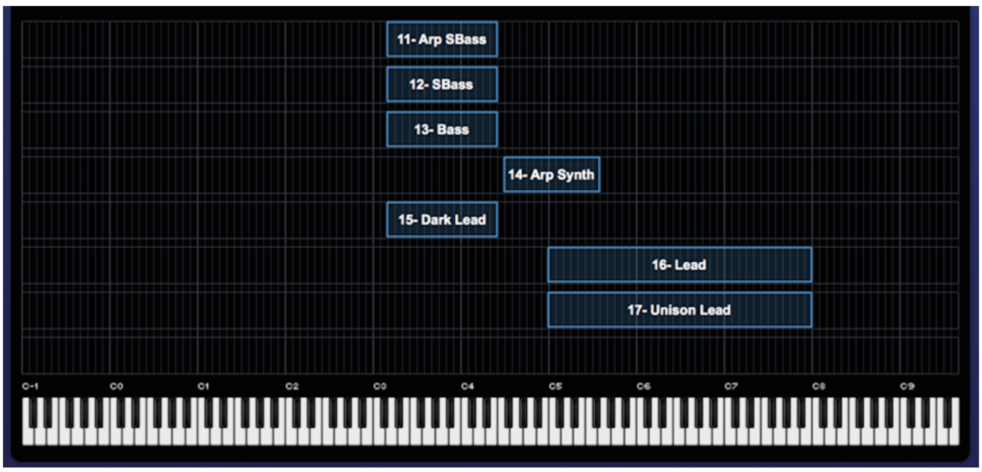


Figure 42: MORALIS, C. (2016) 'Stack Mode Keyboard Setup' [Screen Shot]

5.3 Real-time layering

A typical instrument in EDM is the electric guitar. Many artists of electronic music, such as David Guetta, in the song 'Titanium' featuring Sia, use electric guitar riffs. However, this instrument is often sampled or looped. In David Guetta's song, the guitar plays a four-bar looped riff. The repetition of motifs, riffs, melodies, lyrics and other, in EDM, is a common approach as it is essential for the listener's entrainment. In addition, a common technique is the layering of different instruments playing together to enrich the timbre of the part. It is also common to layer a guitar track with a synthesized sound. This makes the electric guitar sound closer to the EDM aesthetic.

To achieve this in real time, the wireless midi guitar controller from Fishman has been used. This is a device that captures the guitar sound with a hexaphonic pickup, translating the separate sound of each string into midi notes and sending it wirelessly to the laptop where a synthesizer plays them accordingly.

Following the examples shown earlier from the guitar part of the song 'So True' are two examples: one only with the synthesized sound and one with both the guitar and the synthesized sound.

- **So True (Synthesized Sound Only):** Audio Example 22
- **So True (Synthesized and Guitar Sound Together):** Audio Example 23

The midi notes received from the guitar are fed into the Virus synthesizer from TC electronic using the Powercore X8 hardware. This allows the producer to virtually design the desired synth sound on the screen like every other software but with the difference that this synthesizer runs from external hardware without consuming any CPU power.

Fig. 43 shows the interface of the Virus synthesizer:

Figure 43: MORALIS, C. (2017) 'Virus Powercore for Guitar' [Screen Shot]

To further control and manipulate the synthesized sound, a filter effect has been applied and automated along with a volume shaping tool. As shown in Fig. 44, these two effects both help the mixing process and serve the musicality of the performance.

Figure 44: MORALIS, C. (2017) 'Auto Filter & Sidechain' [Screen Shot]

- **So True (Without Effects on the Synthesized Sound):** Audio Example 24
- **So True (With Effects on the Synthesized Sound):** Audio Example 25

Since this synthesized sound plays the role of a layered sound, it is not mixed separately from the overall guitar track. As shown in Fig. 45, the synthesized sound is fed into the final group channel and processed with the guitar signal.

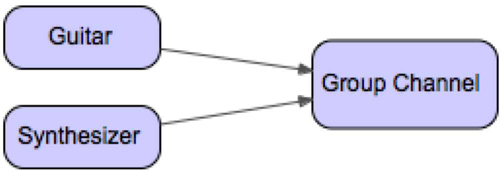


Figure 45: MORALIS, C. (2017) 'Guitar and Synth signal path' [photograph] (Designed with VUE software)

Since the tracking of the guitar is happening in real time, as with other real-time audio-to-midi devices, there might be some error in the tracked notes. To prevent wrong or extra unnecessary midi notes going through, a midi-scale effect has been applied to the midi channel. In the examples below are shown the songs 'Enjoy the Silence' and 'So True'. In the first example, the scale has almost all its notes, while in the second example, since the guitar is not playing any other notes, only two notes from the scale have been used.

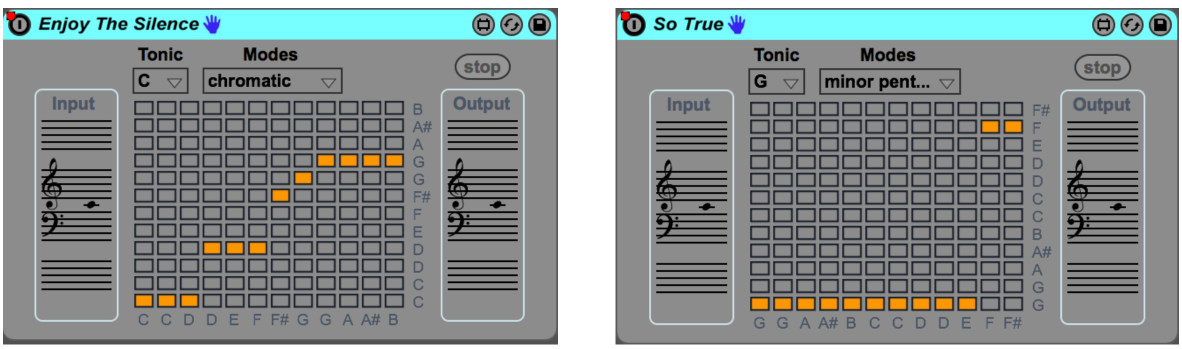


Figure 46: MORALIS, C. (2017) Example of the mode effect on the guitar midi channel' [Screen Shot]

By filtering all the unnecessary midi information, we preserve clean audio-to-midi process while the performer can play the guitar without paying attention to every detail of the audio-to-midi process and instead focusing on his performance.

5.4 Real-time pitch quantization

All the effects included in the TC-Helicon (see Appendix 6) are based on contemporary studio production techniques and thus allow the combination of studio and live aesthetics as it is required in this project.

However, apart from the creative effects, this hardware offers real-time equalization as the manufacturer names it: '*adaptive – automatic equalization*'. This technology tracks the frequency content of the input signal in real time and adapts its frequency bands to offer the best sounding results. According to Haykin (1996)*, 'An adaptive equalizer is an equalizer that automatically adapts to time-varying properties of the communication channel'.*

- **NATURAL PLAY**

The user can specify the key of each song but also can be more specific by sending midi notes to define the chords that the backing vocals should follow. In this case, a midi track has been created in the Ableton with the chords of the chorus and sent to TC Helicon. The hardware follows the KEY that it has been set along with the chords provided through the 'Natural Play' option.

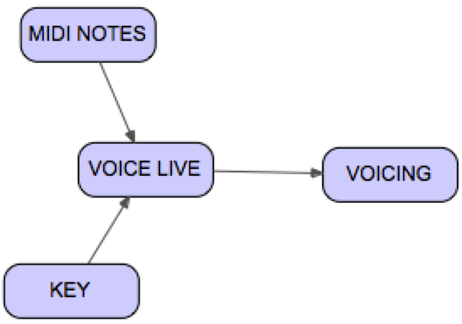


Figure 47: MORALIS, C. (2017) 'Natural Play' [photograph] (Designed with VUE software)

Fig. 48 shows the midi notes written in Ableton for the harmonizer:

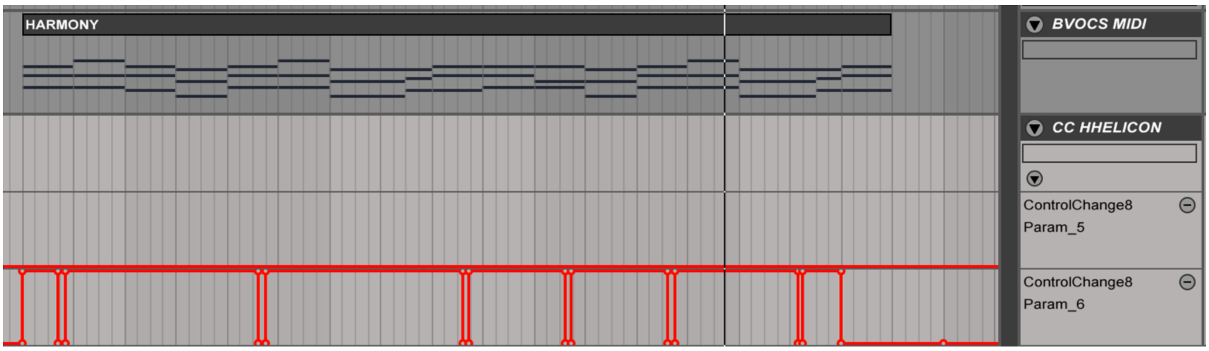


Figure 48: MORALIS, C. (2017) 'Midi Notes for Harmonizer' [Screen Shot]

- **DOUBLER**

One of the most common effects used in this project to enrich the timbre of the voice is the 'doubler'. This effect recreates the unison voice that occurs when two or more people play or sing at the same pitch. The typical settings for this effect are shown in the following table:

- **ITS MY LIFE**

1 Voice Loose	Humanize 35%	Level -40db	
Human Style Random	Portamento 80	Smoothing 0%	
V1 Level 0	V2 Level -36db	V3 Level -36db	V4 Level -36db
V1 Pan: C	V2 Pan: C	V3 Pan: C	V4 Pan: C
V1 Gender 0	V2 Gender 0	V3 Gender 0	V4 Gender 0
V1 Voicing Unison	V2 Voicing Off	V3 Voicing Off	V4 Voicing Off
V1 Porta: 80	V2 Porta: 0	V3 Porta: 0	V4 Porta: 0
V1 Smooth 0%	V2 Smooth 100%	V3 Smooth 100%	V4 Smooth 100%
Lead Level 0db	Global Off		

Figure 49: MORALIS, C. (2017) 'It's My Life 1 – TC Helicon Settings' [table]

These settings mean that a unison voice is being produced at -40db along with the lead that randomly varies at 35% from the original, with some portamento on the produced voice.

Below is an audio example of the above settings:

- **It's My Life – Doubler Effect:** Audio Example 26

Furthermore, this effect can create up to four different voices for a fatter sound or alter the pitch of the other voices to create harmonies. For example, in the song 'Enjoy the Silence', the doubler is being used as an octaver:

- **ENJOY THE SILENCE**

Oct Up Double	Humanize 20%	Level 0db	
Human Style Random	Portamento 20	Smoothing 90%	
V1 Level 0	V2 Level -36db	V3 Level -36db	V4 Level -36db
V1 Pan: C	V2 Pan: C	V3 Pan: C	V4 Pan: C
V1 Gender 0	V2 Gender 0	V3 Gender 0	V4 Gender 0
V1 Voicing Oct Up	V2 Voicing Off	V3 Voicing Off	V4 Voicing Off
V1 Porta: 20	V2 Porta: 0	V3 Porta: 0	V4 Porta: 0
V1 Smooth 90%	V2 Smooth 100%	V3 Smooth 100%	V4 Smooth 100%
Lead Level -1db	Global Off		

Figure 50: MORALIS, C. (2017) 'Enjoy The Silence'– TC Helicon Settings' [table]

The above settings mean that a unison voice is being produced at the same level as the original voice while the lead voice is being reduced by 1db. Furthermore, the lead voice randomly varies at 20% from the original, with a little bit of portamento and much smoothing in the transition between the different notes of the produced voice.

Below is an audio example of the above settings:

- **Enjoy the Silence – Doubler Effect as Octaver:** Audio Example 27

However, this effect has also been used as unison voices creating an exciting chorus and flanged effect:

- **CHANGED THE WAY YOU KISSED ME**

2 voices Wide	Humanize 50%	Level 3db	
Human Style Random	Portamento 25	Smoothing 100%	
V1 Level 0	V2 Level 0db	V3 Level -36db	V4 Level -36db
V1 Pan: L25	V2 Pan: R25	V3 Pan: C	V4 Pan: C
V1 Gender 0	V2 Gender 0	V3 Gender 0	V4 Gender 0
V1 Voicing Unison	V2 Voicing Unison	V3 Voicing Off	V4 Voicing Off
V1 Porta: 25	V2 Porta: 25	V3 Porta: 25	V4 Porta: 25
V1 Smooth 100%	V2 Smooth 100%	V3 Smooth 100%	V4 Smooth 100%
Lead Level -6db	Global Off		

Figure 51 MORALIS, C. (2017) 'Changed The Way You Kissed Me'– TC Helicon Settings' [table]

Below is an audio example of the above settings:

- **Changed the Way You Kissed Me – Doubler as Unison:** Audio Example 28

- **HARMONY**

The harmonizer creates different voicings from the original voice. The typical settings used for this effect are shown in the following table:

- **ITS MY LIFE**

HIGH	Lead Level 0db	Level -4db	
Human Style OFF	Humanize 0%	VIB Style OFF	Vibrato 0%
		Natplay SRC Midi	Tuning Just
MODE V1 Nat Play	MODE V2 OFF	MODE V3 OFF	MODE V4 OFF
V1 Voicing HIGH			
V1 Level 0	V2 Level 0db	V3 Level 0db	V4 Level 0db

V1 Gender 0	V2 Gender 0	V3 Gender 0	V4 Gender 0
V1 Pan: C	V2 Pan: C	V3 Pan: C	V4 Pan: C
V1 Porta: 0	V2 Porta: 0	V3 Porta: 0	V4 Porta: 0
V1 Smooth 90%	V2 Smooth 90%	V3 Smooth 90%	V4 Smooth 90%
HOLD Release 100ms			
Low Gain 0db	Low Freq 1140Hz	High Gain 0db	High Freq 1140Hz
Mid gain 0db	Mid Freq 1140Hz	MID BW 1.00	
Global OFF			

Figure 52: MORALIS, C. (2017) 'It's My Life' 2 – TC Helicon Settings' [table]

The above settings mean that a second voice is being produced following the midi notes, written in a midi track in Ableton (natural play) for creating the chords. There is no humanized level, or any equalization applied in this case. The voice is being smoothed by 90% and is set -4db from the original.

Below is an audio example of these settings:

- **It's My Life – Harmonizer:** Audio Example 29

The harmonizer can emulate choirs as demonstrated in the following example, which adds two extra voices on top of the original:

- **So True – Harmonizer (with 2 voices):** Audio Example 30

Below are shown the settings used:

- **SO TRUE**

HIGH & Higher	Lead Level 0db	Level -4db	
Human Style OFF	Humanize 0%	VIB Style OFF	Vibrato 0%
		Natplay SRC Midi	Tuning Just
MODE V1 Nat Play	MODE V2 Nat Play	MODE V3 OFF	MODE V4 OFF
V1 Voicing HIGH	V2 Voicing Higher		
V1 Level 0	V2 Level 0db	V3 Level 0db	V4 Level 0db
V1 Gender 0	V2 Gender 0	V3 Gender 0	V4 Gender 0
V1 Pan: L60	V2 Pan: R60	V3 Pan: C	V4 Pan: C
V1 Porta: 0	V2 Porta: 0	V3 Porta: 0	V4 Porta: 0
V1 Smooth 90%	V2 Smooth 90%	V3 Smooth 90%	V4 Smooth 90%
HOLD Release 100ms			
Low Gain 0db	Low Freq 1140Hz	High Gain 0db	High Freq 1140Hz
Mid gain 0db	Mid Freq 1140Hz	MID BW 1.00	
Global OFF			

Figure 53: MORALIS, C. (2017) 'So True'– TC Helicon Settings' [table]

The above settings mean that two additional voices are being produced following the midi notes (natural play) for creating the chords. There is no humanized level, or any equalization applied in this case. The voices are being smoothed by 90% and are set -4db from the original.

- **<u>HARD TUNE</u>**

One of the most important processes happening in the TC Helicon hardware is the auto-tuning. Like many other software auto tuners, this effect can be set to operate subtly, but it can be set to extremes for a more musical effect.

The typical preset in this project is as shown in the below settings:

Correct Natural	Gender 0	Shift 0
Key Source: Manual	Rate 75%	
Amount 35%	Windows +-250c	
Manual Key: A		
Manual Scale: Minor-Nat	Note Select A	Note Enable OFF
Global Off		

Figure 54: MORALIS, C. (2017) 'Hard Tune'– TC Helicon Settings' [table]

The type of correction is set to Natural. This type has a smoother transition between the notes. The rate is set high at 75%. However, the amount that is being corrected is 35% using a window of 250 cents.

Below is an example of how this auto tuner operates:

- **Voice Track (Without Auto Tuning):** Audio Example 31

- **Voice Track (With Auto Tuning):** Audio Example 32

However, the auto tuner could be used in a more extreme way, as in the example in the song 'So True':

- **So True (With Extreme Auto Tuning):** Audio Example 33

5.5 Multiple gain and timbral treatment

According to Zagorski-Thomas (2014, p.78), *'The notion of staging refers to the treatment of sound in ways that add meaningful context for the listener to a performance or a perceived musical "event".'*

- **VOICE**

The voice is the most crucial part of popular music. It is the 'instrument' that carries not only musical information but also meaning through the words. The stories and the meaning that the singer intends to deliver to the audience are often supported by mechanical means. These means can be either a microphone that operates through a PA or a series of equipment that could amplify, alter or enhance the attributes of the person's voice.

As has already been explained in this paper, the idea of this research is to bridge the live stage with studio production. The most critical mechanical part in the mixing process of a voice is the microphone. There are different types of microphone used in a studio or a stage, from condensers to dynamic microphones and from analog to digital. For this project, an analog wireless Shure microphone with the Beta 58 capsule was used. This is a middle-range microphone, the most common type of microphone used in a live situation. However, as it has been explained already, a digital microphone could deliver a wider frequency range without any audio issues caused by the compander.

For this research, the same UAD Apollo 8 has also been used on the vocals. To achieve the desired results, the signal follows a long path. Fig. 55 shows the path of the audio signal beginning from the microphone and ending right before the final mixer:

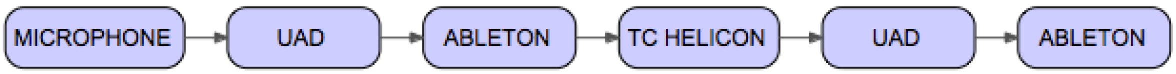


Figure 55: MORALIS, C. (2017) 'Voice signal path' [photograph] (Designed with VUE software)

The voice signal is being transmitted wirelessly to the audio interface and then is fed into Ableton. From Ableton it goes to the external voice modeling hardware, Voicelive 3 from TC Helicon and from there back to the audio interface and into Ableton. After the final processing within the Ableton, the signal is routed to the audio interface's console panel, where a final mastering limiter sums all the instruments.

When it comes to the singer's microphone, the digital wireless Line 6 XD-75 has been used. This specific microphone has been selected as it can model different types of microphone capsule. The variation between different capsules allows this project to enhance the sound of the voice between songs.

The initial input level has been set from the main microphone capsule in such a way to avoid any input distortion caused by loud voices.

The first mixing stage is to create the desired tone for the voice. This will make it easier for the TC Helicon hardware to process the signal. Again, the Unison technology has been used to enrich the sound of the microphone by adding 10db and a slight pushing in the EQ at 70Hz. A high pass filter has been applied to remove any unwanted low-frequency noise. However, to maintain a specific dynamic range that could help the gain staging process, two compressors have been used at this point. Initially, a Fairchild 670 Legacy has been applied, due to its low distortion. S shown in Fig. 56, the time setting has been set to option 1 since this is a fast attack and fast release setting.

Figure 56: MORALIS, C. (2017) 'UAD Voice input effect chain 1' [Screen Shot]

To further balance the dynamics of the microphone, another compressing stage has been applied. This time the 1176LN Limiting Amplifier also helps to balance the input signal. According to UAD (2017), '*1176 Rev E "Blackface": This model covers the early 70's / Brad Plunkett "LN" (Low Noise) era of the 1176 circuit lineage, with variations including a more linear compression response, transistor gain amplification, and a change to the program dependency.'*

A medium speed attack with a very fast release can preserve the timbre of the voice without destroying any consonants that have been compressed earlier. As shown in Fig. 54, the ratio has been set to 4:1 for a more subtle compression approach.

Figure 57: MORALIS, C. (2017) 'UAD Voice input effect chain 2' [Screen Shot]

The signal is fed into Ableton Live where initially a volume shaping tool avoids any constructive gain caused with the drum kick. The signal is being affected on every beat only by 35% to make the process less obvious, without any musical meaning, and to help the mixing process. Fig. 58 shows the volume curve.

Figure 58: MORALIS, C. (2017) 'Voice – Kickstart in input channel' [Screen Shot]

This is subtle, but it helps to better control the lower end on every quarter note where usually in EDM there is a kick playing. However, this plugin is being automated to switch off when the kick is not playing. Below are two examples, one without the volume envelope shaping tool and one with this volume shaping process.

- **Voice (Without the Volume Shaping Tool):** Audio Example 34

- **Voice (With the Volume Shaping Tool):** Audio Example 35

After the volume shaping tool, a high pass filter has been applied to adjust both the overall timbre and dynamic response. In this case, the frequency has been put to 221HZ and the Resonance at 9.3%. There is no envelope, or LFO modulation applied. This plugin has no sample latency. See Fig. 59:

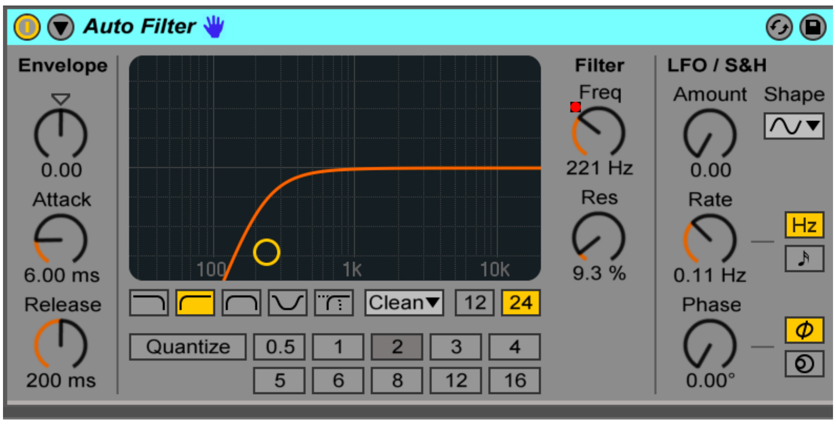


Figure 59: MORALIS, C. (2017) 'Voice Input Filter' [Screen Shot]

Two equalization plugins from Brainworx help to sculpture the timbre of the voice further by applying a more precise equalization to the signal. See Fig. 60:

Figure 60: MORALIS, C. (2017) 'Voice – Brainworx EQ' [Screen Shot]

Finally, the URS equalizer helps to further balance the timbre of voice by adding +4.2 dB in the lower range. See Fig. 61.

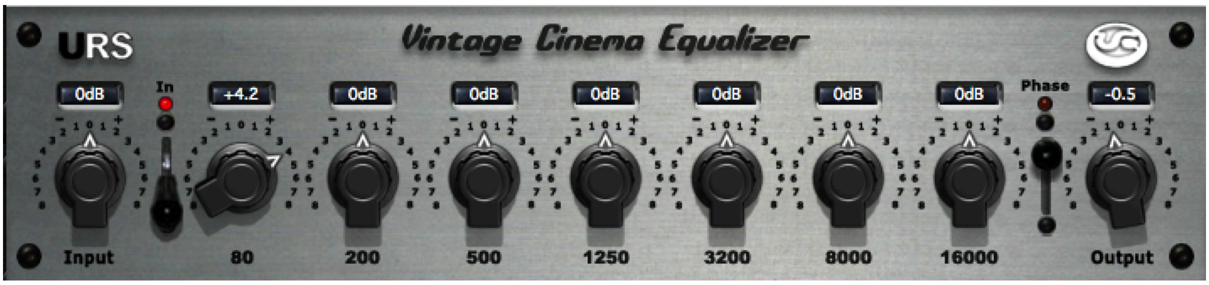


Figure 61: MORALIS, C. (2017) 'Voice – URS Vintage Cinema Eq' [Screen Shot]

At this point where the final timbre adjustments have been made, a limiter helps to avoid any excessive peaks through the processing model. However, the input gain is varied across a range of 5db to push it harder sometimes. This helps the performer to maintain a natural performance without paying attention to the input level of the mic. Fig. 62 shows three different examples.

Figure 62: MORALIS, C. (2017) 'Voice – Limiter Settings in Input Channel' [Screen Shot]

After balancing the timbre of the voice and applying some slight volume control, the signal goes to an external digital processing modelling unit. In this research project, the TC Helicon Voicelive 3 unit has been used. According to the manufacturer, www.tc-helicon.com (2017) is a unit that provides effects such as doubling, hard tune, synth, transducer, micromod, harmony, choir, rhythmic and stutter as well as delay and reverb. See Fig. 63.

Figure 63: www.tc-helicon.com (2017) 'Voicelive 3 - Effects' [Screen Shot]

At this point in the signal path, the internal effects of TC Helicon such as gating, equalization, compression, doubler, harmonizer, etc., have been used. Delay and reverb have also been used to create some space.

Fig. 64 shows the TC Helicon internal signal path.

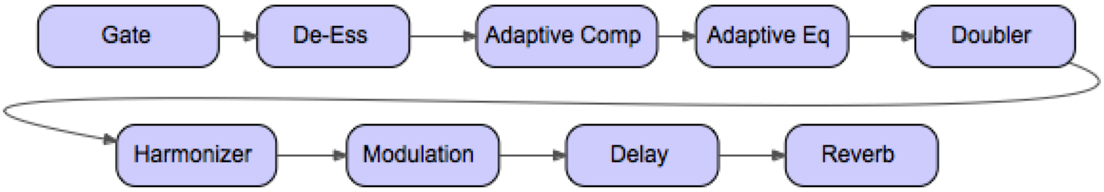


Figure 64: *MORALIS, C. (2017) 'TC Helicon signal path' [photograph] (Designed with VUE software)*

After modelling the voice signal and applying compression, modulation and other effects, the signal goes through the UAD console again. The high pass filter is again enabled along with the preamp for some extra saturation. This time only an equalizer is being applied to further balance the overall timbre of the voice along with the 1176LN. Fig. 65 shows the two effects applied and their settings.

Figure 65: *MORALIS, C. (2017) 'UAD TC Helicon input effect chain' [Screen Shot]*

Furthermore, the character effect has been applied only on certain areas to help the voice track to cut through the mix. According to the manufacturer, this plugin is based on the Adaptive Filtering (IAF) technology. It acts as a dynamic Eq, and in this instance has been used to help make the voice's transients more pronounced. Fig. 66 shows the settings applied.

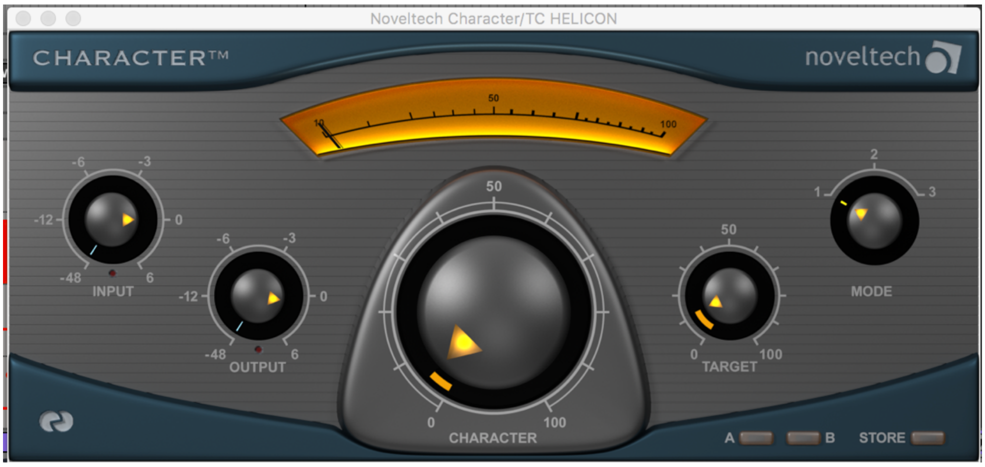


Figure 66: MORALIS, C. (2017) 'Voice – Character' [Screen Shot]

The parameters and the patches of the TC Helicon modelling hardware are controlled through automated CC messages. Fig. 67 shows the ControlChange8 MAX for Live patch that sends different CC messages to the TC Helicon.

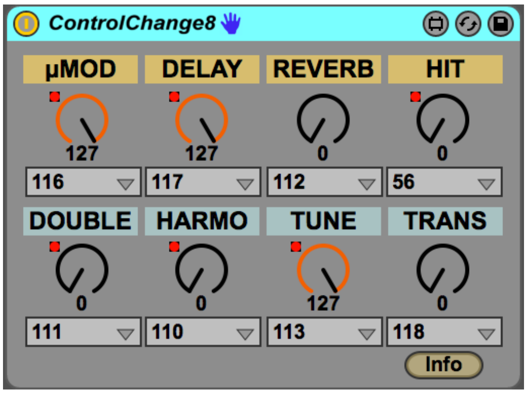


Figure 67: MORALIS, C. (2017) 'TC Helicon ControlChange8' [Screen Shot]

This MAX for Live patch controls parameters such as the tempo, modulation, wah-wah, delay and reverb through CC messages. Since it is essential for the performer to focus on his performance, aspects of these procedures that may affect his performance or the sound quality of the Performable Recordings have been automated. Fig. 68 shows some typical examples of this procedure.

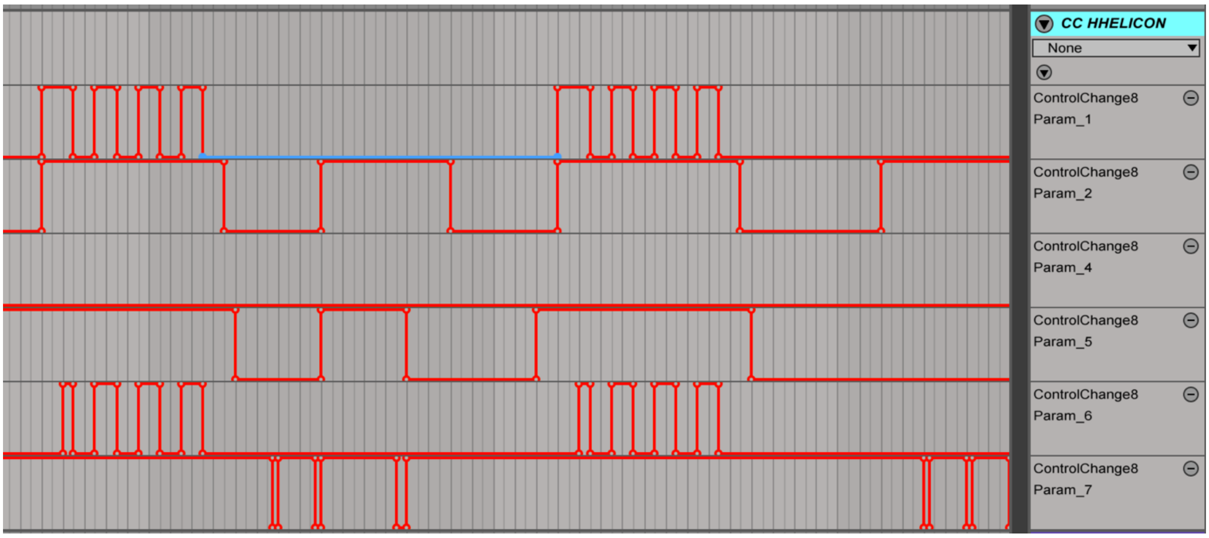


Figure 68: MORALIS, C. (2017) 'ControlChange8 automation example 3' [Screen Shot]

Furthermore, all the patches or presets change automatically with CC messages from the laptop to the TC Helicon hardware. See Fig. 69.

Figure 69: MORALIS, C. (2017) 'ControlChange8 automation example 4' [Screen Shot]

Also, the volume curve is designed to prevent any unnecessary audio going through on parts where the voice is not singing. Fig. 70 shows an example of this process.

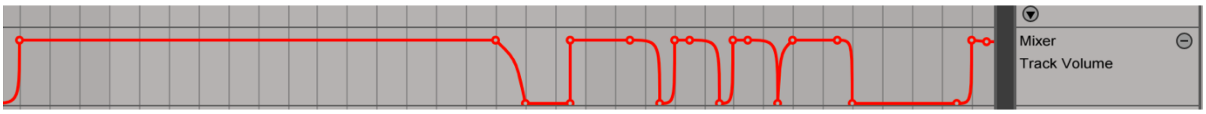


Figure 70: MORALIS, C. (2017) 'Voice – Final Volume Curve' [Screen Shot]

Furthermore, the delay is also being automated and enabled only on certain areas according to the arrangement's needs. Fig. 71 shows a typical example of this process:

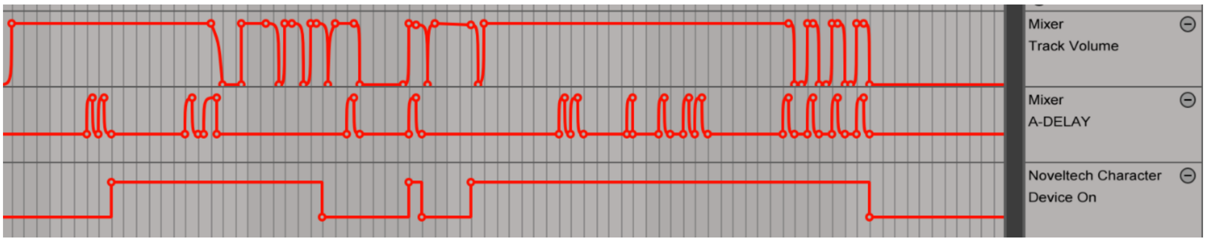


Figure 71: MORALIS, C. (2017) 'Voice – Final Processes' [Screen Shot]

- **GUITAR**

Another important factor in maintaining the timbral consistency is the audio processing before and after the guitar modelling. In this case, everything is mixed in the box using external hardware amp simulators that can be digitally controlled. However, to understand the production process better, it is necessary to explain what equipment has been used, how and why.

The UAD Apollo 8 audio interface has been selected for its low latency abilities and the Unison technology preamp modelling. According to Pro-Tools-Expert.com (2017), '*Unison is an exclusive analog/digital integration system that gives the user continuous, real-time, bi-directional control and interplay between Apollo's physical hardware and UAD software mic preamp models.*'

To achieve the desired results, the signal follows a long path. Fig. 72 shows the path of the audio signal beginning at the guitar and ending right before the final mixer:

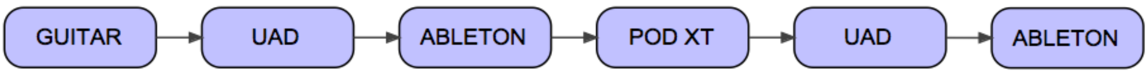


Figure 72: MORALIS, C. (2017) 'Guitar signal path' [photograph] (Designed with VUE software)

The guitar signal is transmitted digitally and wirelessly (see Appendix 7) to the audio interface, then fed into Ableton. From Ableton, it goes to the external guitar amp modeling hardware, POD XT and from there back to the audio interface and back into Ableton. After the final process within the Ableton, the signal is routed to the audio interface's console panel where a final mastering limiter sums up all the instruments.

An initial mixing stage is done with the use of the UAD to create the desired tone of the guitar, helping the guitar modelling hardware (the POD xt) to create later a more authentic guitar tone in the signal path. The strings' timbre and the change in their timbre after a couple of hours of performance have not been taken into consideration, since there is always the option to use different guitars with brand new strings between songs. However, since a contemporary popular music song may vary in length from a couple of minutes to an average of 10, considering the strings' timbral changes within that period will not affect the outcome of the research.

In the UAD console, multiple plugins can be applied in the same way that plugins are inserted in a track in the digital audio workstation. When the signal enters the UAD, a preamp with the unison technology is applied to saturate and model the signal by adding 10db to push the amp simulation harder and saturate the signal as well as applying some equalization. After the preamp, a high pass filter and another EQ have been applied to further equalize the signal by adding back some very low-end frequencies, as shown in Figure 73.

Figure 73: MORALIS, C. (2017) 'UAD Guitar input effect chain' [Screen Shot]

Fig. 74 shows a typical signal path used in this project within this pedal board.

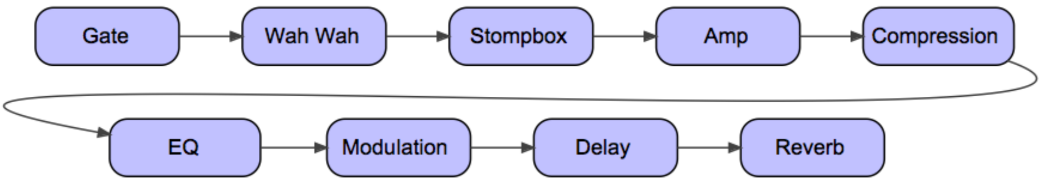


Figure 74: MORALIS, C. (2017) 'POD XT signal path' [photograph] (Designed with VUE software)

After the guitar signal has passed through the desired amp modelling and compression, modulation and other effects have been applied. The signal goes through the UAD console again. This time only a compressor is applied to further balance the overall dynamics of the guitar, setting another stage of gain. Fig. 75 shows the compressor applied and its settings.

Figure 75: MORALIS, C. (2017) 'POD XT to UAD' [Screen Shot]

The signal is fed into Ableton Live where a high pass filter has been applied to adjust both the overall timbre and dynamic response. The frequency has been set at 238HZ and the resonance at 6.6%. As shown in Fig. 76, no envelope or LFO modulation is applied. This plugin has no reported sample latency.

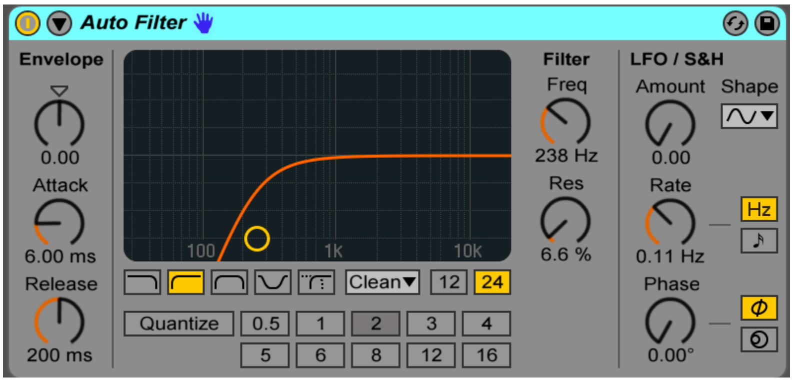


Figure 76: MORALIS, C. (2017) 'Guitar Input Ableton' [Screen Shot]

As explained previously, the signal is fed twice in the mixing chain to manipulate its timbre extensively, but also, as explained later, to effectively manipulate the timing consistency. Since the band has one guitarist, to achieve a true stereo signal the 'Mimiq Doubler' from TC Electronic has been used. This guitar pedal splits the mono signal into left mono and right mono channels and creates a doubling effect to produce a stereo signal. This stereo signal is being generated by delaying one track and continuously varying its pitch. This feature has been used to exaggerate the stereo image of the guitar.

Since this pedal is operated by the performer's foot, to achieve real-time automation without the performer's intervention the pedal has been placed as shown in Fig. 77.

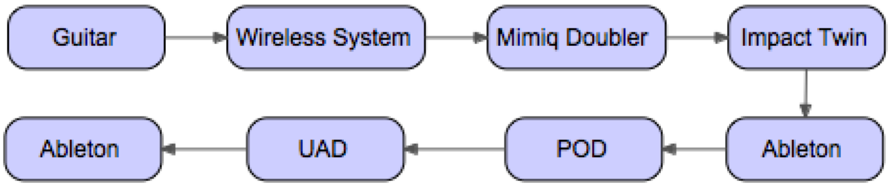


Figure 77: MORALIS, C. (2017) 'Mimiq Doubler Connection' [Screen Shot]

The flat guitar signal from the wireless system is fed into the Mimiq Doubler pedal and split into two channels. These two channels are fed into another additional audio interface; because of the limited input channels of UAD, the Impact Twin has also been used, and from there to Ableton Live. In Ableton, there are two tracks with different panning automation. By automating the panning and the number of channels operating we can define how many guitars will be heard in the project and what their stereo image will be.

- **AUTOMATION**

All sounds and effects are controlled automatically and are pre-programmed with the use of CC messages. This includes patch changes, modulated parameters and others. This process allows the performer to focus on his/her emotional expression and technical aspects of the live show. To control the parameters and the patches of the POD xt modelling hardware accurately, automation has been applied through MIDI Controller Command (CC) messages. Fig. 78 shows the ControlChange8 MAX for Live patch that sends different CC messages to the POD xt.

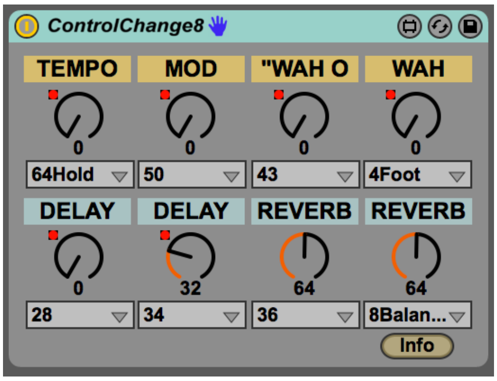


Figure 78: MORALIS, C. (2017) 'ControlChange8' [Screen Shot]

This MAX for Live patch controls parameters such as the tempo through CC messages, since this is the only way the POD xt can receive information to alter tempo, modulation, wah-wah, delay and reverb. Since it is essential for the performer to focus on his performance, these parameters have been automated. Figs. 79 and 80 show some typical examples of this procedure.

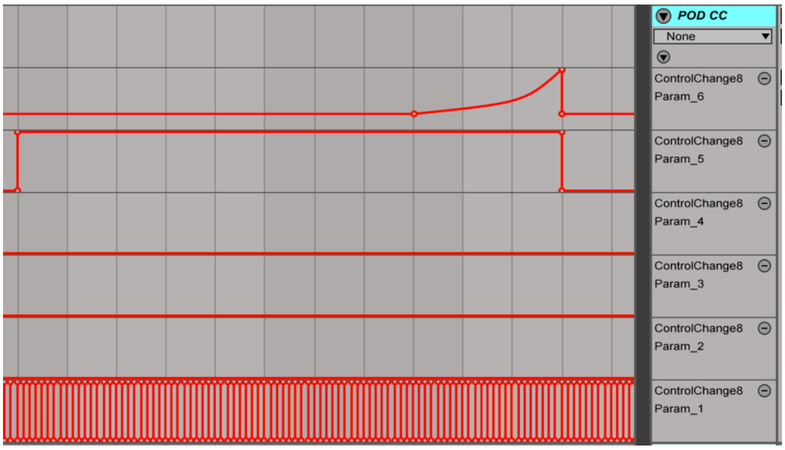


Figure 79: *MORALIS, C. (2017) 'ControlChange8 automation example 1' [Screen Shot]*

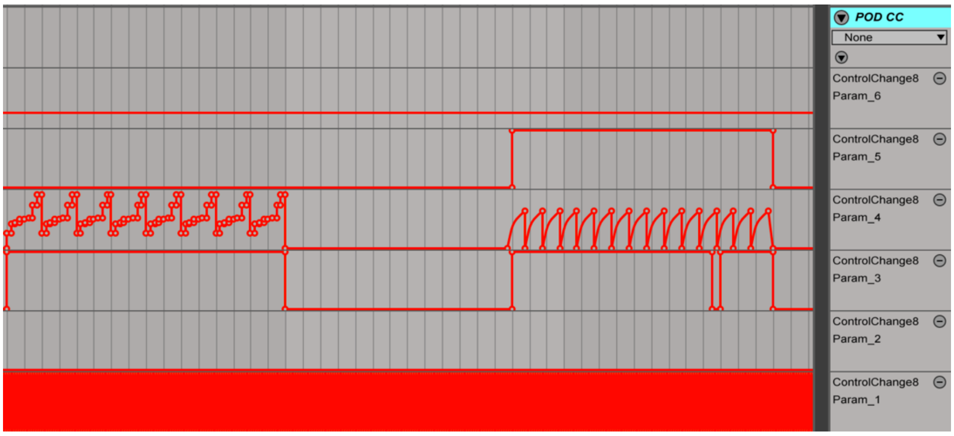


Figure 80: *MORALIS, C. (2017) 'ControlChange8 automation example 2' [Screen Shot]*

Concerning the synthesized sounds, since the midi guitar controller continuously sends midi data to the laptop, the midi track is automated to be enabled or disabled at certain parts where the synthesized sounds should be heard, while CC messages are programmed to change the patches of the synth automatically. See Fig. 81.

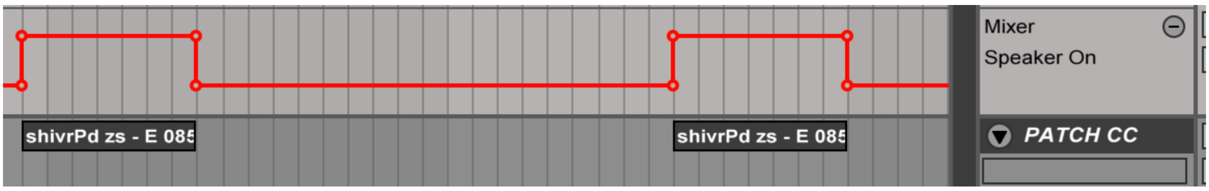


Figure 81: *MORALIS, C. (2017) 'Virus Powercore patch and activation' [Screen Shot]*

Furthermore, two effects have also been used: a delay and a filter. These two plugins affect the overall guitar signal right before it hits the final compressor and limiter. There is no need for extra reverb on the overall signal. However, at this stage, a reverb could also be applied according to the producer's aesthetics. These two effects are also automated to balance the studio production aesthetics and the precision of the automated parameters with the live human performance.

Fig. 82 shows an example of how the filter has been applied to the guitar riff in the song: 'So True'.

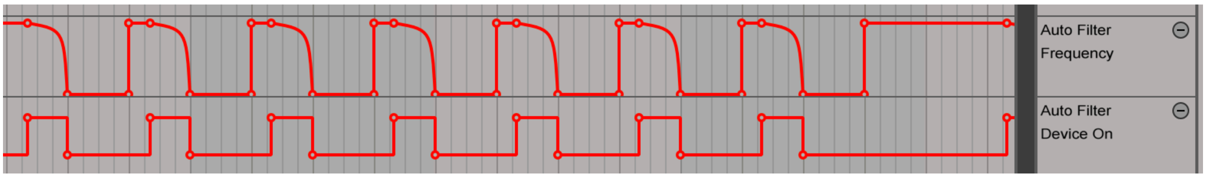


Figure 82: MORALIS, C. (2017) 'So True' - Filter on the Guitar Track' [Screen Shot]

- **SO TRUE (Filter on the overall guitar track):** Audio Example 36

The delay also has been produced in the same way. Fig. 83 shows an example from the same guitar part.

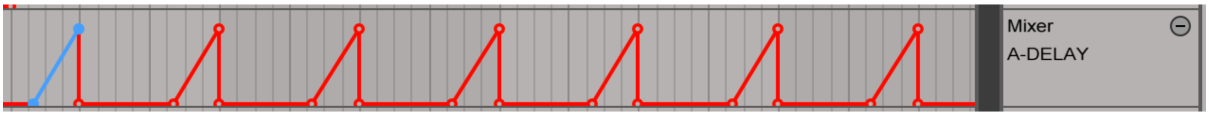


Figure 83: MORALIS, C. (2017) 'So True' - Delay on the Guitar Track' [Screen Shot]

- **SO TRUE (Delay on the overall guitar track):** Audio Example 37

However, there are times in EDM where the whole track needs to be muted musically to follow the overall rhythmic behaviour. In this same guitar riff, a mute process is applied to help 'exaggerate' the silent parts. See Fig. 84.

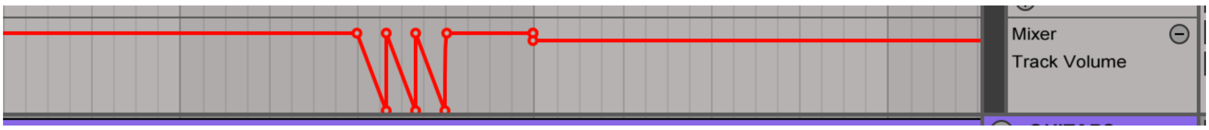


Figure 84: MORALIS, C. (2017) 'So True' – Mute parts on the Guitar Track' [Screen Shot]

- **SO TRUE (Mute on the overall guitar track):** Audio Example 38

CHAPTER 6

6. TIMING CONSISTENCY

The following section discusses the different ways to combine machine-like processes with the human, live performance. In EDM, the synthesizers are mostly fixed to the grid, especially when it comes to bass sounds. However, lead synthesizers or melodies, in general, tend to be left unquantized. According to Zagorski-Thomas (2014, p.51), *‘whenever a piece is performed, the tempo, the precise tuning, the rhythmic microtiming, and the instrumental and vocal timbres will always be different and the combination unique.’* The amount of tolerance of the unquantized performances relies on the producers’ or the performers’ aesthetics. However, small adjustments to the microtiming deviation of some parts are necessary as part of a ‘studio feel’ production.

6.1 Midi quantization

In EDM, the drum samples, especially the kick and the snare, are fully quantized to the grid in most songs to maintain a time consistency and to allow DJs to recognize the bpm (beats per minute) of the current song and mix it with the following song in the DJ’s list. However, these quantized performances, which according to Anne Danielsen (2014, p.1) are the ‘*exaggerated rhythmic expressivity of the machine’,* eliminate the micro-timing deviations that may result from a human performance. For this reason, and to amplify the liveness of the track, producers tend to overlay anything up to 16 bars of recorded, human-performed, audio loops of hi-hats or other percussive sounds such as bongos, shakers, tambourines, and other synthesized rhythmic elements.

The layering of multiple rhythmic recordings enriches the micro-timing deviations between the sounds, providing the listener with a more ‘human’ performance. However, as Basil and Samplecraze.com (2006) suggest, ‘*the more layers you use, the more cluttered the sound will sound. This is down to a number of factors; primarily, frequency clashes, frequency boosts, and phase*’ and ‘*layering drums “correctly” is both technical and artistic*.’ For this reason, any other percussive loops triggered by the DJ in this project will act in the same way as a percussionist adding more rhythmical patterns on top of a drummer’s groove pattern.

According to Fruhauf, Kopiez, and Platz (2013), ‘*Our results show that microrhythmic deviations have a considerable influence on the perceived quality of a groove-oriented drum pattern*.’ Bringing that to electronic music, it is necessary to focus on the timing deviations that may affect the perceived quality of the drummer’s performance and suggest ways that this may amplify the perception of liveness. To achieve timing consistency and fixed-to-the-grid live performances, without affecting the drummer’s expressivity, a real-time midi quantizer has been tested. As explained earlier, the concept of real-time quantization is that the incoming midi is forced to the next interval that has been set. However, to understand better how the midi quantization process is working, two examples are shown below.

For the next examples shown in Figs. 85 and 86, a sixteenth-note midi quantization has been set. On the left is shown an unprocessed recorded performance and on the right is shown its quantized version.

<u>Performed (Unquantized)</u> <u>Processed (Quantized)</u>

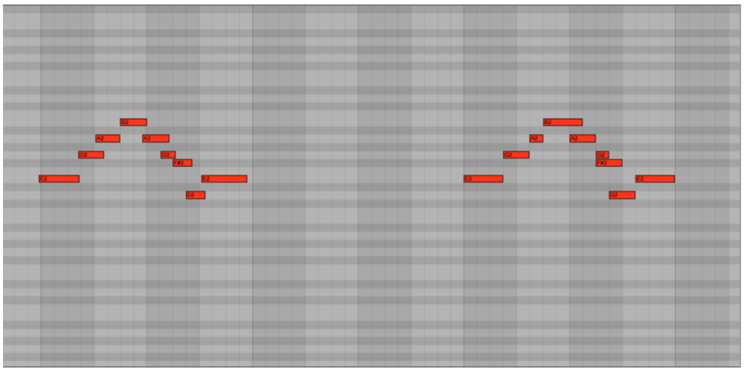


Figure 85: MORALIS, C. (2016) 'Midi Quantization – Example 1' [Screen Shot]

<u>Performed (Unquantized)</u> <u>Processed (Quantized)</u>

Figure 86: MORALIS, C. (2016) 'Score: Midi Quantization – Example 2' [Screen Shot]

As shown in Fig. 86, any notes played after the sixteenth interval have been moved to the next one. Furthermore, the melody that is now heard is very different from the one the performer intended to play. To have the processed (quantized) notes correctly distributed over the timing grid, the performer must play them before the sixteenth interval, as shown in Fig. 87:

<u>Performed (Unquantized)</u> <u>Processed (Quantized)</u>

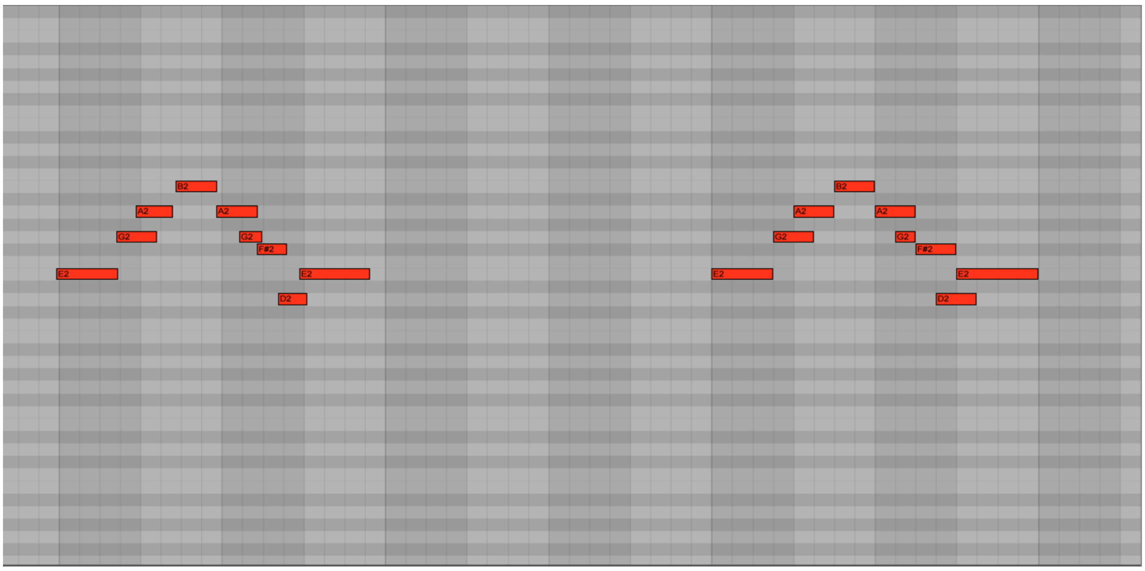


Figure 87: MORALIS, C. (2016) 'Midi Quantization – Example 2' [Screen Shot]

The approach followed suggests that the first note of the bar will be quantized, to trigger the arpeggiated kicks and snares, while the arpeggiator will generate the next notes. However, to maintain the expressivity of the performance based on timing, the hi-hats are left unaffected, as are the dynamics of the drum kit, including those arpeggiated. To understand this procedure better, Fig. 88 shows an example of a two-bar score.

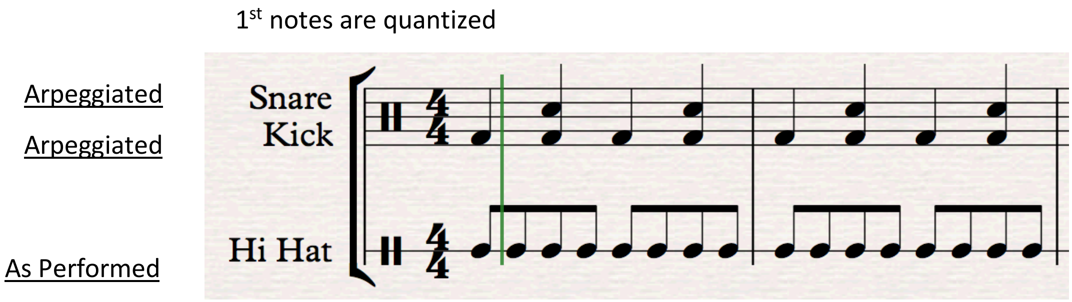


Figure 88: *MORALIS, C. (2016) 'Partial Quantization – Score' [Screen Shot]*

To trigger the arpeggiator on time, the first note(s) of the bar need(s) to be played slightly before the click so that the midi quantizer will send them to the first beat. Since an arpeggiator comes on the first beat, grasps the performed note and duplicates it, and sends it to the next interval, in this case there is a quarter note for the kick and half note for the snare. However, since the arpeggiator creates half notes on the first and third quarter of the bar, a quarter note arpeggiation has been selected and in between notes have been muted. Furthermore, the first hit of the snare is also unquantized, unless it comes arpeggiated from the previous bar. Also, all drum fills are not quantized and in parts where the arrangement suggests non-fixed timing (claps on a pre-chorus bridge, for example), these are left unaffected and as they are naturally performed. Putting that in context, Fig. 89 shows how the midi quantizer and arpeggiator have been set up for the following song:

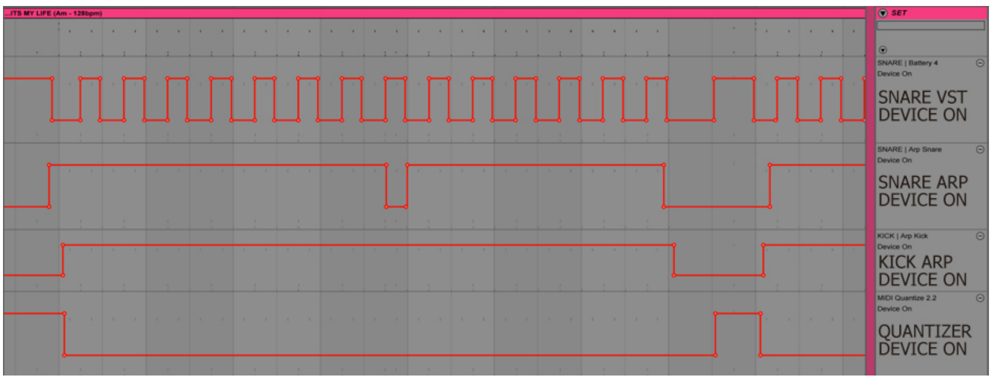


Figure 89: *MORALIS, C. (2016) 'Partial Quantization – Programming 1 – Score' [Screen Shot]*

For example, in the song 'It's My Life', the first snare is quantized along with the first kick and crash. Then the midi quantization is bypassed followed by the arpeggiator, which repeats the kick and snare. On the fourth bar, the snare arpeggiator has been bypassed to let the drummer do a small snare fill, while on the eighth bar both arpeggiator and midi quantization are bypassed to prepare for the next groove. As is also shown, the snare has been muted on the first and third note since the arpeggiator repeats the sound on every quarter. Fig. 90 shows the midi quantization device and the arpeggiator device:

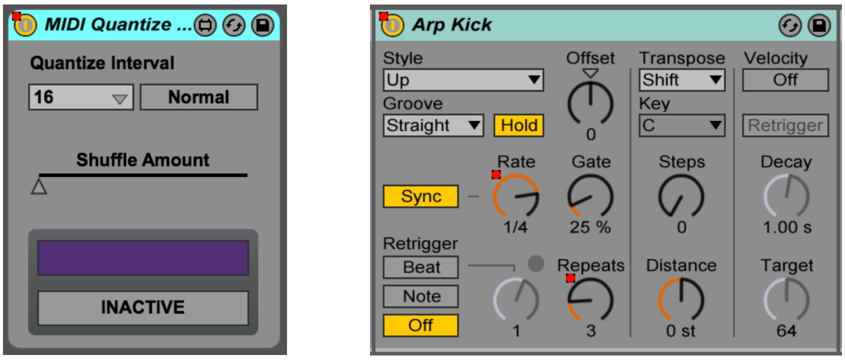


Figure 90: MORALIS, C. (2016) 'Midi Quantizer and Midi Arpeggiator – Score' [Screen Shot]

For Hass (2014, p.190, p.221), *'the 'remarkable' and 'miraculous' become 'mechanical' and 'inhuman,' and this impossibly perfect articulation becomes a marker of a lack of expression- inhuman rather than super-human…The notion that an edited solo is a creative collaboration rather than 'cheating' is anti-intuitive in most forms of a musical audience.'*

In the case of 'American Woman', the arpeggiator has been used in a slightly different way to avoid a 'mechanical' performance. As shown below, the arpeggiator switches off on the third eighth for two quarters and comes back on the last eighth of the bar while the snare is arpeggiated on the second and fourth as usual. This setting allows the drummer to perform differently every time the weak or upper beats and to give the appropriate 'feeling' to his groove. See Fig. 91.

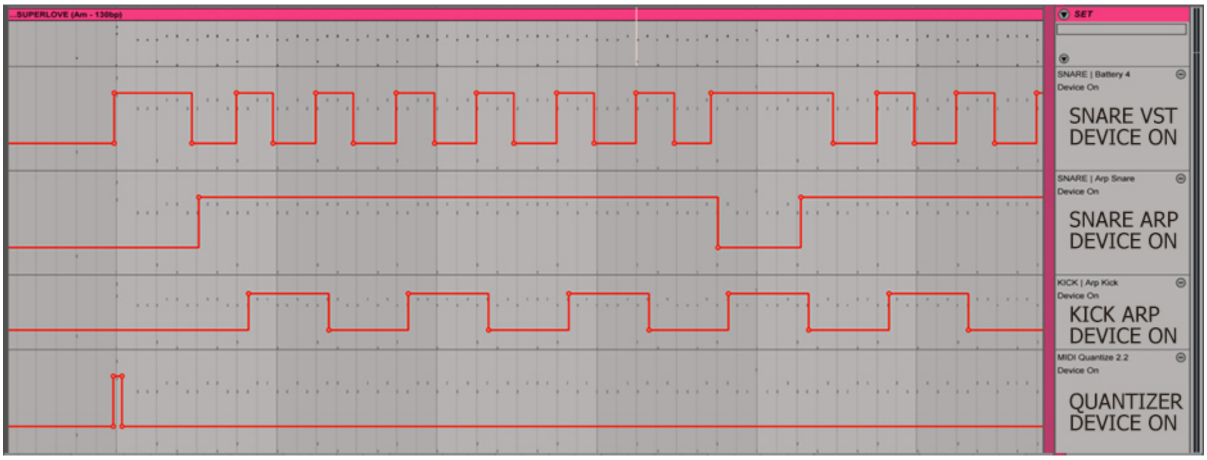


Figure 91: MORALIS, C. (2016) 'Partial Quantization – Programming 2 – Score' [Screen Shot]

As the strong beat (for example, the first beat of the measure, or the third in 4/4 time) or the most pronounced parts of a groove are quantized, and the rest are left unquantized, this is another way to balance timing consistency and human performance.

This technique of using partial midi quantization with the arpeggiated samples may also amplify the drummer's expressivity, while maintaining the time consistency needed for this type of production.

Below are three video examples demonstrating the real-time midi quantization process:

- **SUPERLOVE - TEST 1:** Video Example 1
- **SUPERLOVE - TEST 2:** Video Example 2
- **AMERICAN WOMAN – TEST:** Video Example 3

When it comes to 'tension and resolution', a common technique in EDM is to speed up the repeated melodic lines from quarters to eighths to sixteenths to thirty-seconds up to sixty-fourths (usually also including a filter sweep). The arpeggiator midi effect has also been used for the performance of these long EDM snare rolls. Although these parts are automated by the arpeggiator, the timbre and dynamics of these fills are again controlled by the drummer so the performer can still give his meaning to the sound.

As shown in Figs. 92 and 93, the first four bars of the fill are performed by the drummer with no quantization applied, while the arpeggiator creates the snare roll for the next four bars.

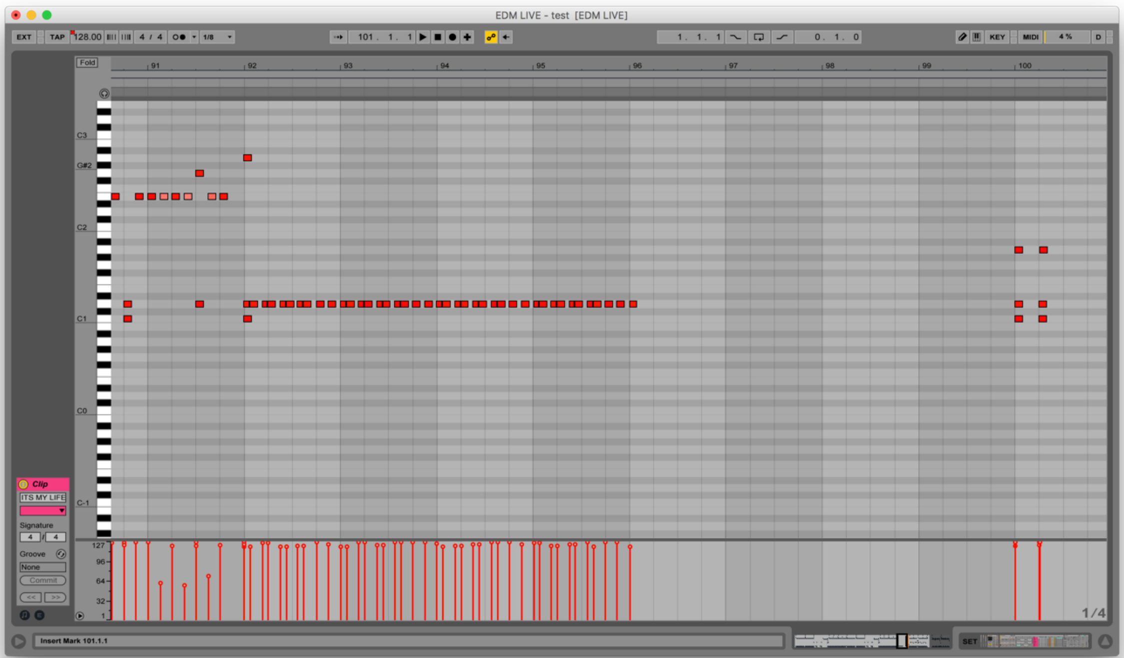


Figure 92: MORALIS, C. (2016) 'Snare Roll 1 – Score' [Screen Shot]

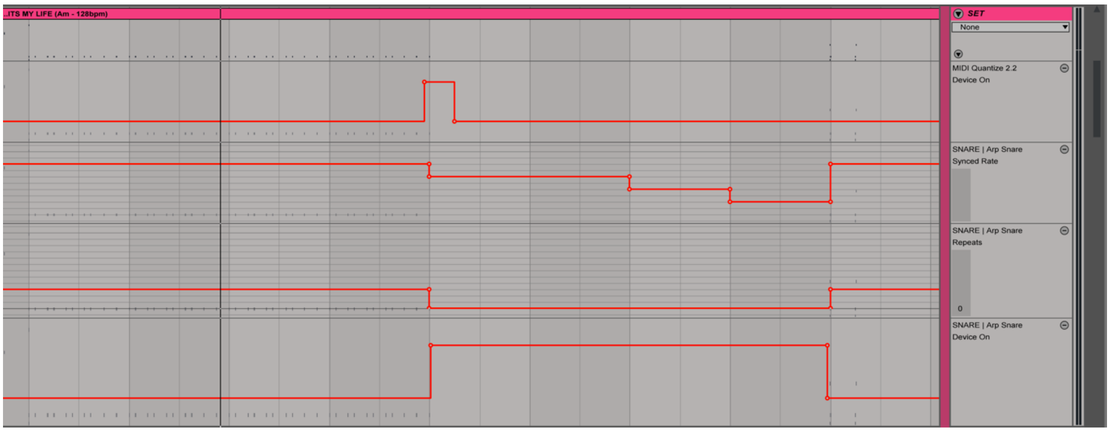


Figure 93: MORALIS, C. (2016) 'Snare Roll 2 – Score' [Screen Shot]

The midi quantizer on keyboards has been used in the same way as it was used earlier for the drums. The midi quantization triggers the arpeggiator precisely on the first beat of the bar, following the tempo correctly. It is necessary to mention that Ableton Live provides the option of a 'clip', meaning pre-recorded looped bars are triggered on time. The reason for using the midi quantizer instead of the 'clip' option is to allow the performer to define the timbre and the dynamics of the arpeggiated melody. See Fig. 94.

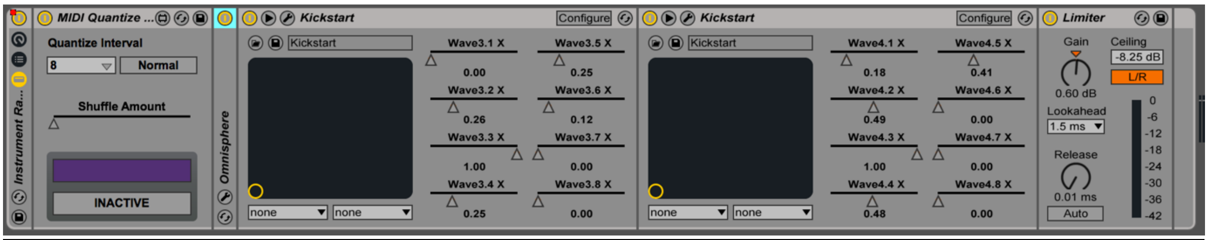


Figure 94: MORALIS, C. (2016) 'Synth FX Chain' [Screen Shot]

In the example of 'Smack My Bitch Up', since the melody needs to be fully quantized, all notes are performed slightly before the click, enabling the performer to control his dynamics and deliver a different performance every time through a variation of the midi velocity. See Fig. 95.

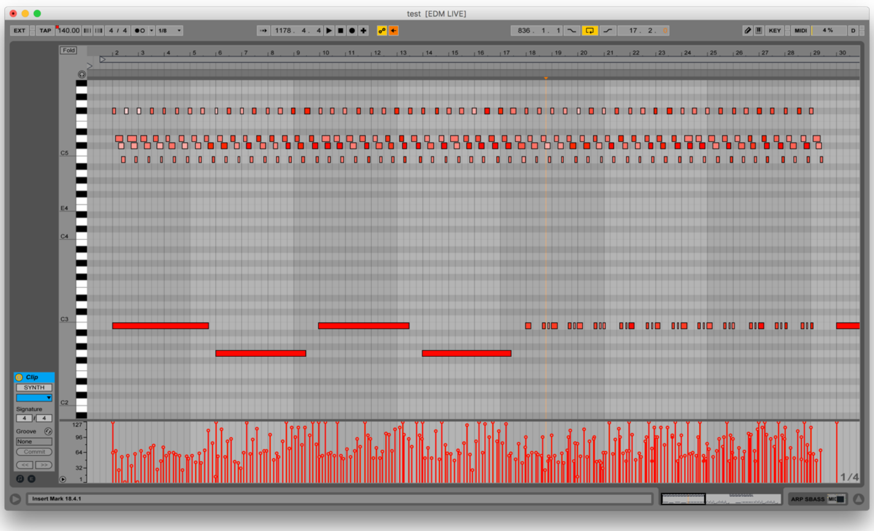


Figure 95: MORALIS, C. (2016) 'Midi Velocities' [Screen Shot]

6.2 Partially synchronized arpeggiation

Another way to control the timing consistency while maintaining the performer's expressivity through notes played off the click, is the use of arpeggiation synchronized to the song's position. When the performer triggers the arpeggiator, the notes will follow the song's structure. With this technique, the effects are out of time following the natural performance, while the arpeggiator is quantized to the grid.

Fig. 96 shows the lead used in the song 'Explode'. The exaggerated unquantized performance is only to show clearly how the delays are behaving while the fixed-to-the-grid arpeggiator keeps its rhythmic pattern quantized.

- **LEAD (PARTIAL SYNCHRONIZATION):** Audio Example 39

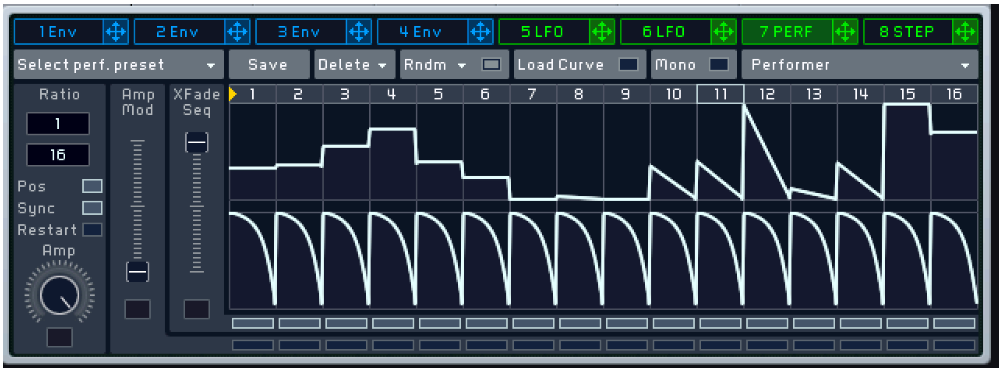


Figure 96: MORALIS, C. (2016) 'Partially Synced Arpeggiation' [Screen Shot]

As Zagorski-Thomas (2014, p.182) mentions, *'However, the ubiquitous use of machine-accurate tempi in popular music forms has also led to the interesting phenomenon of pop and rock drummers playing their parts to a click track: an agent – like aspect of technology that not only configures the drummer but, by default, becomes the "leader" of the entire ensemble.'* Bringing that to the keyboard performance, in the case of the song 'Otherside', a different approach has been applied for combining timing consistency and emotional expression. The element that varies here is the note placement, or in other words the rhythmic pattern of the lead synth. To achieve this, a midi quantizer has been applied, along with an arpeggiator, and the keyboard player is improvising alongside these processes.

- **LEAD SYNTH (One note, Arpeggiated):** Audio Example 40
- **LEAD SYNTH (Creatively Performed):** Audio Example 41

Fig. 97 shows the midi notes.

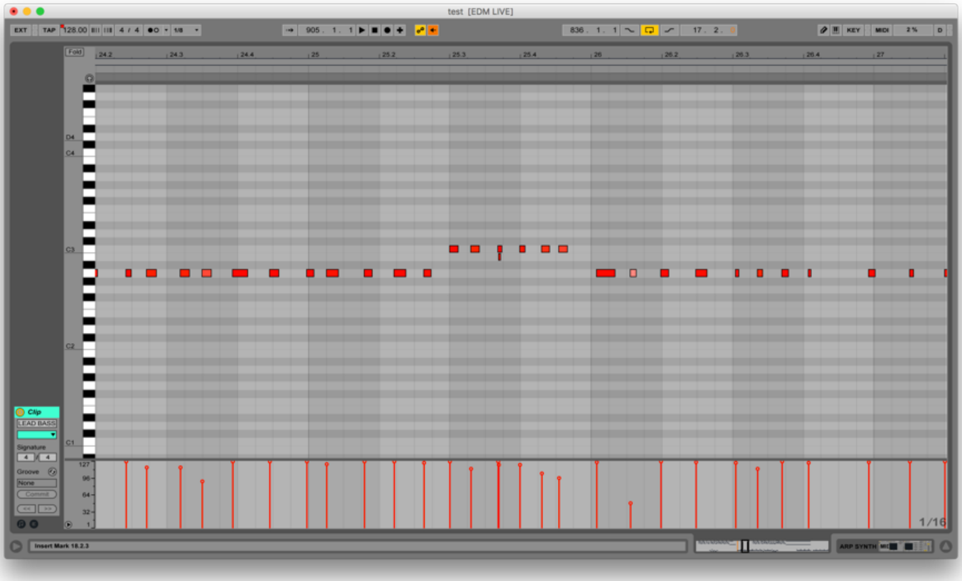


Figure 97: MORALIS, C. (2016) 'Creative Arpeggiation – Midi Notes' [Screen Shot]

6.3 Partially quantized arpeggiation

Another way to maintain the timing consistency, as well as the expressivity of the performance, is the combination of arpeggiation and unquantized performance on the same instrument at the same time. This technique can be applied to parts of a song where only one synthesized sound is played, and it is necessary to bypass the sidechain and LFO volume shaping process since this volume modulation will sound strange without the presence of the kick.

In the following example, a lead synth sound is created by combining the left hand playing a quantized arpeggiator with the right hand playing without any timing quantization.

- **LEAD SYNTH (Quantized Arpeggiator):** Audio Example 42
- **LEAD SYNTH (Without Quantization):** Audio Example 43
- **LEAD SYNTH (Both Hands):** Audio Example 44

Fig. 98 shows the recorded midi notes of this performance. The left hand plays slightly before the click to trigger the arpeggiator on time with the help of the midi quantizer, while the right hand is playing freely according to the emotional expression that the performer intends to give

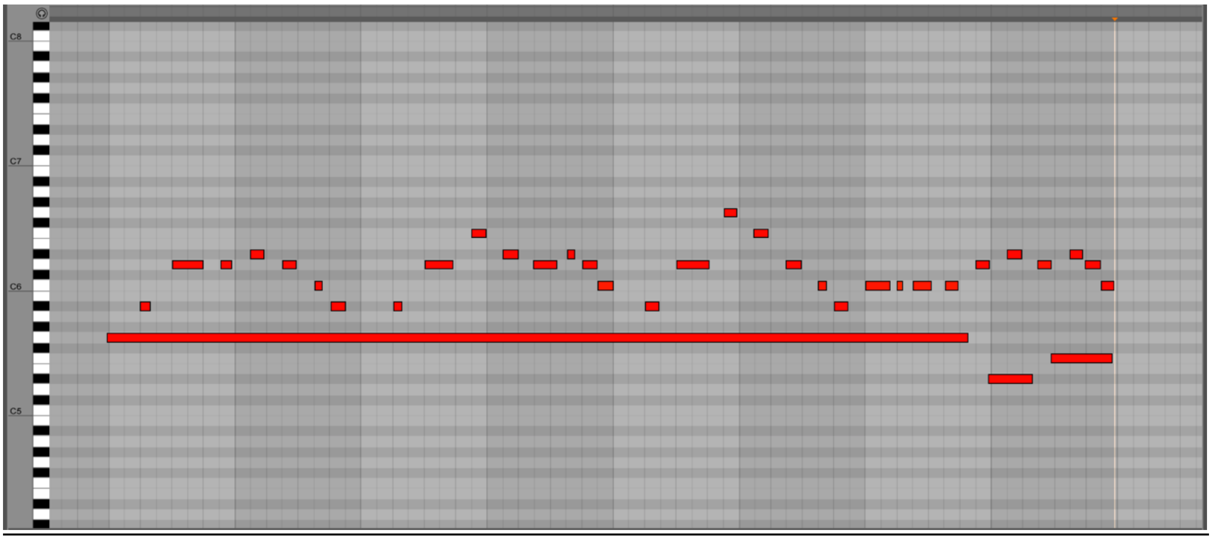


Figure 98: MORALIS, C. (2016) 'Partially Quantized Arpeggiation – Midi Notes' [Screen Shot]

6.4 Volume shaping

According to Kristoffer Yddal Bjerke (2014, p.86), *'In his Auditory Scene Analysis, Bregman defines perception as 'the process of using the information provided by our senses to form mental representations of the world around us.' (Bregman 1990:3)'*. Apart from the midi quantization that enables the performer to play on time, sidechain compression and volume controls have been applied to maintain the perception that an instrument is being performed on time, and its performance is not affected by midi quantization. Instead of quantizing the performance, only by processing in a synchronized to the time grid way the volume of the audio waveform gives the perception of an edited studio performance. To achieve this, the Nicky Romero sidechain effect and the LFO volume envelope shaper tool have been applied along with the muting of specific parts of the performance. See Fig. 99.

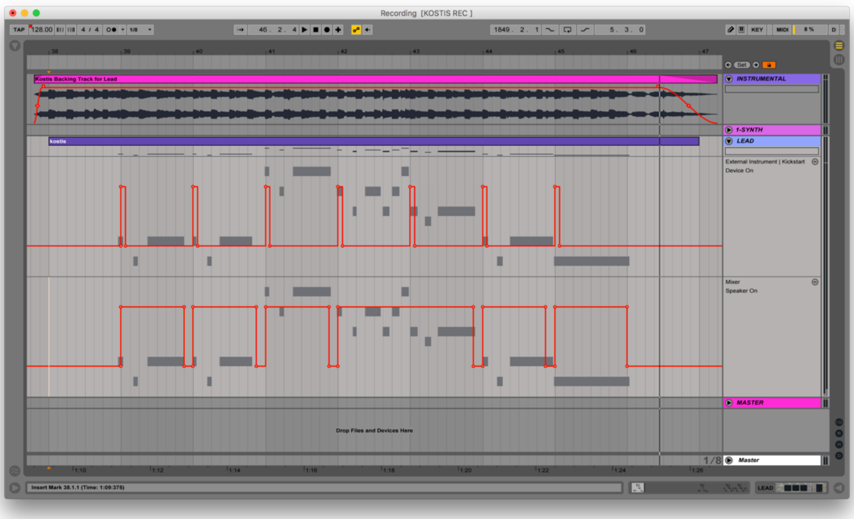


Figure 99: MORALIS, C. (2016) 'Volume Shaping' [Screen Shot]

As shown in the above example of the 'It's My Life' lead, on the last eighth note of each bar the sound mutes, while on the first sixteenth note of every bar the sound is fading in with the use of the sidechaining compression. The notes in between are affected by an LFO that mutes the sound on every quarter note to avoid the constructive gain caused by the lead and the kick playing together. Furthermore, parallel sidechain compression has been applied to improve the instrument's timing perception. See Fig. 100.

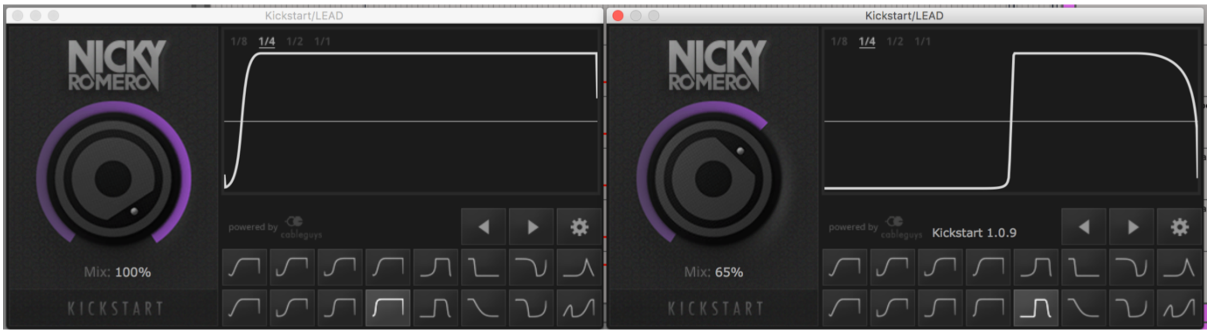


Figure 100: MORALIS, C. (2016) 'Parallel SC' [Screen Shot]

Below is a demonstration of how the lead synth sounds without the volume manipulation and then with the sidechaining and volume process as the keyboard player has performed it:

- **LEAD (Volume Shaping):** Audio Example 45
- **LEAD (Without Volume Shaping):** Audio Example 46

The following example, taken from the song 'Explode', combines the synchronized arpeggiation, as explained in the previous section, along with the volume shaping technique. The first sample is without the audio manipulation and the second is with. The combination of quantized and unquantized elements in the same performance serves both timing consistency and expressivity.

- **LEAD:** Audio Example 47
- **LEAD (SC+LFO):** Audio Example 48

Fig. 101 shows an example from the song: 'Cause I do' with the following guitar riff:

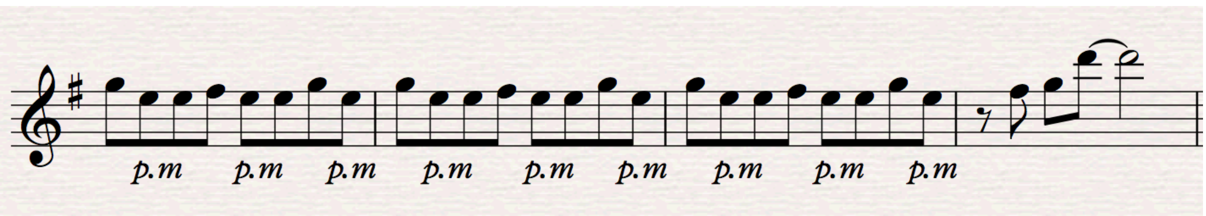


Figure 101: MORALIS, C. (2017) 'Cause I Do – Guitar Riff' [Screen Shot]

The volume curve applied to this riff emphasizes both the desired timing consistency and the expressivity. Apart from the volume curves that lower or amplify the signal, there are also certain areas where the input signal is completely muted. By completely removing the signal from certain areas, it is possible to create the illusion of a 'correct' or in other words fixed-to-the-grid performance, hence a performance with timing consistency. The example below shows the volume process applied to the guitar riff above. See Fig. 102.

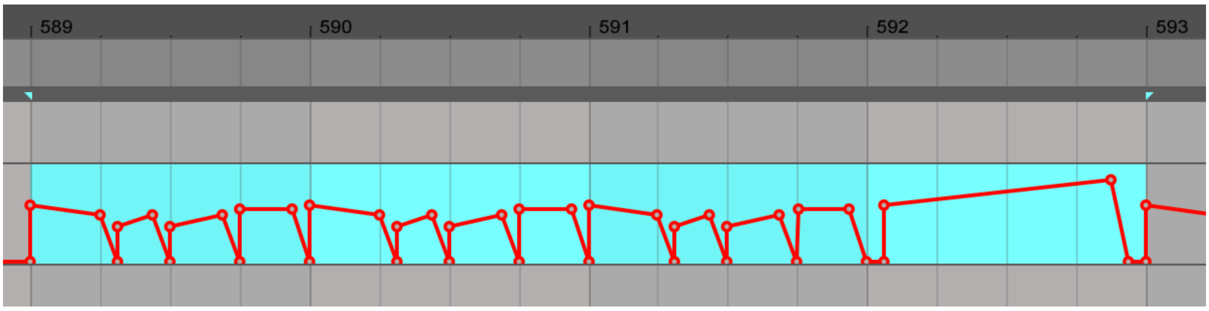


Figure 102: MORALIS, C. (2017) 'Cause I Do – Guitar Riff Volume Curve' [Screen Shot]

At the beginning of this 4-bar guitar riff, the signal is completely silenced to remove and avoid any notes played before the click. The way the curve is designed lowers and amplifies the signal by diminishing or exaggerating the expressivity. Below are two examples, one without the volume curve and one with the above volume curve:

- **GUITAR RIFF (Without the Volume Curve):** Audio Example 49
- **GUITAR RIFF (With the Volume Curve):** Audio Example 50

The volume curve applied to the voice emphasizes both the desired timing consistency and the expressivity. Apart from the volume curves that lower or amplify the signal there are also certain areas where the input signal is completely muted. By completely removing the signal from certain areas it is possible to create the illusion of a 'correct' or in other words fixed-to-the-grid performance, hence a performance with timing consistency. The following pictures show the volume process applied to the voice channel. See Figs. 103 and 104.

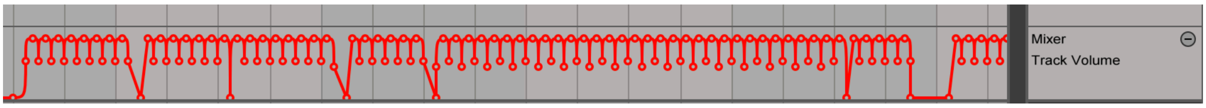


Figure 103: MORALIS, C. (2017) 'Changed The Way You Kissed Me - Voice Volume Curve (1)' [Screen Shot]

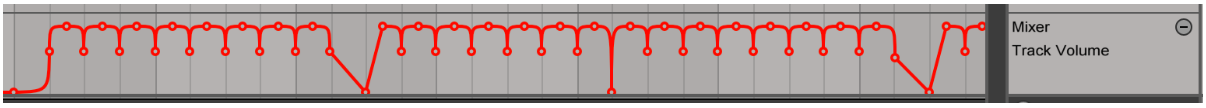


Figure 104: MORALIS, C. (2017) 'Changed The Way You Kissed Me - Voice Volume Curve – (2)' [Screen Shot]

As in the earlier guitar track, an automated volume curve helps balance the signal while giving the perception of timing consistency. However, in the case of voice, it is possible also to control the breaths before each phrase without the necessity for extra plugins such as the ‘Debreath’ from Waves. Below are two examples that demonstrate how the volume curve affects the voice channel.

- **VOICE TRACK (Without the Volume Curve):** Audio Example 51
- **VOICE TRACK (With the Volume Curve):** Audio Example 52

6.5 LFO timing quantization

However, to further assist the performance, an LFO tool has also been applied to control the expressivity in the performance. The following two pictures show an LFO tool used on the open notes while a filter closes on the palm-muted notes to ensure consistency in the timbral shaping. See Fig. 105.

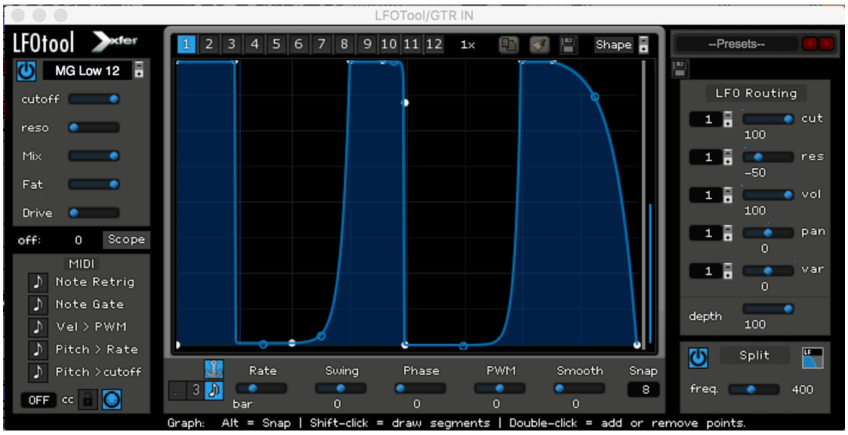


Figure 105: MORALIS, C. (2017) ‘Cause I Do – Guitar Riff LFO tool’ [Screen Shot]

The LFO tool with the volume curve affects the gain input to the amp. However, a filter applied to the palm-muted notes help maintain the intention of the performer. This filter is synchronized to the volume curve, and it operates only on the palm-muted notes. See Fig. 106.

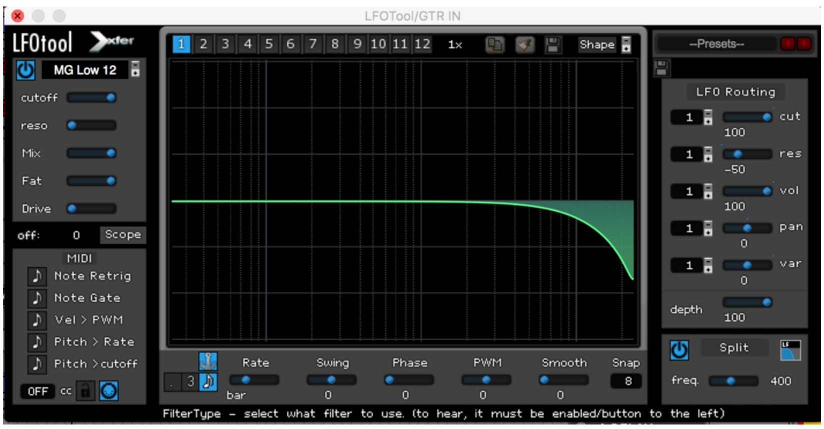


Figure 106: MORALIS, C. (2017) ‘Cause I Do – Guitar Riff LFO tool - filter’ [Screen Shot]

Below are two examples, one without the LFO tool and one with this process.

- **GUITAR RIFF (Without the LFO Tool):** Audio Example 53
- **GUITAR RIFF (With the LFO Tool):** Audio Example 54

Figs. 107-110 show another typical example of this process:

- **Enjoy the Silence – power chords with palm muting**

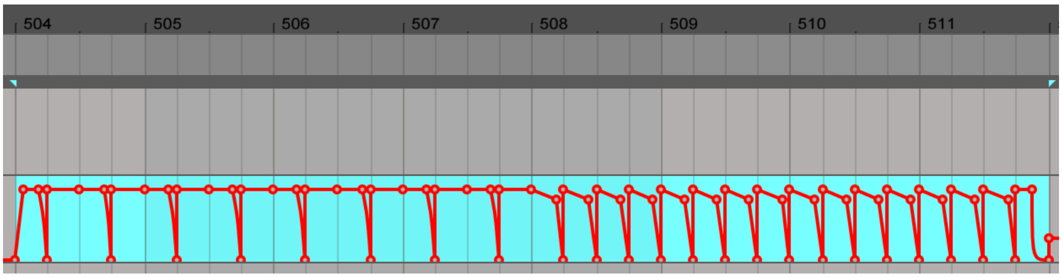


Figure 107: MORALIS, C. (2017) 'Enjoy The Silence – Guitar Power Chords Volume Curve' [Screen Shot]

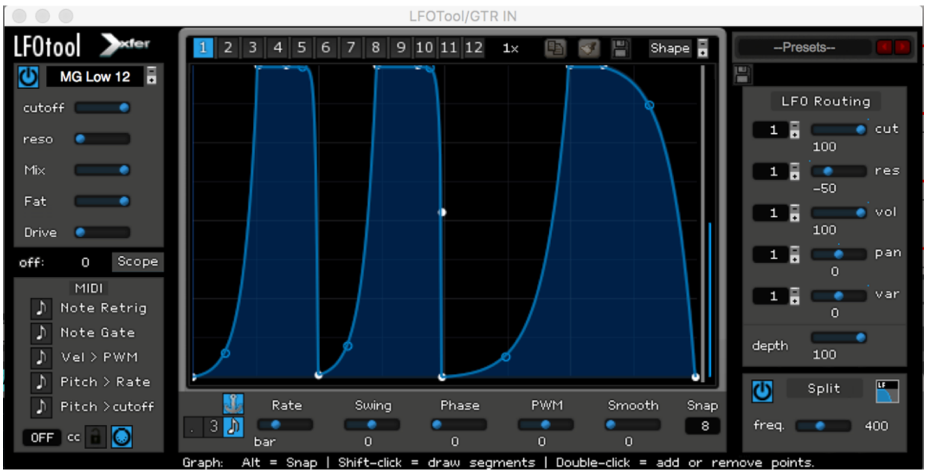


Figure 108: MORALIS, C. (2017) 'Enjoy The Silence – Guitar Power Chords LFO tool' [Screen Shot]

- **GUITAR POWER CHORDS (Without the Process):** Audio Example 55
- **GUITAR POWER CHORDS (With the Process):** Audio Example 56

- **<u>Otherside – clean guitar riff</u>**

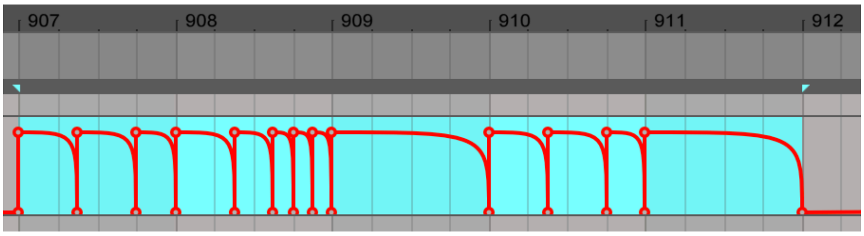


Figure 109: MORALIS, C. (2017) 'Otherside – Clean Guitar Riff Volume Curve' [Screen Shot]

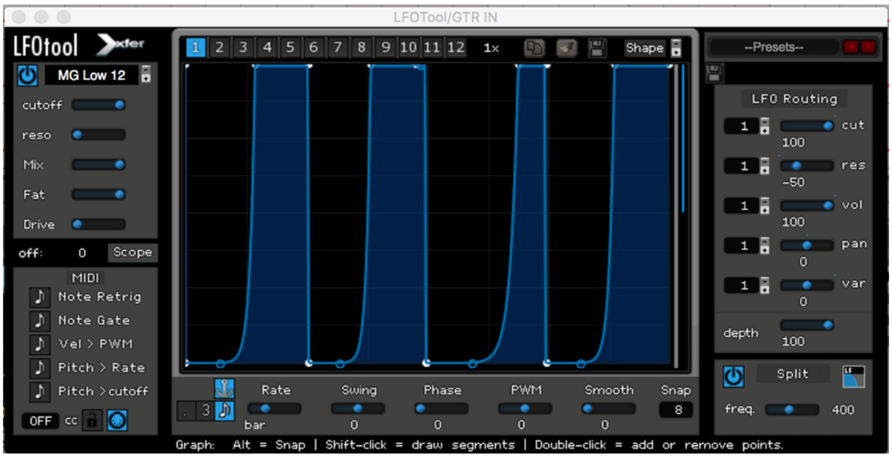


Figure 110: MORALIS, C. (2017) 'Otherside – Clean Guitar Riff LFO tool' [Screen Shot]

- **CLEAN GUITAR RIFF (Without the Process):** <u>Audio Example 57</u>

- **CLEAN GUITAR RIFF (With the Process):** <u>Audio Example 58</u>

6.6 <u>Sequenced quantization</u>

Another way to preserve the timing consistency in a guitar performance is to repeat in real time certain notes on fixed timing positions. To achieve this technique, the plugin 'Beat Repeat' from Ableton Live has been applied. According to Ableton.com (2017) *'Beat Repeat is an extremely powerful effect – capable of longer loops, shorter stutters, wild pitch effects, and more.'* The purpose of this effect is to create stutters and looped phrases; however, it can also help the performance by repeating specific notes on a fixed time grid. By repeating small notes like eighth or sixteenth notes, we can achieve a more timing-consistent performance.

The guitarist can play on top of the repeated notes, without being audible, to maintain the accuracy of his performance. Fig. 111 shows the notes played as they are written on a score to understand the timing pattern better.

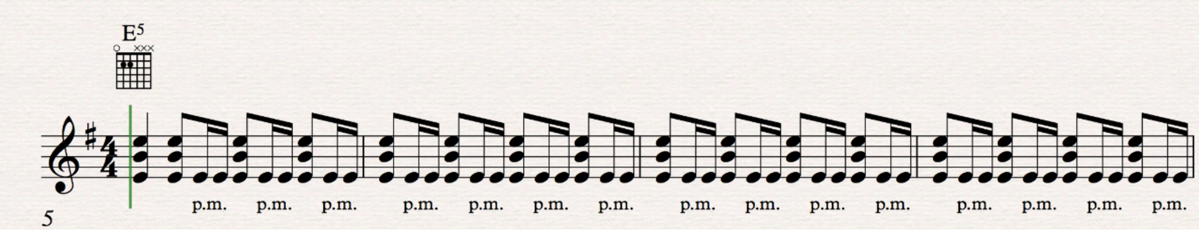


Figure 111: MORALIS, C. (2017) 'Changed the Way You Kissed Me – Dist Power Chords Beat Repeat' [Screen Shot]

In this case, to create a more timing-consistent performance, we apply the 'beat repeat' effect only on the first sixteenth note to produce a second quantized one. Fig. 112 shows the automation in Ableton Live.

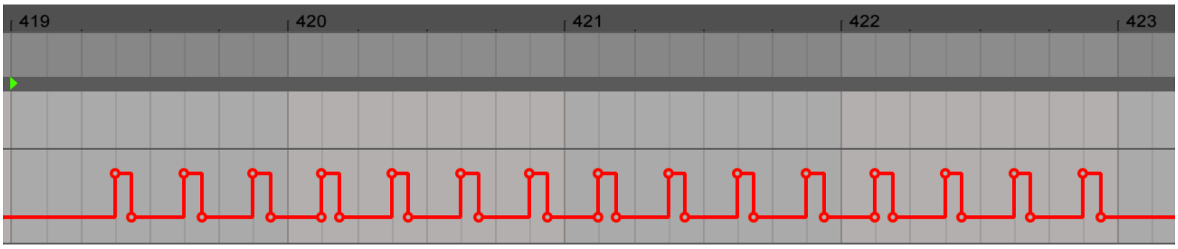


Figure 112: MORALIS, C. (2017) 'Changed The Way You Kissed Me –Power Chords Automation of Beat Repeat' [Screen Shot]

Fig. 113 shows the settings for this effect for the specific guitar part.

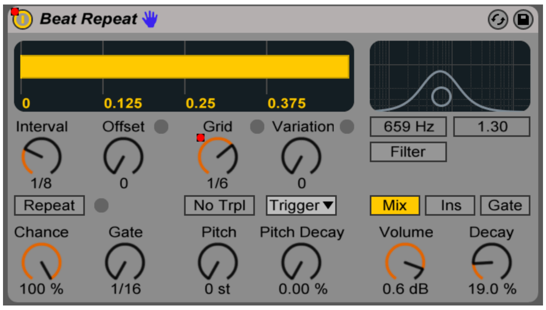


Figure 113: MORALIS, C. (2017) 'Changed the Way You Kissed Me – Dist Power Chords Beat Repeat Setting' [Screen Shot]

The interval defines how often this plugin captures the audio signal. In this case, it has been set to every eighth note. The grid, the size of every repeated slice, is set to sixteenth notes. The variation, which defines how tightly fixed to the grid the repeats will be, is set to zero in order to maintain a fully quantized repetition. Last, the chance is set to 100%, meaning repetition will always take place at the given interval time.

Below are two examples, one without the Beat Repeat and one with this process:

- **DIST POWER CHORDS (Without the Beat Repeat):** Audio Example 59
- **DIST POWER CHORDS (With the Beat Repeat):** Audio Example 60

6.7 Synchronized to the grid effects

The delay effect used in the return effect channel is the UAD Precision Delay synchronized at one quarter. See Fig. 114.

Figure 114: MORALIS, C. (2017) 'Delay Return Channel' [Screen Shot]

However, like all delays, it repeats whatever is fed into the module, so any timing inconsistencies in the performance will be reproduced. To maintain the perception of timing consistency, a volume LFO has been placed before the delay to feed audio only on fixed timing positions. See Fig. 115.

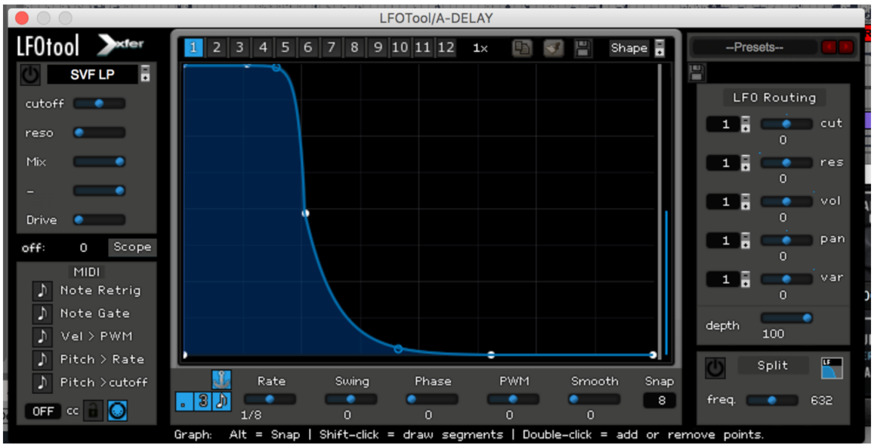


Figure 115: MORALIS, C. (2017) 'LFO on the Delay Return Channel' [Screen Shot]

This LFO tool operates permanently on every eighth note. By being a fast LFO, it helps to maintain the timing consistency and get noticed as a further process on the delay signal. Below are two examples, one without the LFO tool and one with this volume process.

- **CAUSE I DO (Guitar without LFO on the Delay):** Audio Example 61
- **CAUSE I DO (Guitar with LFO on the Delay):** Audio Example 62
- **CAUSE I DO (Voice without LFO on the Delay):** Audio Example 63
- **CAUSE I DO (Voice with LFO on the Delay):** Audio Example 64

However, an extra synchronized volume envelope curve has been applied to amplify the perception of the timing consistency. See Fig. 116.

Figure 116: MORALIS, C. (2017) 'Sidechain Volume on the Delay Track' [Screen Shot]

6.8 Synchronized compression

Referring to the EDM artist Skrillex, Pretolesi (2015) said *'…his mixing is actually a very small part of his sound, it's actually sound design and arranging.'* This intricate process of sound design is the critical element of electronic music for achieving a clean, loud and unique sound.

As Pretolesi continues (2015), '*I want to mix into the compression, so the track is breathing a certain way. It forces me to make certain decisions based on what the compressor is giving me back.'* A widely used technique in electronic music when it comes to synthesized or sampled sounds is the creative usage of compression. The standard approach, to give a pumping effect and to create movement in the synth track, is to apply side-chain compression synchronized to the bpm. However, apart from the musical context that this technique delivers to the song, it is also used for mixing purposes since it creates space when the kick hits. This improves the summing process dramatically by avoiding excessive loudness caused by the constructive gain of these two sounds.

Is also important to mention that all instruments are treated as stem group channels, similarly to the stem mastering process. At this stage, a final compressor and limiter have been applied to control and help the final mixing process. Fig. 118 shows the final compression and limiting effects chain:

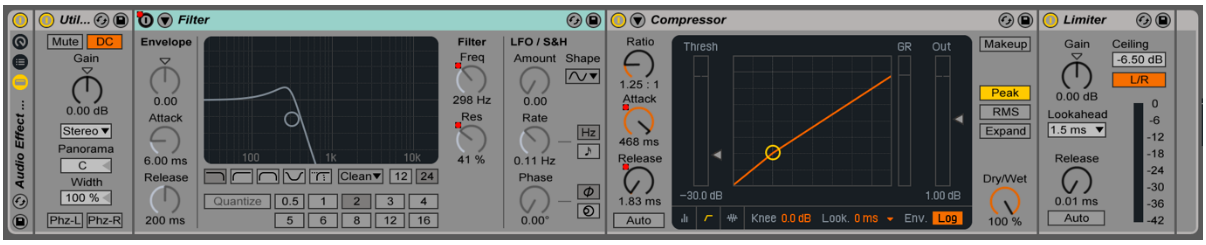


Figure 117: MORALIS, C. (2016) 'Drum Group Channel' [Screen Shot]

The Utility plugin, along with the compressor and the limiter, helps the mixing process while the filter effect is used for musical and aesthetic purposes during specific parts of the arrangement. The enabled DC button in the Utility plugin helps to remove DC offset and extremely low frequencies that are not within the human hearing range. The compressor settings used in this case serve the musicality of the performance.

Although most producers prefer to set these settings according to their aesthetics and not on a mathematical formula, lately a lot of compressors in the digital domains can be synchronized to the project's bpm. Since this research is all about fixed-to-the-grid processes, the synchronization of the attack and release to the song's bpm could help the perception of consistent timing. Below is an example of the duration of the notes at 128 bpm. See Fig. 118.

Delay and Frequency calculator

1) BPM: 128 Calculate Reset

Delay Times and Frequencies:

Dot 1/2 note:	1406.25	ms.	00.711	hz
1/2 note:	937.5	ms.	1.067	hz
Dot 1/4 note:	703.125	ms.	1.422	hz
1/4 note:	468.75	ms.	2.133	hz
1/4T note:	312.5	ms.	3.2	hz
Dot 1/8 note:	351.563	ms.	2.844	hz
1/8 note:	234.375	ms.	4.267	hz
1/8T note:	156.25	ms.	6.4	hz
1/16 note:	117.188	ms.		hz
1/16T note:	78.125	ms.		hz
1/32 note:	58.594	ms.		hz

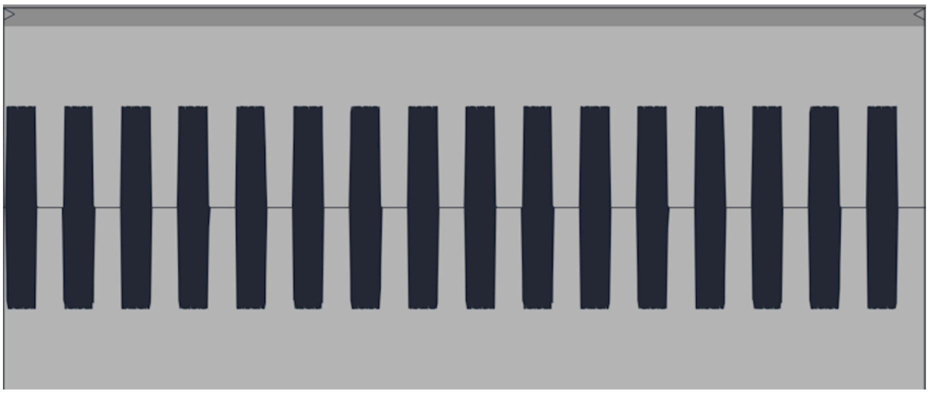
Seconds Per Measure:

1.875 seconds

Figure 118: *MORALIS, C. (2016) 'Delay Time Calculator - 128bpm Note Durations' [Screen Shot]*

According to Anne Danielsen (2014, p.10), *'recording as well as post-production processes such as equalizing and mixing deeply affect how we hear rhythmic phenomena*.' To further understand the importance of a synchronized-to-the-grid compression setting, an example is shown below. Here a quarter note sine waveform is played on every first and third beat at a tempo of 128bpm. The settings used are attack 468ms, release: 1,83ms, knee: 0, ratio: 2:1. Setting the compressor's attack time to 468ms is equal to a quarter note at 128 bpm. See Fig. 119.

Before:

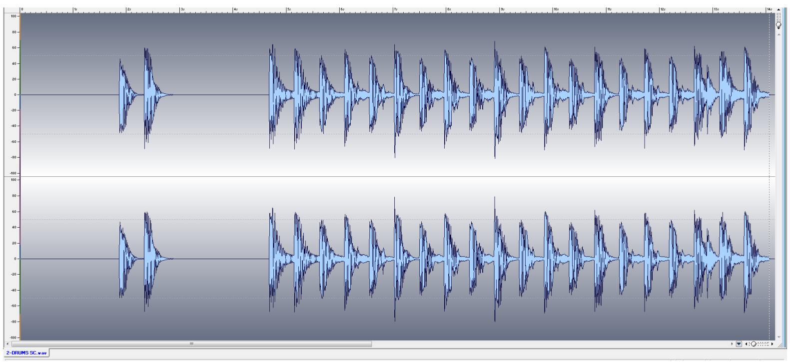


After:

Figure 119: *MORALIS, C. (2016) 'Compressor Test - 128bpm' [Screen Shot]*

The specific limiter used on the overall drums in the group channel is hitting the incoming audio signal fast and acting as a brickwall limiter. Furthermore, the release time is set to the extreme value of 0,01ms. Fig. 120 shows the Drums' waveform:

Before:



After:

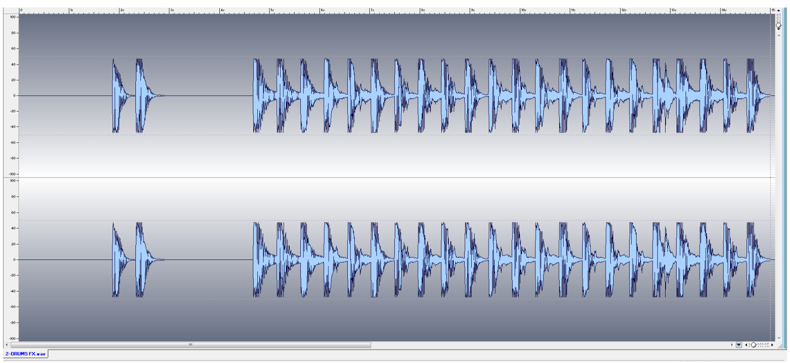


Figure 120: *MORALIS, C. (2016) 'Drums Group Comparison' [Screen Shot]*

According to Ragnhild Brovig-Hanssen (2014, p.159), *'listeners within the culture of country music may perceive an aggressive use of the compressor as opaque, while listeners within the hip-hop culture may perceive the same mediation as transparent.'* However, nowadays even genres such as country music might borrow contemporary processes from other genres. In EDM, the application of excessive compression is widely used, but since the perception of 'liveness' is based on the expressivity of the performance, trimming the audio peaks makes no great audible differences and serves to minimize the constructive gain effect in the final mixing process and its effects on the master compressor and limiter of all the instruments of the song.

Below is the audio example:

- **DRUMS - NO GROUP FX:** Audio Example 65
- **DRUMS - GROUP FX:** Audio Example 66

CHAPTER 7

7. THE MUSICAL DESCRIPTORS

This chapter discusses the variations and combinations of the fixed and varied musical descriptors as they are identified in this research project.

7.1 Drums

The following table explains which musical descriptors are affected and how. See Fig. 121.

SOUND	TIMBRE	DYNAMICS	PITCH	TIMING
KICK	VARIED	VARIED	PARTIAL VARIATION	PARTIAL VARIATION
SNARE	VARIED	VARIED	PARTIAL VARIATION	PARTIAL VARIATION
TOMS	VARIED	VARIED	PARTIAL VARIATION	VARIED
HIHAT	VARIED	VARIED	FIXED	VARIED
CYMBALS	VARIED	VARIED	FIXED	PARTIAL VARIATION
CLAPS	VARIED	VARIED	VARIED	VARIED

Figure 121: MORALIS, C. (2016) 'Drums – Musical Descriptors Table'

Rather than alter the attack, decay and pitch of the compound sounds, these variations are carried out at the sample component level as follows. See Fig. 122.

SAMPLE	ATTACK	DECAY	PITCH
KICK TOP	VARIED	FIXED	VARIED
KICK SUB	FIXED	VARIED	FIXED
SNARE TOP	VARIED	FIXED	VARIED
SNARE SUB	FIXED	VARIED	FIXED
TOM TOP	VARIED	FIXED	VARIED
TOM SUB	FIXED	VARIED	FIXED
HIHAT	VARIED	VARIED	FIXED
CYMBALS	VARIED	FIXED	FIXED
CLAPS	VARIED	VARIED	VARIED

Figure 122: MORALIS, C. (2016) 'Drums – Sonic Characteristics Table'

For example, in the case of the pitch of the kick, the partial variation of the pitch mentioned in the musical descriptors table refers to the top layer, whose pitch varies, and to the sub layer, whose pitch is a specific sample with no real-time variations in its pitch. However, as mentioned in the sonic characteristics table, the fixed pitch sonic attribute of the kick sub, snare sub, and tom sub refers to the note being triggered based on the harmonic content of the song at that moment.

7.2 Keyboards

Regarding the synthesized sounds, since these cover a wide range of timbres and instruments, it is necessary to explain the different combinations that serve the perception of a studio-produced song while allowing the artist to express his ideas and emotions, rather than explaining how the musical descriptors of every sound are affected. The following table shows four different combinations applied in this project. See Fig. 123.

TIMING	TIMBRE	SYNCED EFFECTS	PITCH
FIXED	VARIED	FIXED	VARIED
PARTIAL	VARIED	FIXED	VARIED
PARTIAL	VARIED	VARIED	VARIED
VARIED	VARIED	FIXED	VARIED

Figure 123: MORALIS, C. (2016) 'combined musical descriptors - Table.'

As Auslander (2009) suggests, *'Digital liveness emerges as a specific relation between self and other, the experience of liveness results from our conscious act of grasping virtual entities as live in response to the claims they make upon us'* and *'I am suggesting that some real-time operations of digital technology make a claim upon us to engage with them as live events and others do not.'* As is shown above, 'timing' is the main element that determines the perception of a song that is studio-produced but at the same time live-performed. When the notes are not quantized, other elements such as volume control or synced delays should be applied to give the notion of a performance synchronized to the grid, and when the performance is quantized, other musical descriptors such as timbre and volume should vary.

7.3 Voice

Following the musical descriptor explanation for the keyboards, the following table shows the four different combinations applied on the guitar. See Fig. 124.

TIMING	TIMBRE	SYNCED EFFECTS	PITCH
PARTIAL	VARIED	FIXED	PARTIAL

Figure 124: MORALIS, C. (2017) 'combined musical descriptors of voice - Table.'

From the table above, we can see that all four elements contribute to the perception of a studio-produced song. The use of envelope shaping to emulate the type of timing correction that goes on in the studio allows the balance of characteristics that again suggests the first-person authenticity, the momentary expressive variations of a live performance, and those of the third-person authenticity; in this musical style, the polished 'perfection' of editing in electronic music. Furthermore, the voice comes from the human body, meaning there will always be small variations in the timbre caused by the singer's mood and physical condition. Therefore, its consistency relies on fixed dynamics and equalization.

7.4 Guitar

The following table shows the four different combinations of musical descriptors applied on the guitar. See Fig. 125.

TIMING	TIMBRE	SYNCED EFFECTS	PITCH
FIXED	VARIED	FIXED	VARIED
PARTIAL	VARIED	FIXED	VARIED
PARTIAL	VARIED	FIXED	VARIED
VARIED	VARIED	FIXED	VARIED

Figure 125: MORALIS, C. (2017) 'combined musical descriptors of guitar - Table.'

As shown above, the main element that determines the perception of a song that is studio-produced but at the same time live-performed is the 'timing'. When the notes are not quantized, other elements such as volume control or synced delays should be applied to give the notion of a synchronized-to-the-grid performance. However, where the guitar is sequenced, using the 'beat repeat' for only one sixteenth note, since this is fast, the synced effect can still be fixed as it amplifies the perception of a studio-produced performance.

7.5 DJ

The following table shows the different combinations applied on the DJ track. See Fig. 126.

TIMING	TIMBRE	SYNCED EFFECTS	PITCH
PARTIAL	VARIED	FIXED	PARTIAL

Figure 126: MORALIS, C. (2017) 'combined musical descriptors of voice - Table.'

The table above suggests that the timing and pitch of the DJ samples can vary or be fixed while their timbre should always vary. Again, the effects are synchronized to the grid to help amplify the perception of timing consistency.

CHAPTER 8

8. LIVE SETUP

According to Benediktsson (2017), *'As a live sound engineer, you're always in a lose-lose situation. If the band sounds good, it's their awesome performance. If a band sounds bad, it's all your fault.'* This chapter will discuss the live setup and how all these instruments and processes work together. In most cases, the goals of the live sound production are to provide artists with a comfortable and ergonomic setup on stage while at the same time getting the best sound from their performances and the best sound in the hall.

However, since in this project the goal is to create the aesthetics of a studio-produced song, and there are no acoustic instruments other than the voice making a direct sound, this model is not concerned with the acoustics of the environment. If necessary, the live sound production is focused on enhancing and balancing the pre-installed sound systems and the acoustics of the venue, rather than mixing the band (which is an automated or pre-determined process). This means the final mixing process of this model will remain the same despite the acoustics of the venue, club, festival area or other performance space.

8.1 Band setup

Fig. 127 demonstrates all the instruments and equipment used along with their connections (see Appendix 8 for larger image resolution):

PERFORMABLE RECORDINGS SETUP

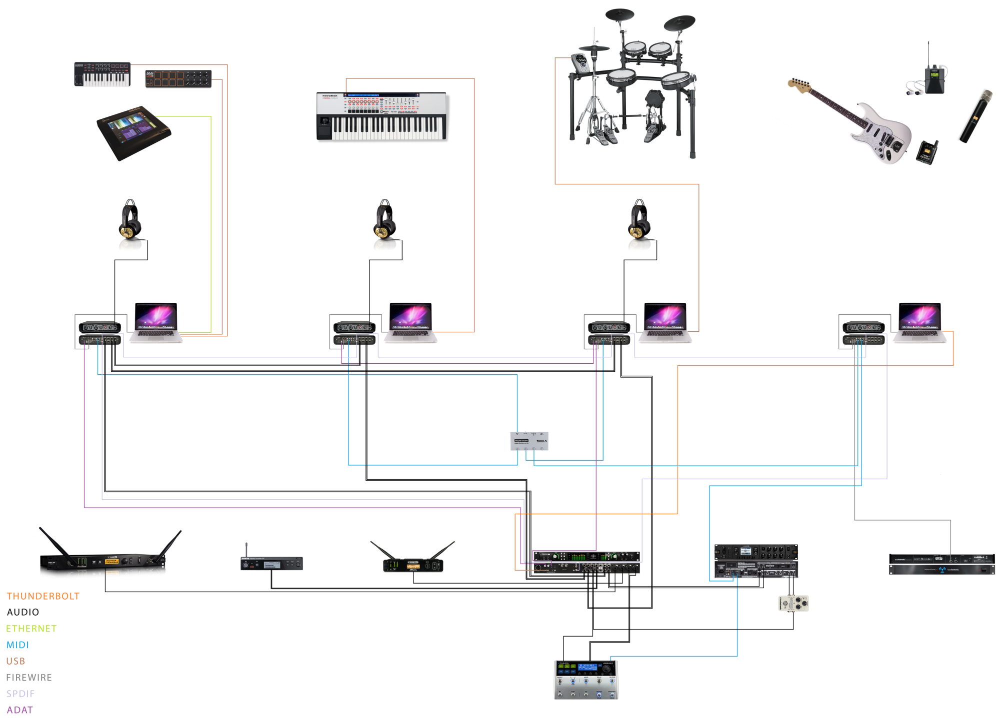


Figure 127: MORALIS, C. (2017) 'The Performable Recordings Setup' [photograph] (Designed with Photoshop software)

Four laptops have been used to provide the necessary overall CPU power. Each computer is working with Ableton Live software. One is for the drums project, one is for the keys project, one is for the guitar and voice project, and one is for the DJ project.

Each of the four musicians uses a separate laptop with a dedicated audio interface. In this case, the TC Electronic Impact Twin audio interface has been used. The master audio interface, where all stems are summed, is the Apollo8 from the Universal Audio. To synchronize the sample rate of each audio interface, the signal is passed through SPDIF connection from one interface to another, beginning from the UAD.

The ADAT (see Appendix 9), as well as the SPDIF connections (see Appendix 10), have been used to minimize the audio signal quality loss when transferring audio between interfaces. The drum and the keyboard stems are being transferred to UAD through ADAT connections, while the DJ's stem is being transferred through a SPDIF connection. The guitar and vocal stems are using the UAD so are mixed internally with the other stems.

8.2 Midi clock synchronization

To synchronize the four computers, Ableton's Live link technology has been used (see Appendix 11). This technology is wireless, and it uses a local Wi-Fi network between the laptops. It has been found to be more accurate than the older midi-clock synchronization technology. See Figs. 128 and 129.

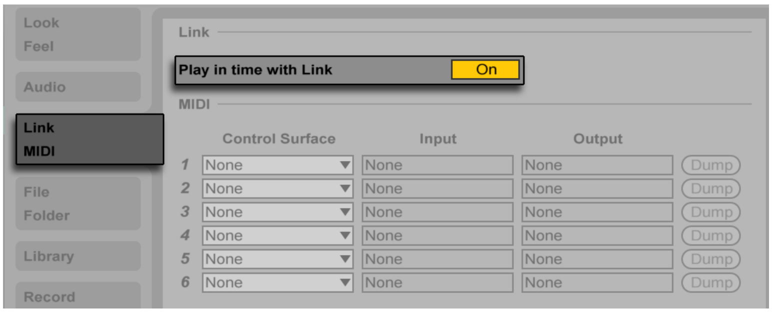


Figure 128: Link 1 (Ableton.com, 2017)

Figure 129: Link 2 (Ableton.com, 2017)

8.3 Play / Stop control

All projects are synchronized to the midi data fed by the DJ's Ethernet controller, as shown in the previous diagram. Since it is not possible to control the start/stop option of all Ableton live projects when using Link, the DJ is sending a midi-note message to the midi-splitter box and from there it is fed to every single audio interface. Hence Ableton Live is used for controlling the projects remotely. Fig. 130 shows how the midi data is being distributed over the four laptops.

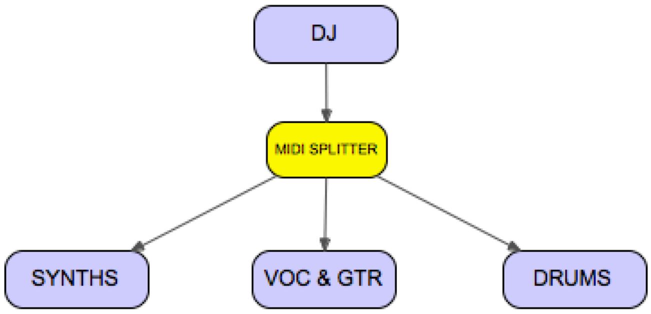


Figure 130: MORALIS, C. (2017) 'Modulated Parameters- Midi Distribution' [photograph] (Designed with VUE software)

8.4 Tempo track

This study has been based upon songs with a fixed bpm, as this is mostly found in electronic music. At this point, all musicians are playing along with a fixed click track. However, there is the option to change the bpm of the project to follow the band's natural performance and in real-time. This can be done with the 'BeatSeeker' Max for Live patch. This patch works as an intermediate timing agent between the drummer's performance and song's tempo track. The song has an initial pre-set tempo, but the drummer defines the tempo of the song. Also, the negotiation and communication between the performers affect the drummer's performance, resulting in humanly created tempo shifts. Fig. 131 shows how the tempo of the song can be defined in a live performance through this real-time process:

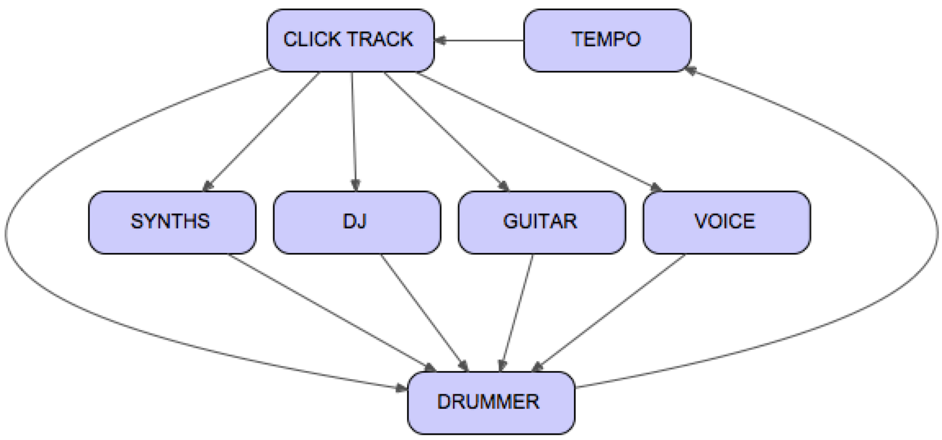


Figure 131: MORALIS, C. (2017) 'Stems – 'Tempo' [photograph] (Designed with VUE software)

Based on the drummer's performance, this patch changes the song's tempo in real time. Since the musician's tempo shifts might be between 4 and 5 bpm, this patch tends to change the tempo gradually, rather than instantly, to maintain a more consistent feel. Fig. 132 shows the 'Beatseeker' plugin.

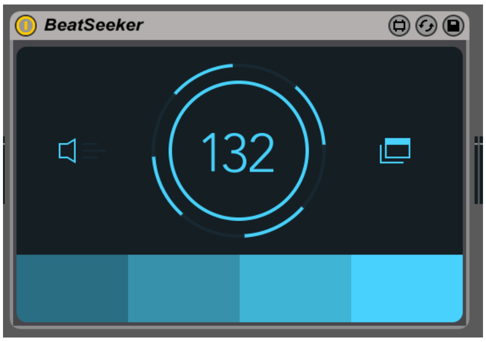


Figure 132: MORALIS, C. (2017) 'BeatSeeker' [Screen Shot]

By applying tempo change, but with fixed mixing and timing relationships, the overall sound could potentially match the aesthetics of a studio-produced live record. The following two examples show the difference between a steady fixed-tempo track and a varied one, with both using the timing correction processes that have been explained in previous chapters.

- **FIXED TEMPO TRACK:** Audio Example 67
- **VARIED TEMPO TRACK:** Audio Example 68

As shown above, the varied version sounds more organic and humanly performed. The interaction between the click track and the drummer's performance improves the overall aesthetics of a live-performed song.

In the following two pictures, the midi data from the drummer's performance is shown. The drummer's rhythmical behaviour and the timing consistency do not have significant differences. Furthermore, Asquini suggests that this process is hardly noticeable when you are not aware that it exists in the background, and at the same time it helps a lot to shape the overall tempo of the song when you know it is there (Asquini, 2017). This means that the tempo shifts caused by his performance, whether intentional or unintentional, cannot affect the overall quality of his performance because of this smoothing effect.

Figs. 133 and Fig. 134 show the two performances:

FIXED TEMPO TRACK

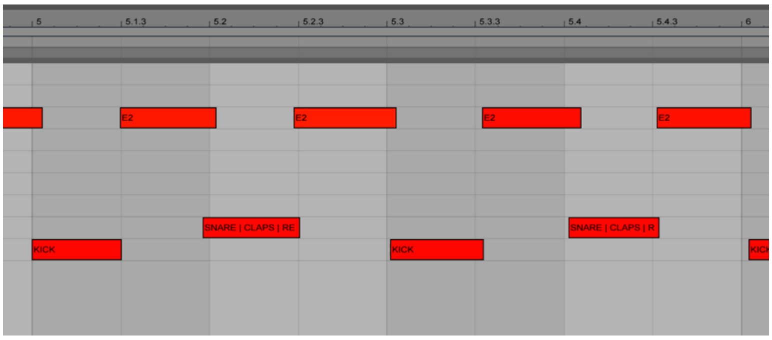


Figure 133: MORALIS, C. (2017) 'Drums: Fixed Tempo Track' [Screen Shot]

VARIED TEMPO TRACK

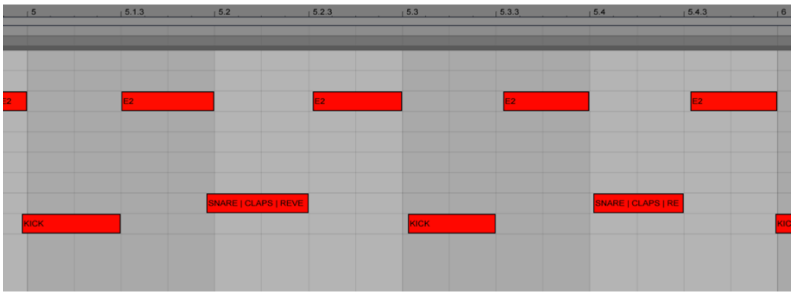


Figure 134: MORALIS, C. (2017) 'Drums: Varied Tempo Track' [Screen Shot]

Although this technology works well on one laptop, in this case of four laptops will not be integrated to avoid potential synchronization issues between the devices over the Link technology.

8.5 Live sound issues

Since the instruments used in this project need a sound reinforcement system, the potential acoustic gain (PAG) is calculated every time according to the singer's microphone and the live venue's acoustics (see Appendix 12). This is also helpful in small venues where the PAG is easily exceeded.

In this case, the only device that could potentially create feedback is the microphone. To minimize or eliminate the frequencies that might cause feedback, the X-FDBK Feedback eliminator from Waves (see Appendix 13) has been applied to the microphone channel. Applying digital anti-feedback technology, in the box, helps to maintain and control the sound of the microphone better than a hardware version.

Fig. 135 shows a typical setting of this technology:

Figure 135: MORALIS, C. (2017) 'X-FDBK' [Screen Shot]

8.6 Latency

To be able to perform through this technology, the objective was to use plugins and hardware that have no or near to no latency. According to Miller (1968), in his paper 'Response time in Man-Computer Conversational Transactions', there are three different orders of magnitude of computer mainframe responsiveness:

A response time of 100ms is perceived as instantaneous.
Response times of 1 second or less are fast enough for users to feel they are interacting freely with the information.
Response times greater than 10 seconds lose the user's attention entirely.

(Miller, 1968)

However, today's real-time applications require near-instantaneous responsiveness. 100ms may be acceptable in conversation but not in dance music.

Below is shown the average latency caused by the software and the hardware:

DRUMS:	10ms
SYNTHS:	15ms
GUITAR:	15-23ms
VOICE:	24-32ms
DJ:	10ms

Figure 136: *MORALIS, C. (2017) 'Latency - Table'*

The guitar and the voice tracks are routed twice into Ableton, causing more overall latency. Also, these are using digital wireless systems that add another 2ms. However, all the musicians have no problem with the latency, and this is due to their engagement with technology and ability to entrain. This also agrees with the research of Barbosa and Cordeiro (2011), as well as that of Boley and Lester (2007), who suggest the 40ms range to be the threshold for optimal performance. This is also in accordance with Lago and Kon's (2004) findings that a propagation latency of 30ms will be unnoticeable since this latency amount is still tolerable.

8.7 Monitoring

Since this project is all about the final processed audio signal, there are no stage monitor speakers used during the performance, only in-ear or regular headphones. This is also because all of the musicians rely on the click track, even with the BeatSeeker software, as this works only with the drummer's performance. However, the balance of every monitor is adjusted according to the performers' needs.

The live performance is separated into different songs, and markers define the beginning of each one. For the performers' convenience, there is a guide track with a pre-recorded guide voice. This gives the performers the song name at the beginning of each song, and if necessary the voice counts during long fill breaks.

8.8 The final process

According to Izhaki (2008), *'…a mix is a sonic presentation of emotion, creative ideas, and performance.'* Since the performers' expressivity is very important for this production and performance model, the final stage of summing of the tracks should not affect the audio signals any further.

All the stems, drums, synths, vocals, guitar and DJ are routed to the UAD console and to the master bus channel, where only a mastering limiter has been applied. In this case, Precision Limiter has been used as this is the most transparent one in the UAD series.

'*Since Precision Limiter is a colorless, transparent mastering limiter — no upsampling is used, nor does Precision Limiter pass audio through any filters — audio remains untouched unless the compressor is working, in which case only gain is affected.'*

(UAD, 2017)

Since all stems are sent to the master channel as mastered stems, the final limiter does not affect their sound but only adjusts the overall gain if necessary.

However, there is always the option to provide the house engineer with the stems rather than a stereo signal. In some cases, this would be preferred as the compressors and limiters installed in clubs and venues may work better than the software plugin.

Fig. 137 shows the final routing path.

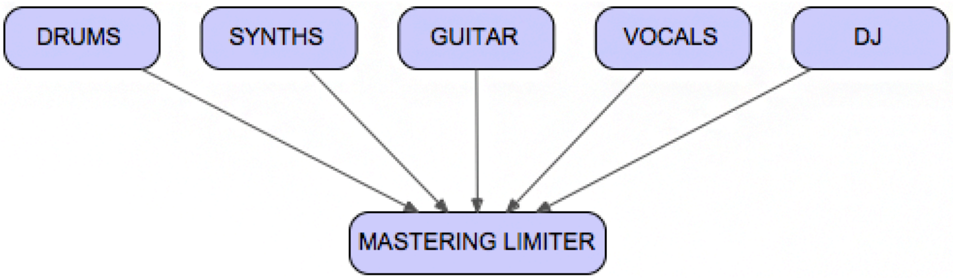


Figure 137: *MORALIS, C. (2017) 'Stems – Final Mastering Limiter' [photograph] (Designed with VUE software)*

CHAPTER 9

9. LIVE PERFORMANCE

The previous chapters established the conditions for creating a production approach that serves the concept of the Performable Recordings model. This chapter discusses the way musicians perform through this model.

The aspects of the live performance analyzed in this chapter are the entrainment process, the cognitive process regarding notes and scores, and finally the creative and improvisational aspect of their performances.

9.1 Entrainment

Bringing all this technology and processing together, apart from synchronizing the machines properly to avoid any phasing or entrainment issues, also requires the human performances to be 'synchronized', i.e. entrained with the mechanical processes.

> *Metric Entrainment can only occur with periodicities in the range from (approximately) 100ms to 5-6 seconds. Within this range, we may grasp a sense of beat (also known as pulse or tactus) in a sub-range of 200-250ms to about 1.5 seconds (240-40 beats/minute), and we prefer to hear beats in the range of 500-700ms (120-86 beats/minute). Thus, very rapid periodicities are almost automatically heard as subdivisions of a slower beat.*

(London, J. 2004).

The average human reaction, according to humanbenchmark.com (2017) is between 200ms and 250ms. However, humans have different reaction times between each other. According to Bilder (2015), *'we have to always think about this as a loop, the perception-action cycle. The cycle occurs every 300 milliseconds. So, three times a second, we're going through this process of evaluating our plans, getting inputs, and through this resonance architecture and through mismatch that alters the resonant states, developing new plans for behaviour.'*

Fig. 138. shows our reaction response times based on our visual reflexes. These results are created with the online humanbenchmark.com test.

DRUMMER:	204ms
KEYBOARDIST:	303ms
VOCALIST/ GUITARIST:	249ms
DJ:	238ms

Figure 138: MORALIS, C. (2017) 'Response Times - Table'

This table works only as an indication and not as a scientific result. These results depend on how fast we perceive the visual cue, the colours used, the process of clicking the mouse and not playing an instrument, our mood and how relaxed we are, as well as the computer itself and the internet connection.

However, it does appear from this that musicians who are familiar with percussive elements get better results, with the drummer having the quickest response time.

This is in accordance with the following parameters that affect the response times:

- *Sensory perception*
- *Receipt of input into our consciousness*
- *Context applied to the input*
- *Decision made based on processing output.*

(Pubnub.com, 2017)

Musical-rhythmic performance is all about the entrainment process. An insight of Jones's research with implications for the study of musical aesthetics is the distinction between two different modes of attending: '*future-oriented attending*' and '*analytic attending*' (Jones & Boltz 1989, Drake, Jones & Baruch 2000). In this model, the 'future-oriented attending' is on highly coherent events. These are defined by the two levels of entrainment:

The first level of entrainment, the 'future-oriented attending', is based on non-human-performed elements such as the click track, facilitating a shift in attention to longer time spans such as the overall tempo. The second level of attention is based on the elements fixed to the grid or the synchronized sound effects, such as the arpeggiated kick and snare, the arpeggiated synthesizers and the audio processing that is synchronized to the grid. This helps the participants to identify performance entrainment errors and to make decisions about whether these are made intentionally or not.

The second mode of attending, 'analytic attending', is based on human-performed elements. This mode of attending tends to occur when the event stimuli are less coherent and more complex, such as where expectations are extremely difficult to formulate. For example, a complex rhythmical pattern on the hi-hats with which musicians might synchronize their own groove.

Fig. 139 shows the entrainment process in the 'Performable Recordings Model'.

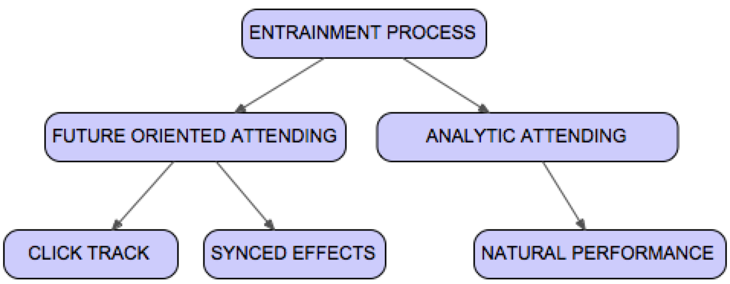


Figure 139: MORALIS, C. (2017) 'Stems – Entrainment Process 1' [photograph] (Designed with VUE software)

Jones (Jones & Boltz 1989) argues that humans have an initial bias to entrain to simple, coherent rhythmical agents. This requires a coherent timing agent known as the 'referent time level', which serves as a temporal agent to entrain the listener with the speech or the musical event. When people lock themselves into a hierarchical rhythmic context, then they can selectively shift the focus to different elements. In a musical context, these would include keeping track of temporal structures from the smallest sub-divisions through bars, phrases, sections and songs to the entire set's duration.

In this project, the future-oriented attending comes before the analytic attending. Also, participants use the second level of entrainment to define the participatory discrepancies through which they can adjust the rhythmical behaviour and character of the song. See Fig. 140.

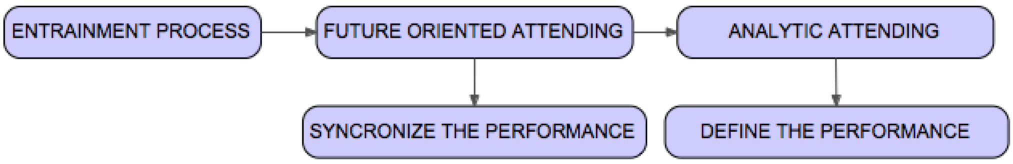


Figure 140: MORALIS, C. (2017) 'Stems – Entrainment Process 2' [photograph] (Designed with VUE software)

'The term "participatory discrepancy" is carefully chosen, as Keil demonstrates ([1987] 1994), to suggest both that musicking involves a sense of participation (referencing Levy-Bruhl and Barfield), and that participation is founded not on exact synchronization but on appropriate degrees of being "out-of-time". According to Keil, discrepancies – particularly in timing – are what create "groove", or activation of positive feel in the music.'

(Clayton, M., Sager, R., Will, U., 2004)

In addition, Clayton, Sager and Will (2004) suggest that *'Entrainment research within ethnomusicology relies upon the integration of musical, cognitive, and cultural theory, thereby allowing a broader description of how musical experience, while individually unique in every case, is nevertheless always social.'*

Having said that, all four participants use a multi-temporal level of attention to entrainment perceiving each sound response and activity differently. For the drummer, being used to play along with a click, this is a natural process:

'I don't really focus on it, I kind of just pretend it's just me. I don't focus on the click because I've played with a click for so long... It's like walking; you don't concentrate on putting one foot in front of the other, you just look at where you're going.'

(Asquini, L. 2017)

For the keyboardist, the entrainment process is based on self-experiences and memories. The focus is mainly on the click track but also, rather than focusing only on the click, Tsoubris (2017) suggests:

'Similar to me, approaching the keyboard sound in my left hand as a sound with decay rather than an arpeggiated quantized sound which would make my life a nightmare.'
'I do consider it as a mechanical process because I know that the drummer has to perform in a particular way. However, during the performance, I just hear the kick in the right place, not thinking about the practicalities.'

(Tsoubris, K., 2017).

For the DJ, the focus is solely on the click track concerning synchronization and he changes his attention to what the other members of the band are doing only to add different musical contexts to his performance.

'I focus on the click and on what the others are doing musically.'

(Skoutelis, P. 2017)

For me, since I am the creator of this model and I can identify every single aspect of this:

'I tend to focus on the click and on the audio processes rather than what the other members of the band do. I bring visual cues to my mind to help me focus on the entrainment process. I only change my focus to confirm that we are all on the same page.'

(Moralis, C. 2017)

Speech and gestures are apparently strongly coupled in adults. Coupling two different elements invokes the concept of oscillators, as Iverson and Thelen (1999) suggest. They propose that speech and hand gestures are coupled from birth. It has been a longstanding debate whether or not multiple entrainment processes in humans are governed by a central clock or through embodied perception such as the feeling of how long a gesture takes to perform. Ivry and Richardson (2001) suggest that every human motor action is controlled by an independent timer, but the evidence for a biological clock is, as yet, inconclusive. Since I have to cope with different latency times for the voice and the guitar, I tend to focus on the overall tempo by adjusting these two performance processes differently from time to time. That might suggest that my timing is based on the two different gestural feelings but could also be based on divergence from a biological clock.

9.2 Score / Notation

The art of musical expression depends on human variation, personal style and, as Kyle (1987) suggests, on the 'imperfections'. In addition, human spontaneity defines live human performance. Bringing all these expressive and entraining elements on to a score makes it difficult to establish a universal language between the members of the band. However, a model based on music notation has been adopted to act as an aide memoire for the performers in respect of emotional expression and timing. This includes the entrainment process regarding midi quantization as well as the performance of the musical effects.

Different colours have been used to achieve a score showing both the different notes with their timings but also indicating which of these are quantized or arpeggiated. The red notes have to be played slightly before the click so that the real-time midi quantizer can place the notes on the grid, while the blue notes indicate that there is no kind of process.

Figs. 141 and 142 show two examples, one is for the drum score and one for the synth score:

Figure 141: MORALIS, C. (2017) 'Drums: American Woman Score' [Screen Shot]

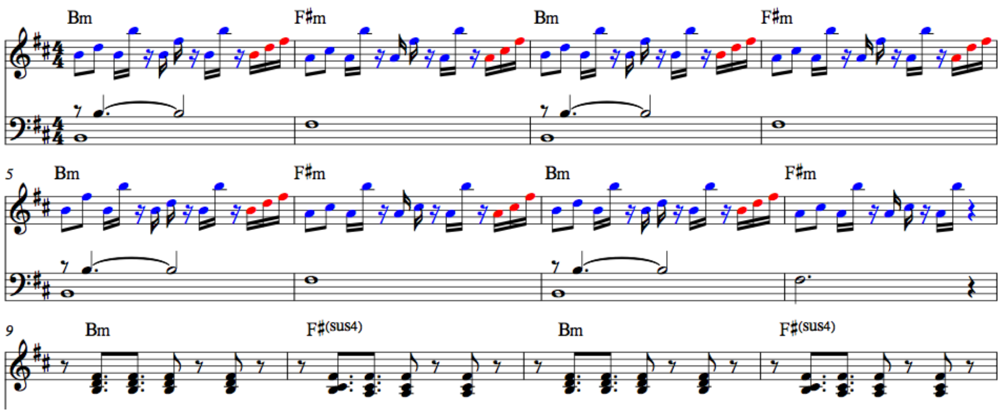


Figure 142: MORALIS, C. (2017) 'Synths: American Woman Score' [Screen Shot]

Below is the audio example of the first 12 bars as shown in the scores above:

- **AMERICAN WOMAN (Drums & Synths for 12 bars):** Audio Example 69

However, the drummer's and keyboard player's colouring schemes are not the same. The drummer likes to see only the quantized notes and, when it comes to parts that are naturally performed, just breaks and long bars. Although the cymbals, including the hi-hats, are naturally performed, the drummer does not want them in blue. This is because the cymbals are always naturally performed and there is no reason for an additional indication. For the keyboard player there is a similar process but, in this case, the arpeggiated notes are left in black as most of the parts are arpeggiated.

For the DJ, since most of the midi notes are played but reproduced by the digital work station, there was no significant reason to indicate quantized or arpeggiated sounds at that point. However, during the negotiation for the creation of the project and his part, the indication of the filters or any other real-time performances of the effects was necessary. Fig. 143 shows a typical example of closing filters.

Figure 143: MORALIS, C. (2017) 'DJ: American Woman Score' [Screen Shot]

The note B indicates the position of the sound on the keyboard while the arrow indicates the movement of the filter from the high frequencies down to the lowest ones. However, the control of this sampler was later transferred on to a non-keyboard controller, so the pitch of the notes became irrelevant, and this score was used only to indicate the different processes and their position in the song.

For me, since I remember the musical context of the songs, there was no need to write scores. However, there were a lot of audio processes, such as volume shaping, that I could not remember, forcing me to create the following type of score. Fig. 144 shows the final volume process of the guitar.

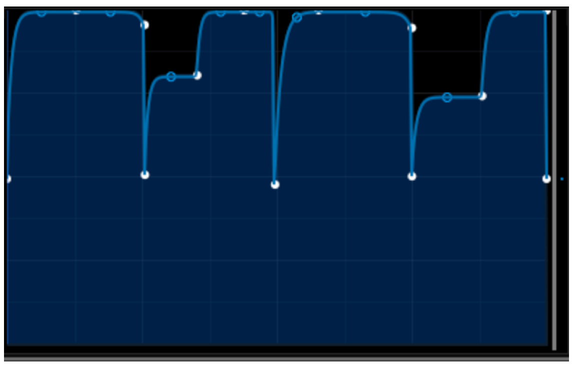


Figure 144: MORALIS, C. (2017) 'Guitar: American Woman – Volume LFO' [Screen Shot]

As this mostly affects the overall timing perception, I place it as a picture above my score. See Fig. 145.

Figure 145: MORALIS, C. (2017) 'Guitar: American Woman – Volume LFO' [Screen Shot]

Below is the audio example of all the instruments as they are shown in the scores above:

- **AMERICAN WOMAN:** [Audio Example 70]

The different cognitive approaches that the band members apply are based on their cultural differences. The visual cues used in these scores are considered to be indicators of focus on different attentional energies. However, to 'catch' upcoming events on time within the temporal frame, musicians would like to process less information when it comes to score reading, and more when it comes to expressivity. This helps synchronization to happen through a quick verification process of the correctness of our expectations.

More specifically, the cognition process is based upon Peirce's triadic model of semiotics. See Fig. 146.

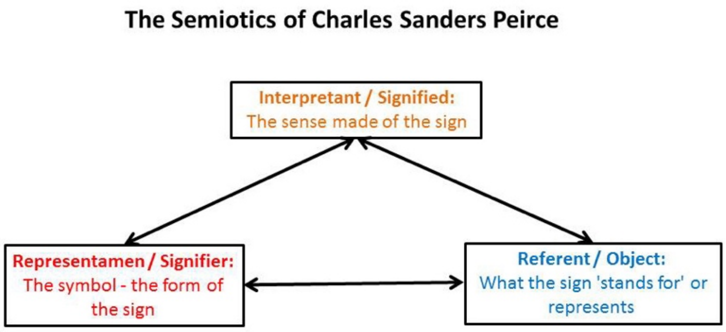


Figure 146: The Semiotics of Charles Sanders Peirce (DecodeScience.com, 2017)

However, to explain this cognition process, the following diagram is applied. See Fig. 147.

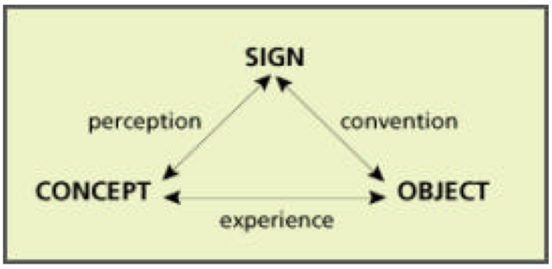


Figure 147: The Semiotics of Charles Sanders Peirce (Evreinova, T., 2017)

Although the nature of the sign and its referent meaning is shared – both regarding the historically established meaning of notation and the definition of the colour coding that has been mutually agreed between us for the purposes of this project – each of us has a slightly different way of interpreting that meaning. For example, the drummer thinks of it as a sixty-fourth or thirty-second note before the grid, while the keyboard player thinks of it as a sound with a slow attack and adapts his performance. The object, in this case, is the musical context of the music note regarding the musical genre and its rhythmical character.

However, the way they perceive the quality of their entrainment process is based on Jones's (2004) three primary stages:

Regarding the mechanics of entrainment in human cognition, Jones theorizes that there are three primary stages: (1) perception, which primes the listener to form expectations; if expectations are met, (2) synchronization; and if expectations are not met, (3) adjustment or assimilation.

(Clayton, M., Sager, R., Will, U., 2004)

9.3 Creativity

There are three key ways in which creativity has been studied and characterized: by people, by product, and by process. According to Williamon A., Thompson S., Lisboa T., and Wiffen C. (2006), current discourse on creativity – from anecdotal accounts to systematic investigations – often conflates three quite distinct concepts: (1) 'creativity' as a component of human cognition and psychological functioning; (2) 'originality' as the probability that a thought, behaviour or product has not occurred previously; and (3) 'value' as determined by the society that witnesses the thought, behaviour or product.

The domain of performance where creativity is most conspicuously present is improvisation. The ethnomusicologist John Baily writes that improvisation *'implies intentionality, setting out to create something new in each performance, "composition in real time" as it is sometimes described.' (Baily, 1999: 208)*.

As Clarke E. suggests (2005), *'Novelty and uniqueness, which Reber (…) takes as defining attributes of creativity, are central to that powerful Romantic notion of creativity which still dominates our culture – creativity portrayed as the mysterious appearance of the radically new, apparently from nowhere.'*

The following nine headings are drawn from Czikzsentmihalyi's (1996) 'Flow of Creativity' model, through which the conditions of creativity in the Performable Recordings model can be analyzed:

- **<u>There are clear goals every step of the way</u>**

The main goal is to create an expressive performance and avoid mistakes that could affect the overall production and performance model. All participants responded that the Performable Recordings model was easy to understand as they are all also music producers. This means that they can understand the technical aspects more easily than someone else who is, for example, just an instrumentalist. Given this understanding of the overall project goals, they could also understand the smaller task-related goals that were generated, and which required them to alter their normal performance practices.

- **<u>There is immediate feedback to one's actions</u>**

Every member of the band can define and evaluate the meaning of the performance through his own actions. Body entrainment, hand gestures and shared feelings can affect musicians' overall performance. Although the nature of the feedback to their natural performance gestures was often altered by the technology involved, they were all prepared to learn to work with and creatively respond to these novel forms of feedback.

- **<u>There is a balance between challenges and skills</u>**

Once again, the success of the project required the participants to 'accept the challenge' of these activities and view the process as one of working creatively with the technology rather than of battling against it. Working with the timing consistency processes such as arpeggiators, synchronized volume shaping and real-time quantization, along with the naturally performed elements, poses new challenges that require a high level of skill to overcome.

- **<u>Action and awareness are merged</u>**

In addition to accepting the challenges and possessing the right skills, the musicians need to internalize the changes they have had to make to their regular practice. They have learned to be able to alter their performance instantly in response to the slightly altered musical goals that they have adopted. They have moved beyond having to think about the technical aspects while they perform and have absorbed these new skills into their *habitus* of creative practice.

- **<u>Distractions are excluded from consciousness</u>**

As for all human beings, when it comes to specific work and effort, concentration and focus play a significant role. The monitoring setup of this model helps maintain focus and avoid any external distracting factors. A key feature in the success of this project has been that the musicians had to move beyond the perception of these new forms of haptic feedback from their instruments as a distraction.

- **<u>There is no worry of failure</u>**

Having designed the model in such a way helps the performers to avoid unnecessary mistakes, but this is also predicated on them becoming sufficiently familiarized with these altered modes of performance. This was achieved by maintaining timing and timbral consistency through complicated processes, as explained previously, but also with simple things such as automating the output volume and sometimes completely muting the sound. One example is the arpeggiation on the drum kick and snare. As Asquini (2017) suggests, it '*is like cruising*.' This means the performer feels less stressed as there is minimal chance of failure.

- **<u>Self-consciousness disappears</u>**

The band members commented that when the technical aspect of the project works perfectly, and when there are no mistakes in the performance, they forget the technicalities. Obviously, the 'when there are no mistakes' aspect of this is about them becoming comfortable with these new challenges and skills. Csikszentmihalyi's notion of flow requires the technical skills required for a task to become subconscious, and this is the moment when the performers realize that they can give that 'extra' element that will enrich the meaning of their performance.

- **<u>The sense of time becomes distorted</u>**

As the state of 'flow' is in large part driven by subconscious cognitive processes, the notion of time, which is conscious, becomes distorted. Also, this process is joyful and very exciting for the participants. Most of the time they are not conscious of time passing because, as Meadows (2013) suggests, one's perspective during joyful emotions is timeless.

- **<u>The activity becomes autotelic</u>**

The joy and satisfaction of successful musical activity is for all musicians an end in itself. The key word in the previous sentence, though, is 'successful', and that relates to the musicians involved feeling that they are in control and acting expertly. The Performable Recordings musicians have had to adapt their goals and skills in ways that provide them with both a sense of first-person authenticity as skilful and expressive musicians and third-person authenticity as creators of music that is true to the musical style of EDM. Once they feel they have a working balance between these two forms of authenticity, they can achieve that sense of satisfaction that makes the activity autotelic.

9.4 Improvisation

So, can the musicians improvise and express their ideas as much as they would like to?

- **DJ**

According to the DJ, the way a song is arranged in real time is a significant factor in a live performance. As an Ableton live performer, he is used to switching through different 'scenes' in real time, hence different sections of the arrangement. Besides, his parts are not tight to the arrangement as he can freely switch between different sounds and patterns. However, it is important to mention the different point of view that this member of the band has when it comes to improvisation and keyboard. He thinks that the keyboard player is also not free to alter his performance, but this is related to the real-time quantization process that may limit his timing variations.

- **Synths**

The keyboard player perceives improvisation as the enrichment of the melodies, rather than as adding different melodies or even arranging the song differently. However, Tsoubris (2017) feels that the arrangement constrains him: *'You cannot improvise much.'* This comes from the limited freedom of playing various and spontaneous melodies. However, he thinks this is more psychological – a statement that mirrors the aspect of 'flow' whereby the musician has to alter their goals and embrace the challenge positively – as, in any case, he would not play much of the song differently from what is written in the score.

- **Drums**

The drummer's opinion is much the same as that of the other two. Although he does not improvise much in a live performance, as he is forced to follow the exact structure of the arrangement, this gives a result of feeling constrained.

- **Voice & Guitar**

Regarding the guitar and the voice, I think that this project allows me to improvise as much as I would like to. This means I do not consider arranging in real time to be 'improvisation', but rather 'conducting a performance'. For me, improvisation is the enrichment of my expressiveness through variances in timbre and melody. I would arrange the song differently in real time only for the purposes of engagement with the audience during a live performance.

CHAPTER 10

10. LIVENESS

This chapter contributes to the broader discussion upon the perception of liveness on contemporary mediatized performances. Based on Moore's (2002) tripartition of authenticates and Sanden's conceptual filters of liveness, an analysis has been made to understand better the aspects of the Performable Recordings model.

Having said that, the conceptual filters that Sanden (2013, p.31) suggests are temporality, spatial proximity, fidelity, spontaneity, interactivity, and virtuality. As Sanden (2013) argues, *'Liveness is lived.'* However, it is essential to consider by whom and under which circumstances. As Sanden (2013, p.32) continues, *'If liveness, then, is a discursive concept, marked not only by its fluidity and complexity but also by its emergence from particular social environments and historical moments for particular ideological purposes, it must be examined with these factors in mind'.*

10.1 Temporal and spatial liveness

As Sanden (2016, p.33) suggests, *'when we speak of witnessing a live performance we mean that we have witnessed a performance at the time of its occurrence (temporal liveness) and in the physical presence of the performer(s) (spatial liveness).'*

Since this project is not tied to a specific natural environment, both 'temporal' and 'spatial' can be true to this concept of liveness. The latest technological advances allow this type of live performance to occur in various situations, such as a new type of live event on the internet and other means of dissemination. This means that the concept of 'musicking' can be found in different situations rather than the traditional physical co-existence of the band and the audience in a specific environment.

The first-person authenticity remains the same because the band will be performing in the same way despite the natural environment, while the third-person authenticity can be evaluated every time from the type of engagement with the band and the visual or aural limitations that may occur.

10.2 Liveness of fidelity

Sanden (2013) suggests that people are associated with the *'real'* and *'authentic'* while mechanical processes are associated with '*corrupted*' products. Of course, the concept of corruption about any given mechanical process is a product of the musical culture or tradition in which it is found. The violin and the piano both involve complex mechanical technologies. A jazz singer's microphone in a nightclub is a form of electronic mediation of their voice. And yet, as Sanden (2013, p.35) says, '*A certain appreciation of musical liveness, then, stems from the perception that a musical performance is unaltered by electronic mediation.'*

As discussed, the two forms of authenticity that are most salient in this process of 'musicking' are the first and the third person as described in Moore's (2002) model. The first-person authenticity relates to the extent to which the participants feel that the performers engage in authentic human expression through their performance. The third-person authenticity relates to the participants' assessment of what constitutes an authentic sonic example of a musical tradition or genre – in this case, EDM. In addition to what it should sound like, third-person authenticity is also concerned with the appropriate 'tools' that should be used and factors such as the coherence between aural and visual, employment of skill, performativity and the constant awareness of a 'standard of achievement'.

In this case, the mechanical processes of studio production are such integral tools to the making of EDM that, rather than it being a question of whether it is appropriate or authentic to use them, the question is whether it is possible to make something that would be considered authentic EDM without these tools. All musical styles require musicians to fashion their ideas of first-person authenticity around the third-person authenticity of the styles in which they work. When they move between styles or traditions, they have to alter the form and extent of the expressive practices that they utilize. The mechanical processes, in this instance, work as an extension to the physical gestures of the performers and they have to establish a balance between the third-person authenticity of a piece of music that sounds like EDM and the first-person authenticity of their expressive identity as performers.

Furthermore, although the Performable Recordings model is not seeking to transfer the sound of an existing recording to this type of production, it is clear that the audio produced should meet the traditions of this musical style. Furthermore, as this group of musicians have rehearsed and performed together, they have gradually built up a template in their minds of what the 'right' version sounds like – a kind of conceptual 'original' that is informed by both the genre or tradition the song belongs to and the specifics of this particular song.

10.3 Liveness of spontaneity

The degree of spontaneity involved during a live performance is a factor that amplifies the perception of liveness, although it is only the potential for spontaneity that is a defining characteristic of liveness. The absence of spontaneity doesn't mean the absence of liveness and the presence of spontaneity doesn't make an event 'more live'. When spontaneity does happen, though, it is evidence of liveness. Improvisation is often used as a measurement for the degree of spontaneity. Sanden (2013, p.37) explains the conditions of spontaneity by suggesting that *'a performer's skill level does indeed measure up to the challenge posed by a particular composition, or to a level of improvisation expected by a demanding fan base.'* If this is not happening, it can result in disappointment but, of course, this does depend on the musical context. Despite the expectations about consistency that third-person authenticity in respect to EDM engenders, the third-person authenticity associated with a live performance of popular music as a generalized cultural event produces a parallel set of expectations for authentic performances that involve the employment of musical skills.

According to Sanden (2013), Aaron Copland suggests that improvisation is linked with the small variations in the performance, different '*nuances'* (Gould, 1966: 47), as well as '*those awful and degrading and humanly damaging uncertainties which the concert brings with it*' (Gould and McClure 1968). According to Johnson (2010), 'live' in a performance is 'the lack of a second chance.' It is clear that the spontaneity and imperfectness of human nature – and hence of human performance – are indicators of 'liveness'. This ties in with the first-person authenticity where performers seek momentary expressive variations in their live performance.

10.4 Corporeal liveness

Taking into consideration that 'corporeal' is based on the interaction with the environment and not only with what our mind thinks of the environment, as Merleau-Ponty (1964) suggests, then '*not only perception but also expression is rooted in this corporeality'* (Sanden 2013). Based on Shove and Repp's (1995) discussion on performance as the result of the physical movement, then performance is experienced through an embodiment process.

Again, this project meets this concept of liveness as already explained in section 13.2: the mediatization is not limiting the physical gesture. In fact, the embodiment process of the various expressive nuances expressed by the performers helps the mechanical process, whether this is a particular movement of the singer's head to avoid any microphone 'pops' or the movement of the body to help the entrainment process.

10.5 Virtual liveness

As Sanden (2013, p.43), explains, '*I reserve virtual liveness for discussions of performance contexts in which liveness and mediatization are both performed with great emphasis.'* In the Performable Recordings model, this coexistence of mediatization and human performance amplifies the perception of liveness, as the mechanical process is masking the natural activity. This aligns with Bown, Bell and Parkinson's (2006) findings that '*Mediatization, may, in fact, amplify perceptions of liveness.'* From this point of view, the Performable Recordings model acts as the bridge between the aesthetics of a studio-produced song and of a live performance and not as a third-party interfering process.

CHAPTER 11

11. POST EVALUATION OF PRACTICE

The methodology for creating the artefact has been focused initially on each band member individually since every instrument needed a different production and performance approach. It was not a linear procedure because it was essential to receive feedback from the performers to finalize the audio processes. However, the group performance later determined the final tuning of the Performable Recordings model.

Fig. 148 shows the designing process followed for the creation of the Performable Recordings model:

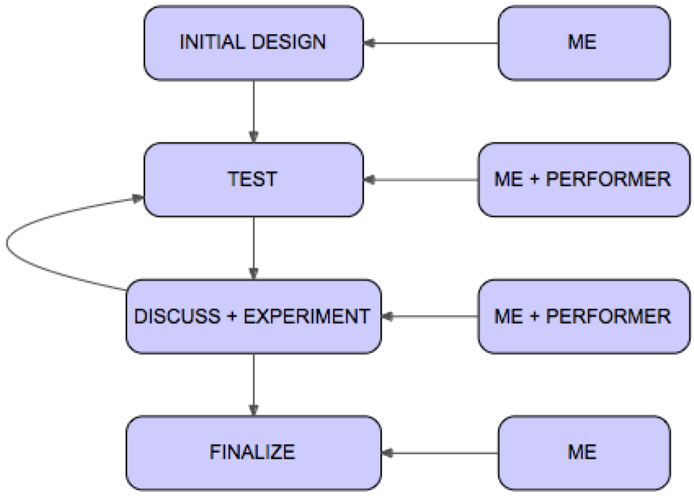


Figure 148: MORALIS, C. (2018) 'Design Process Plan' [photograph] (Designed with VUE software)

This has been achieved mostly through informal face-to-face conversations after the tests or rehearsals. Social media such as Facebook was also used to discuss further tweaks of the process settings. The band members' responses were helpful in understanding that this creative process should be based on the three main factors: first, on the performers' needs and point of view, taste, satisfaction and ability to perform; second, on my aesthetics and the aesthetics of electronic music; and third, on the system's technical aspects and limitations. See Fig. 149.

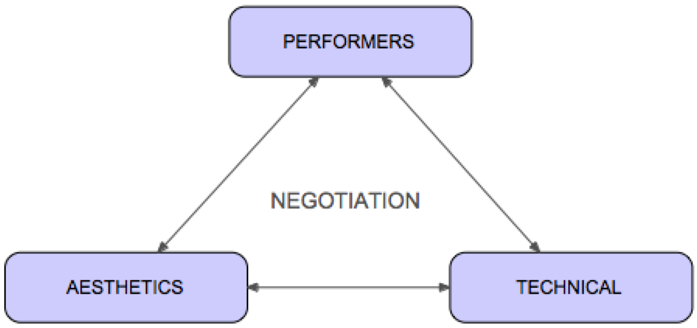


Figure 149: MORALIS, C. (2018) 'Negotiation' [photograph] (Designed with VUE software)

11.1 Initial design

It was necessary to consider from the beginning the final mastering attributes I was looking for that could be affected by stereo-to-mono compatibility, phasing issues, loudness levels, and frequency and dynamic response along with timing consistency similar to the tradition and style of EDM. Additionally, taking into consideration that some of the real-time processes need a large amount of CPU power, and the available equipment and technology for this research, led me to think more creatively and work around traditional mixing processes and approaches.

For example, modulating the attack and release times, according to velocity, allowed me to reproduce such sound attributes from only one audio sample to match the expressivity of human performance as explained in chapter 5. This approach also led to less or no compression and better control of the dynamics and loudness levels, allowing me to think more creatively regarding the musical content and to worry less about the audio processes.

Another aspect of the real-time process that made me think creatively is the amount of latency introduced. For example, I had to overcome the latency issue from excessive equalization, particularly real-time pitch tracking equalization, which requires a considerable buffer size. I came up with the idea of a two-part audio sample, as explained in chapter 5, to match the tonal response of the musical content with the frequency balance of the mix.

Also, the CPU power and latency limitations forced me to select specific equipment that helped me reduce the amount of 'fixing' required in my audio signals. These were the digital wireless systems, the microphone and guitar emulations embedded on the hardware, and the audio-to-midi real-time converters, regarding triggers, synthesized and sampled sounds. Additionally, reducing the preparation time of the sound of my tracks resulted in moving the centre of my focus more into the musical aspect of this project, such as the melodies and the performances.

However, researching also into existing technologies such as midi quantization and LFO tools inspired me to figure out ways to combine the studio aesthetics with the live performance and realize this research. The appropriate sound design, production processes and specific performance techniques of every individual instrument were based initially on my vision and the aesthetics of this model. As a musician who already plays these instruments, I could clearly see from the beginning that the final production and performance practices should serve both first- and third-person authenticities. This means that performing the instrument should feel natural through this extensive use of mediation, while at the same time the technical aspects should not affect the performance much and, most importantly, should serve the aesthetics and the sound attributes that define the genre of electronic music.

11.2 Tests and experiments

After the initial design of each instrument according to my point of view as a musician, testing this process with the other members of the band did not reveal significant incongruences between the way I would perform and the way they would like to perform through this technology. It was more a matter of adjusting the existing technology and its settings to their abilities and point of view.

Testing the midi quantization technology with the drummer, we found that performing before the click was not always possible. Although he sometimes managed to play before the click, his ability to maintain this kind of performance was affected by the musical content, groove and the tempo of the song. In simpler patterns, where the timing was limited to eighth notes, the drummer managed to perform them; however, when a sixteenth-note division was used, playing ahead of the beat was not consistent.

As Pearce and Rohrmeier (2012) argue, *'Music perception involves the cognition of complex and parallel temporal processes that combine local and hierarchical structures at multiple levels of organization (cf. Koelsch, 2010; Levitin & Tirovolas, 2009) according to the syntax of a style.'* Setting the midi quantization to sixty-fourths, the drummer managed to perform better for two reasons: first, he was paying more attention because of the timing division analysis that was higher; and second, he was feeling more confident since the mistakes were not so obvious.

However, the drummer argues that playing slightly before the click within a previous sixty-fourth note is not possible to be achieved because a sixty-fourth note is the smallest division that he can play (Asquini, L., 2015). Although the drummer has never been asked to play faster than a sixty-fourth note, but only to play ahead of the click, this suggests that the timing perception is based on cultural differences between the way I conceptualize this performance as a producer and guitar player and the way the drummer is used to performing.

Furthermore, always playing ahead of the click made the drummer try harder hitting constantly at 127 velocity. It was necessary to combine partial real-time quantization and arpeggiation to avoid the discrepancy between stimulus and action. Changing the production approach from midi quantization to arpeggiation affected the performance approach dramatically. When Asquini, L. (2015) tried this performance technique, he mentioned, *'I feel like I am directing.'* Although the kick and snare are arpeggiated, he found he still needed to perform naturally to vary their timbre. Without having to focus on the 'before the click' performance, but instead performing naturally, the drummer could then vary the timing of other elements for expressive purposes. For example, the hi-hats could be slightly delayed or speeded up, giving a different meaning to his performance every time. As he later mentioned (Asquini, L. 2015), drummers tend to rush on fills, and this will help to land earlier, before the click, correctly triggering the midi quantizer and thus the arpeggiator.

According to Zagorski-Thomas (2014, p.47), *'no matter how good the audio quality of a recording is, if the other modes of my perception are telling me that I am not in the presence of musical performers I will recognize it as a representation rather than as the "real thing".'* In this case, the drummer suggests that this partial quantization performance '*feels more natural'* and the only thing he needs to know is where the quantization is enabled or disabled.

Testing the midi quantization with the keyboard player, Tsoubris K. (2016), said that *'many times keyboard players play with slow attack sounds and the way for performing is thinking a sound with a slow attack rather than trying to play before the click.'* Here again are shown the different perceptions regarding timing that musicians have according to the instrument they play. While the drummer tries to play according to the notes written on the score, the keyboard player tries to match his performance to the instrument's sound and the way the sound behaves. In this case, the only adjustments done to the process were on tweaking different settings like setting the midi quantizer from sixteenth notes to eighth notes or vice versa.

However, apart from the midi quantization, another process that helps define timing consistency was the automated volume envelope curves. None of the participants ever felt that this process affected their performance, and in some cases they did not even recognize them. When the keyboard player (Tsoubris, 2016) was asked if he thought the lead synthesizer was quantized, he responded that he thought not, but that it sounded like a studio-produced performance. The volume curve approach for timing consistency on those two instruments did not affect their performance, so I did not make any changes. However, in the case of the guitar and the voice, and the volume curve applied on them, I had to change some of them or even mute some parts acting in the same way as the other two participants regarding midi quantization.

As with the other two participants, I was selecting the settings according to what I could play more comfortably, not necessarily more easily. For example, I had to compose simpler guitar riffs that allowed me to follow the volume curves accurately, not necessarily because the guitar parts would be difficult. However, the ability to perform in all cases had to be familiar, fun and comfortable without diminishing the performer's agency by making it too simple. Therefore, the performers' needs took priority as this is the key to a more accurate and mistake-free live human performance. As Asquini (2017) mentioned, *'I think all the changes have been done to make it work, not necessarily because I couldn't perform. All served to adapt to the way I play drums rather than serve the computer'.*

Regarding the DJ's part, everything seemed to be pretty straightforward for him, and the only aspect that I had to add was some extra filters to allow him to manipulate the different stems of the band – voice, guitar, drums, synths – in a similar way to how he shapes the songs he plays in clubs. In this case, I had to adjust his part to what a DJ does, rather than to what a percussionist or a laptop performer does.

11.3 Negotiation

Communicating my ideas with the band members has been done in a very constructive and inspiring way. However, there were times when I had to negotiate the selected processes further, because on the one hand the performer could not perform, and on the other I could not change the settings. An example of this is the keyboard riff in the song 'Superlove'. The keyboard player felt that it was too difficult to keep playing along with a real-time midi quantizer, much as the drummer did when he was introduced to this technology. In this case, the alternative option would be to create an arpeggiated, more simplified, version similar to the approach followed on the keyboard riff in the song 'Changed the Way You Kissed Me'. However, since the keyboard player wanted to be able to enrich the riff with different notes or play it slightly differently, he accepted the difficulty of the initial design approach. After a couple of rehearsals, he was able to perform the riff without any significant difficulties, but it never became the ideal production approach for him since it did not have the ideal balance between diminishing the performer's agency by making it too simple and augmenting the performer's agency by making it too complicated and challenging.

Consequently, the negotiation did not only point out how the musicians have had to utilize extended and altered performance techniques to participate in this project, but also revealed that they had eventually to embrace them as extensions of their creative practice rather than obstacles that have to be overcome.

For example, the way the musicians adapted to the altered entrainment process that real-time quantization and envelope shaping required, included not only future-oriented or analytic attending but also developing cognitive strategies for playing ahead of or after the beat. These strategies – different colouring schemes in notation, information about hardware control such as the movement of filters, and pictures demonstrating the timing-related sound processes – are based on the performers' cultural differences, previous experiences and their physical limitations. Additionally, the mechanical processes, in this instance, worked as an extension to the physical gestures of the performers and they had to establish a balance between the third-person authenticity of a piece of music that sounds like EDM and the first-person authenticity of their expressive identity as performers. The embodiment process of the various expressive gestures may have helped the mechanical processes and amplified the perception of liveness.

The way that each of the musicians contributed to the specific customization of the form of notation that they use has been based on Peirce's triadic model of semiotics (sign, concept, object) and their personal interpretation of it. However, all of the musicians' entrainment processes have also been based on Jones's (1989:466) three primary stages of the mechanics of entrainment in human cognition: '(1) perception, which primes the listener to form expectations; if expectations are met, (2) synchronization; and if expectations are not met, (3) adjustment or assimilation.' The notation did not change the way the band performs but contributed to a better understanding and embracing of the altered performance practices.

The negotiation also revealed a cultural difference in what improvisation means to each of the performers. Baily (1999:208) suggests that it is 'composition in real time'. However, what 'composition in real time' means to these band members is different from one to another. According to their point of view, improvisation can be achieved through variation in the parts of the song. These are timing variation and melodic variation or enrichment, but also include variation in the sound attributes. More specifically, it involves the ability to create different small nuances in timbre, timing, pitch, and dynamics as well as the potential to enrich the musical context without changing the structure.

The achievement of a state of flow under Csikszentmihalyi's model required the musicians to invest in the idea and go further than simply learning how to perform in these circumstances, but also to 'buy into' the project by accepting the challenges that have been set. They may want to perform with more expressive control in other circumstances, but they have accepted the limitations that our assessment of what constitutes third-person authenticity in EDM means for their own idea of first-person authenticity. In addition, they have accepted that the proposed technical solutions to the problem produce an authentic-sounding musical product.

11.4 From rehearsals in the studio to live performance on stage

Bringing this technology on to the stage and having the opportunity to test it in real-life conditions revealed some other aspects that initially could not be identified. Apart from some technical aspects such as power distribution, issues with the signal of the wireless systems, magnetic field issues and feedback issues, there was the need to balance the overall sound according to the venue's sound systems and acoustics. In this case, all of these problems are typical in the professional world and may happen from time to time.

However, the most critical difference between the studio and the live lies in the preparation of this project, both in the production of it and in the people who will deliver a live performance through it. Having as an example a live performance on 25th April 2018 in London, the short notice and the addition of an extra member to the band challenged the design and delivery of this production model.

The keyboard player could not participate, and his part was added as a backing track. There was no keyboard player on stage, so, to avoid the discrepancy between the visual and the aural, his part was enriched with additional elements such as extra backing vocals and other sounds. This amplified the perception of liveness; it was easy to compare the live with the pre-produced as no one we asked from the audience felt that we were miming or that the keyboard sound on the backing track was part of what we were doing on stage. It was a clear distinction between live-performed and pre-recorded. This proves again what Bown, Bell and Parkinson (2006) suggest that 'Liveness and mediatization can co-occur… Mediatization, may, in fact, amplify perceptions of liveness.'

The short notice and the extra member of the band forced me to simplify some aspects of this process. However, there was a negotiation between what should be simplified and how much. Since the new member of the band, an additional singer, was not familiar with the process, I had to explain in detail and rehearse a few times in the studio. However, some mechanical processes, such as the auto-tuning and the volume envelope shaping, could not be included in the same way as been explained in this thesis because the performer needs a certain period of training on these. To overcome this issue, I minimized the auto-tuning process at a level where it was serving the process and completely removed the volume envelope shaping. This resulted in a version with a less 'studio' aesthetic but helped the performer to deliver. Since I had to change the production approach on her part, I also changed it on to mine to maintain the same perceptual cognition of what we were doing on stage and not to stimulate comparisons of how much processing there was on my part or hers. The processes were identical. This also relies on the fact of what Bown, Bell and

Parkinson (2006) suggest as avoiding the identification of live performances and mechanical processes.

The short notice of the live performance and the new song list also proved that creating scores with the notation, as suggested in this research, helps the participants to learn the songs quickly. However, when it came to the lyrics, both singers had to use a monitor screen to read them. As in all other types of live performance, that created a barrier between the singer and the audience since we had to focus most of the time on the screen and not on the audience.

Another aspect of the live performance was the creative aspect. All the band members enriched their performances either by adding additional notes or being more expressive in a similar way to that explained in section 9.3 of this research. The connection with the audience amplified their employment of musical skills and proved that a generalized cultural event produces a parallel set of expectations for authentic performances. However, apart from the performance aspect, as Skoutelis (2018) suggested, some of the priorities may change in a live performance. To overcome a technical issue with synchronization between laptops or loss of sound, the DJ also acts as a sound engineer on stage. For this reason, being responsible for solving any technical issues, he focuses primarily on ensuring that there are no problems on stage and then focuses on the musical and performance aspect – something that is the opposite to rehearsals. In the case of loss of synchronization between laptops over the 'link' technology, there are markers on the beginning of every song, since it is a 90-minute medley. If the synchronization fails, the DJ waits for the last bar of the previous song to arrive and then triggers the next marker, which is in the next bar. Since Ableton expects the next bar to come, there are no glitches or skipped parts, and at the same time the project falls into synchronization.

The live performance revealed that the band members need a certain amount of time to prepare and train for this production and performance model. However, none of the participants felt more anxious or thought this process differed from studio to stage.

11.5 Future technologies

In the live performance mentioned above, I had the chance to include two new technologies. The Smart: Eq Live (see Appendix 14), a real-time adaptive equalization tool, has been used to balance the voices better and faster in the mix and also between each other. Another technology used in this case was Auto Tuning for Guitar (see Appendix 15). This is a guitar modelling technology that preserves perfect intonation by auto-tuning each string separately. This technology helped me to maintain perfect intonation for the duration of the live performance; additionally, since it reproduces the sound digitally, there was no alteration of the timbre of the strings over the length of the show.

Both technologies helped the process and did not change it, proving that this research project relies on the concepts suggested and not necessarily on the specific technologies used.

CHAPTER 12

12. CONCLUSION

This research presented a production and performance model in the style of Electronic Dance Music (EDM) or popular electronic music. This project aims to bridge the gap between 'human' and 'non-human' that requires performers to work with technology in new ways, in this musical style, rather than mimicry or using lip-syncing.

The project utilised research on authenticity and its relation to aspects of liveness in this type of performance. The aim was to create a musical process in which all the participants feel that the band is performing authentically while being faithful to the genre or tradition, which is about making sounds that are true to the genre. The key to this is the combination of machine accuracy with some aspects of human expressive performance in a way that maintains the integrity of the popular electronic musical style.

The Performable Recordings model acted as the bridge between the aesthetics of the studio-produced song and the contemporary mediatized live performance. It is:

a type of music production that enables the artist to perform a musical piece live, using, in real-time, the mixing and post-production processes that create the aesthetics of a studio-produced version.

Taking into consideration the fact that sound affects human performance, and, conversely, that human performance affects sound, it is shown that the participants had to adapt their performance practices and the mixing processes had to be suited to the performers' needs. Also, to avoid miming or lip syncing, it was necessary to adapt real-time machine practices rather than pre-record material. Having said that, what performers think a live performance is, and how this can be evaluated and conceptualized, had to be analyzed in relation to first- and third-person authenticity. Despite the expectations about consistency that third-person authenticity in respect to EDM engenders, the third-person authenticity associated with a live performance of popular music as a generalized cultural event produces a parallel set of expectations for authentic performances, as Carlson suggests (2004) through three concepts for evaluating a performance. It is shown that in this project, as in other musical styles and traditions, performers have to alter the form and extent of the expressive practices that they utilize to fashion their ideas of first-person authenticity in order to accommodate third-person authenticity.

This research has not been based on questioning multiple audiences, but only a specific one: the band. However, what 'live' means to this band and in a performance depends solely on the performers' point of view and their cultural background. Although all participants agree that authenticity, creativity, and expressivity in a performance are strong indicators of what 'live' means, their cultural differences may result in their perceiving these terms differently. Therefore, a 'live music performance' was initially defined by the existing theoretical background as a 'unique human performance'. Having said that, a 'live performance' can exist without the presence of an audience and, therefore, any distinction between recorded and live may be irrelevant.

This project has been based on existing hardware and software systems such as midi quantization, LFO tools and modelling, which enabled me to produce the various musical submissions that constitute the substantive part of the submission.

The Ableton sessions and the hardware configurations that have been outlined constitute the embodiment of the most recent stage of the ongoing process of negotiation and experimentation in relation to the two competing forms of authenticity that have been discussed in this thesis.

While this submission is for a DMus so is focused on my own creative practice, the doctoral element of the project lies in the broader concept that underpins it. This is about the issues of performance authenticity arising in musical styles that rely on machine-like accuracy and consistency as part of their musical aesthetic. My work may in one way be very specific, but it also provides a more general model for ways to tailor the use of technologies that enhance accuracy and consistency to the preferences of the performers and specific aesthetics of the musical genre.

12.1 Original contributions and innovative practice

If music consists of ideas and emotions expressed through various combinations of sounds, this production and performance model can contribute to the less explored areas of contemporary mediatized performances and to the understanding of how these technologies can help to expand creativity.

The audio production has been based on a detailed sound designing and mixing process that acts as a combination of mixing and mastering process in real time:

- The division of the drum kit samples into two parts and their combination with the musical content of the song, the ‘hocketing’ approach on the keyboard, the synchronized frequency shaping, the sequenced timing quantization, the synchronized volume shaping and the arpeggiation can act both as mixing tools and performance indicators. This innovative use of existing technologies for a real-time process has a backward effect, suggesting that they can help the studio-based audio productions to introduce the live element in a controlled environment, benefiting from more expressive performances. While the principles and some of the techniques that underpin these processes are not in themselves innovations, the originality lies in the specific way in which they have been combined and the interactive and collaborative process that led to this configuration.

- The combination, in real time, of fixed and varied elements such as timing, pitch, dynamics and timbre can create a live performance with the aesthetics of a studio-produced sound. However, the way these elements are combined depends on the musical context, and timing is the most critical attribute in the definition of what ‘live’ or ‘studio’ means in this context. New bands can bring their own studio sound on stage successfully and, indeed, may introduce a new way of producing songs in one format that can be used for both recorded and live music. The concept of allowing musicians to explore different combinations of these types of fixed and varied elements has been extended in this thesis and, in addition, points the way to further potential explorations.

- This innovative approach to production and performance is not based on large quantities of hardware and allows performers to have the same sound attributes and perform in the same way on every occasion. For example, these bands can perform in venues from small pubs to big arenas with the same technology but also through the internet on various social media or video platforms. The specific instance that has been developed for this project demonstrates a proof of principle that is flexible enough to be adapted for a variety of musical styles that involve elements of machine accuracy and for different musicians’ preferences about flexibility within their performance.

- The negotiation between the research participants, the aesthetics of electronic music and the technical aspect of this model revealed that parameters such as the cognitive process, cultural differences, perception and creativity play a significant role in designing new production and performance practices, while the design itself cannot be a linear process but a cycle of design and feedback. While this, again, is not an innovation in terms of the concept, the specific example provides valuable lessons about the issues that will arise in this sort of negotiated process.

- This research also suggested new methods of notation and score reading for technologies and audio processes. The different colouring scheme and the graphic representation of the audio processes demonstrate ways in which a notation system can be customised to accommodate new technologies.

12.2 Future work

The Performable Recordings model acts as an intermediate agent between technology and humans, achieving authentic human performances with the aesthetics of contemporary studio-produced songs. Waterman (2008) explains that, in his opinion *‘Technology has killed our industry because people aren’t learning the skills to use the computer as a tool, they’re using computers as the whole thing.’* In this research, technology acts as an extension of human performance, coming into contradiction with what Miller (Theverge.com, 2013) suggests, ‘*Computers are where music goes to die.’* Furthermore, Sanden (2013, p.159) suggests*, ‘Liveness is a dynamic and versatile concept. As for performance practices in any cultural form change, shared understandings of what makes those performances live or not live will also change’.*

Having said that, the Performable Recordings model allows performers and their audience to experience this type of live performance as they would experience a traditional form of live music performance and as a type of event where mediatization may eliminate those attributes that define something as ‘live’.

Future work may include:

- Research with different musical genres and more instruments may reveal new production and performances practices, enriching those suggested in this research.
- Research with more participants may deepen our understanding of the cognitive process of new practices, the role of cultural differences, creativity and perception with this extensive use of mediation technology.
- This model should also be tested in a variety of venues from small clubs and pubs to big stadiums and arenas, but also through the internet (Facebook, YouTube), to research into audiences’ responses to this new type of performance.
- The general principles of this research should be tested with new technologies that are introduced in the future to investigate whether the practices suggested here can have a benefit or not.
- Further research should be done into the way that new studio production approaches used on stage can be notated and be part of the original score.

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APPENDICES

Appendix 1

DRUMS (Luigi Asquini)

PERCEPTION

1) **What was your first impression when I talked to you about my research project? (play live and sound like the studio?)**

- Imagined that the whole thing would sound slightly better than the typical live sound production.
- Didn't expect to fix timing or real-time ‘editing.'

2) **How easy was for you to understand how this model works?**

- Being a musician and a producer, it was pretty easy to understand the whole process.
- Wouldn’t be able to build it, but I understand completely how it works.

3) **When you had to learn the songs, what approach did you follow?**

- I had only to pay attention to the parts where the timing was different like there was no quantizer. Generally speaking, I followed the standard way to learn the songs.
- Learns everything like a normal song. Uses scores and tries to play it.

4) **So, I first built this model and then we adjusted it on to your needs and performance, do you think I should have done it differently?**

- I think that having that already set and then slightly adapt it to my needs worked perfectly. Otherwise would be a waste of time.

5) **Do you like the project? Have you ever felt this whole idea would not work for you?**

- Yes, I like it.

6) **When do you most enjoy this project?**

- Some of the break fillers where all the synths come in (drops), and it becomes ‘BIG.'

NEGOTIATION

7) **When I was building this model, I had to exchange my ideas with you. Do you feel we had a constructive communication?**

- Yes, it worked fine.

8) Do you remember which things we had to change to make you perform easier?

- I think all the changes have been done to make it work, not necessarily because I couldn't perform. All served to adapt to the way I play drums rather than help the computer. I only lately requested to change the dynamics of the HI Hats because I wanted first to fix important things and then go to smaller personal requests.

9) What was the most important fix we made?

- The quantizer was a big change (meaning: changing the quantizer to an arpeggiator).

LIVENESS

10) Do you consider your instrument like a real one? (dynamics, timing, timbre) Or do you think through this process became something else?

- Yes, it corresponds to my performance.

11) When you perform, are you thinking every single process that is happening in the background?

- No, I don't think about any of them, and this happened after a couple of times playing a song.
- I may be only thinking about it if something is not working properly.

12) Have you ever felt that you were not playing in reality?

- No

13) The way you monitor this project, headphones and its balance, affects the way you perceive your performance or your instrument?

- I think it sounds cool, but maybe because I am used to playing with very bad monitoring sound. I used to play, and I hope this will sound properly.

14) Do you think this is real live performance?

- Yes.

15) What do you think would be the audience's idea of this project?

- Yes. People think DJs are playing live.

16) If we play on a backing track, do you think they will still believe we are playing live?

- In their majority, yes. If the visual to aural is correct, then most people will not notice it. Especially those who do not know about music.

17) Then why should we play live and not on a backing track?

- I don't know! Well eventually after a while they might understand what we do on stage. But this is mostly true because people can get away with a lot of stuff, i.e., Muse on the Italian TV where they switched instruments. The presenter didn't even notice that the drummer became the singer etc. – they were playing on a backing track. This is because people do not know where to look, for example, they do not know how a chord looks like on a guitar.

18) What happens if the audience consists of musicians?

- Musicians will eventually figure it out.

PERFORMANCE

19) How easy is for you to play with a quantizer?

- The way we use it now, only to certain hits and parts is fine. It just didn't work for me when I had to play every single note before the click.

20) After I explained how this model works, do you feel this changed the way you perform?

- Not at all.

21) How easy is for you to play with an arpeggiator?

- Very easy. I forget it when I play. Maybe I remember it when I have to do a fill break, and there is no quantization or If I want to do some extra things and I am doing them on the HHs which do not have any kind of quantization. When the arpeggiator kicks in is like cruising.

22) How easy is for you to play with a synced volume envelope? (like sidechain compression)

- I didn’t even notice it ever.

23) How you synchronize your performance with the other members of the band?

- Depends. Sometimes on the click, and other times where there is a very rhythmical part, i.e., a synth that plays 16^{th} notes.

24) Do you sync yourself with the arpeggiator on the kick and snare?

- I guess maybe yes. However, everything is subconscious. I cannot tell if I am using only the click, the synth or the arpeggiator to sync. Perhaps all of them.

25) Do you consider the click track important for this project?

- Yes.

26) Can you improvise as much as you would like to?

- I am not normally a huge improviser. Well, there is less freedom for doing maybe some extra fills. What I think is more like an arrangement thing because I am forced to follow the arrangement. So, I do not think is a problem with the project but with the arrangement.

27) Do you feel confident with the project on stage?

- Yes! Technically is less room for error but fewer places where you can actually make a mistake. Because of the arpeggiation, I feel comfortable, and I can focus on other things like fills and the groove on the hi-hat.

28) Scores?

- I am happy with the way you wrote them and put different colours to suggest the arpeggiation and the quantization. Initially, I thought I should have them written as where exactly I have to play but eventually is better to see the score and only indicating with colours the parts that I have to play before or not worry about the timing.

<u>KEYBOARDS</u> (Kostis Tsoubris)

<u>PERCEPTION</u>

29) What was your first impression when I talked to you about my research project? (play live and sound like the studio?)

- I am open-minded, it was interesting. I was wondering if this could be realistic.

30) How easy was for you to understand how this model works?

- It wasn't difficult to understand it because I mostly use arpeggiators and sounds that have a slow attack.

31) When you had to learn the songs, what approach did you follow?

- I did it as usual. Listening and following the score. Maybe I had to pay attention to some arpeggiators. The only difficult thing was to adapt my performance because what I play is not what always what I hear. Especially for classically trained pianists, like me.

32) What if I midi map the same keys on to pads – you are not looking now a keyboard. Do you think that would be easier?

- It might make more sense because everything I play corresponds to what I am hearing but I prefer a keyboard, but I am open to trying the pads.

33) So, I first built this model and then we adjusted it on to your needs and performance, do you think I should have done it differently?

- No.

34) Do you like the project? Have you ever felt this whole idea would not work for you?

- Yes, I have never felt this would not work for me.

35) When do you most enjoy this project?

- It is down to musical related things. It is more about personal preferences about the songs but rather on how I have to perform.

NEGOTIATION

36) When I was building this model, I had to exchange my ideas with you. Do you feel we had a constructive communication?

- Yes.

37) Do you remember which things we had to change to make you perform easier?

- We removed some quantizers, and on other, we changed their timings.

38) What was the most important fix we made?

- The most important thing is to adapt the quantizers according to the way I perceive my performance and how I should play a sound and not how it makes sense to anyone else.

LIVENESS

39) Do you consider your instrument like a real one? (dynamics, timing, timbre) Or do you think through this process became something else?

- Yes, it still feels like a keyboard, but there is a different performing approach. Not having the freedom to play anything I want maybe is something, but it makes sense since we all follow the score.

40) When you perform, are you thinking every single process that is happening in the background?

- No, not really.

41) Have you ever felt that you were not playing in reality?

- No.

42) The way you monitor this project, headphones and its balance, affects the way you perceive your performance or your instrument?

- No, it's fine.

43) Do you think this is real live performance?

- Yes, I think I perform live as a musician since there is space for errors.

44) What do you think would be the audience's idea of this project?

- Depends on their musical understanding, on our performance (no errors – maybe not that much true).

PERFORMANCE

45) How easy is for you to play with a quantizer?

- Depends on the song and the sounds. I don't think it as a quantizer but as a sound with decay – slow attack. So, I have to adapt my performance according to the sound response and not on the technical aspect of the timing.

46) How easy is for you to play with an arpeggiator?

- Very easy because is part of a synth. However, depends on how complicated is.

47) How easy is for you to play with a synced volume envelope? (like sidechain compression)

- Very easy.

48) How you synchronize your performance with the other members of the band?

- I have to follow the click track and not the drummer as I usually do with other bands.

49) Do you consider the click track important for this project?

- Yes – otherwise nothing will be synced properly.

50) Can you improvise as much as you would like to?

- I think you cannot improvise much but since I can add some notes or alter the position of the chords and enrich the melody without changing the meaning of the song I think this improvisation. After all, I am following the score. I may feel like that only because I know that there are certain areas and parts in the song where I can do certain things and not anything random. It's more psychological.

51) Do you feel confident with the project on stage?

- Yes, unless is something is not working properly. I am not worrying about my mistakes but if the computers fail to work properly.

52) Scores (colours)?

- The score it was making sense. I liked the color-coded score. It was clever.

<u>DJ</u> (Panos Skoutelis)

<u>**PERCEPTION**</u>

53) What was your first impression when I talked to you about my research project? (play live and sound like the studio?)

- I thought it was interesting. It would be helpful for people who try to play electronic music live for the first time (from rock to electronic).

54) How easy was for you to understand how this model works?

- Pretty easy, because is my area.

55) When you had to learn the songs, what approach did you follow?

- It was useful to have the midi data there already as a guide. I didn't use the scores to learn the songs because it was not relevant to me.

56) Do you like the project? Have you ever felt this whole idea would not work for you?

- Yes, is very similar to what I am doing with other bands.

57) When do you most enjoy this project?

- I like the ability to play the sounds of other musicians. Adding effects is a very interesting aspect as you do find it in other genres (master effects on other musicians).

<u>**NEGOTIATION**</u>

58) When I was building this model, I had to exchange my ideas with you. Do you feel we had a constructive communication?

- Yes.

59) Do you remember which things we had to change to make you perform easier?

- We did not change anything.

LIVENESS

60) Do you consider your instrument like a real one? (dynamics, timing, timbre) Or do you think through this process became something else?

- Yes, it is, in the same way, I do my live sets.

61) When you perform, are you thinking every single process that is happening in the background?

- I do think about it because I like to understand. But when it comes to live performance, I do not think of any process.

62) Have you ever felt that you were not playing in reality?

- No

63) The way you monitor this project, headphones and its balance, affects the way you perceive your performance or your instrument?

- No, but I think every musician should have more control over the monitor and on every song.

64) Do you think this is real live performance?

- Yes, it is a live performance. I think the live performance concept needs is actualized.

65) What do you think would be the audience's idea of this project?

- I think will look very real as every sound is produced from the musicians on stage. If it was only one person that could not match the audio.

PERFORMANCE

66) After I explained how this model works, do you feel this changed the way you perform?

- No.

67) How easy is for you to play with all these sound processes?

- Is easy because my part is not complicated. I do not play notes I am mostly affecting the timbres of the sounds.

68) How you synchronize your performance with the other members of the band?

- I focus on the click and on what the others are doing musically.

69) Do you consider the click track important for this project?

- Yes.

70) Can you improvise as much as you would like to?

- Yes. I think Kostis should have less quantization. It is very interesting using the arpeggiators etc. but when it comes to live he should have more flexibility to improvise and not to be tight to the arrangement.

71) Do you feel confident with the project on stage?

- Yes, unless there are technicalities.

72) Are you using any scores?

- I am not using them, but it was interesting on how you scored the sound effects.

Appendix 2

Kick

For the creation of the kick sample, initially, the 'Kick Synth' synthesizer and sampler by Sonic Academy was used to create the initial character layer of the kick. All samples have been created at 128bpm since this is the most popular tempo in EDM. The kick sample has a quarter-note length, and it is tuned at A1 (54Hz) dropping to G1 (49Hz) as shown in Figs. 150-154:

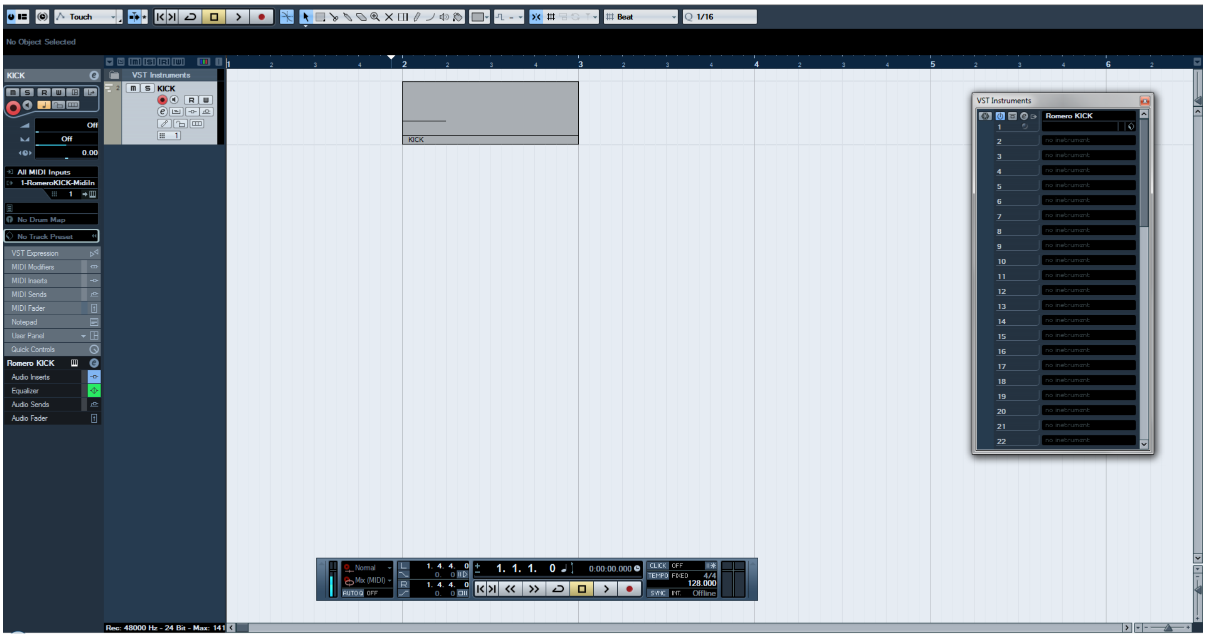


Figure 150: MORALIS, C. (2016) 'Kick Layer 1' [Screen Shot]

Figure 151: MORALIS, C. (2016) 'Kick Layer 1- Settings A' [Screen Shot]

Figure 152: MORALIS, C. (2016) 'Kick Layer 1- Settings B and C' [Screen Shot]

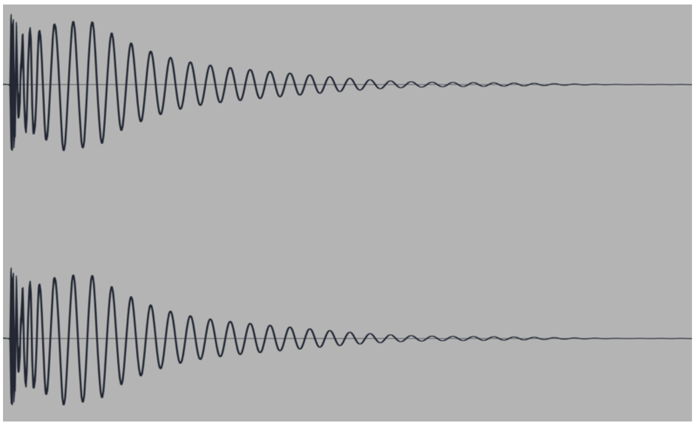


Figure 153: MORALIS, C. (2016) 'Kick Layer 1- waveform' [Screen Shot]

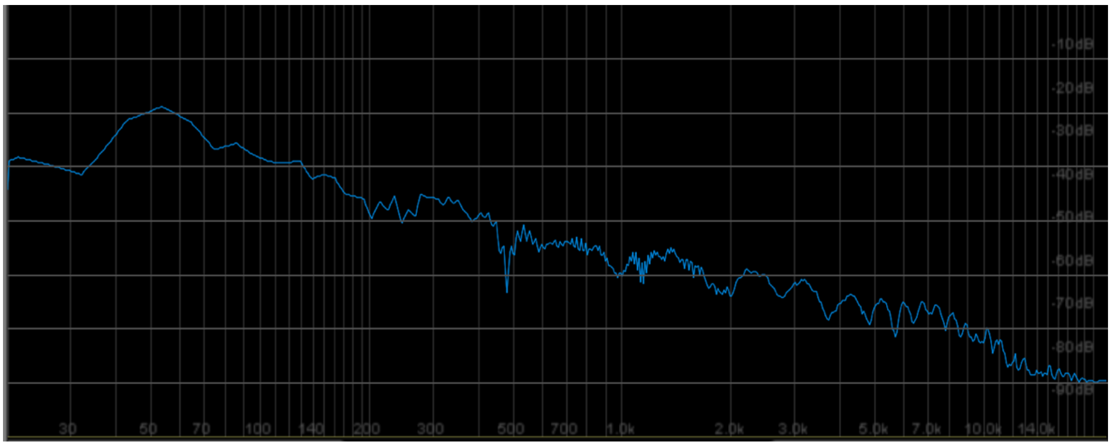


Figure 154: MORALIS, C. (2016) 'Kick Layer 1- spectrum' [Screen Shot]

KICK SAMPLE 1: Audio Example 71

Following the initial sound, a second layer has been added to create the appropriate punch that is found in EDM-style songs, adjusting the volume fades of the second layer to avoid glitches and spikes. See Fig. 155.

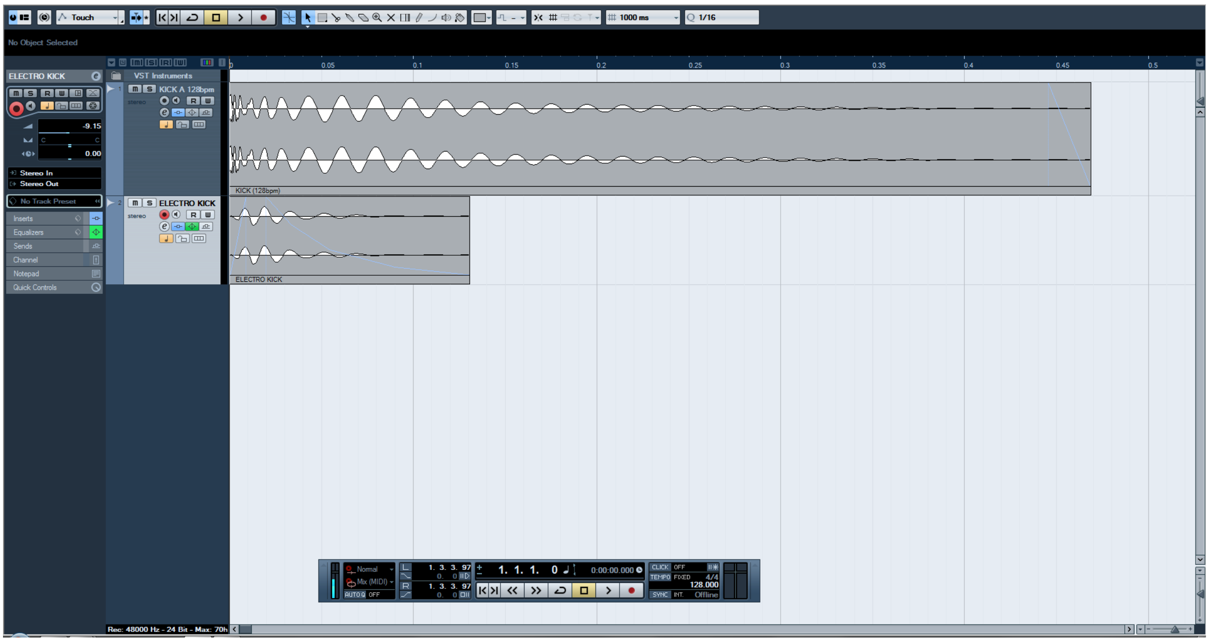


Figure 155: MORALIS, C. (2016) 'Layers 1 and 2' [(Screen Shot]

To combine these two layers without creating phase issues that will affect the overall mixing process later, the phase interaction process has been applied. The next picture shows the 'Pi' plugin by SoundRadix that dynamically rotates the phase to achieve the best possible phase correlation between these two sounds. See Fig. 156.

Figure 156: MORALIS, C. (2016) 'Layers 1 and 2 with Pi' [Screen Shot]

A low-cut filter and a mono maker plugin have been applied to the second layer to centre the focus of the sample and improve the mono compatibility. See Fig. 157.

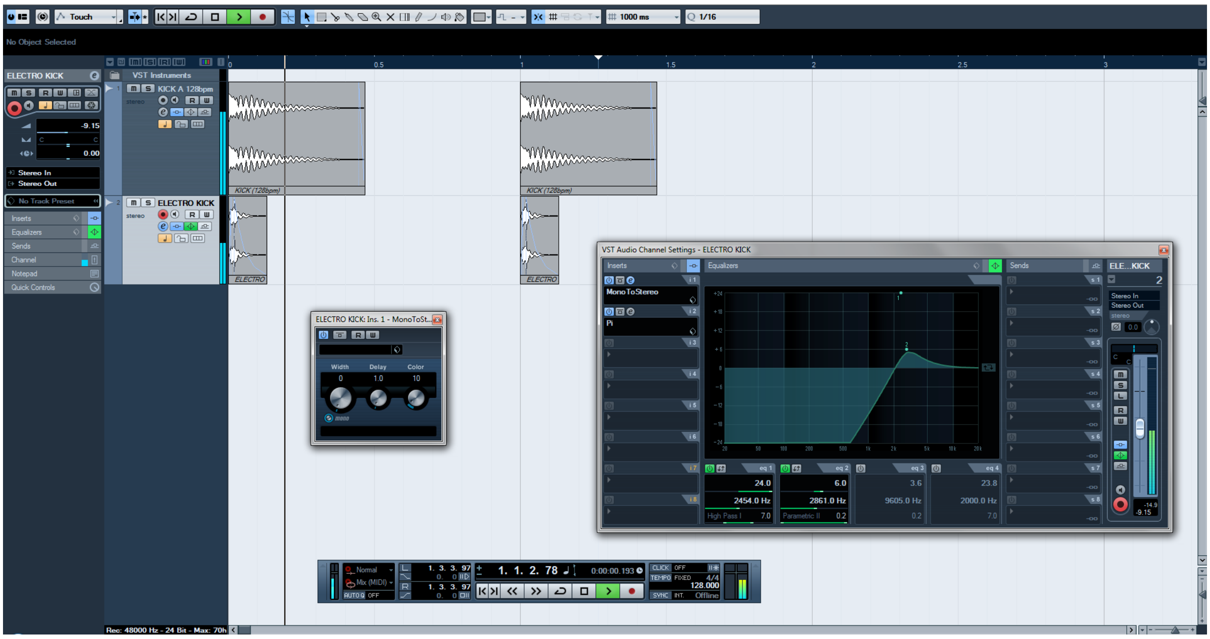


Figure 157: MORALIS, C. (2016) 'Layers 2 settings' [Screen Shot]

To mix the overall signal, equalization, compression and limiting have been applied. The compression follows the common technique found in EDM of the synchronized attack and release according to the song's tempo. However, to avoid excessive equalization and possible comb filtering, as well as to linearize the frequency response, the 'Unfilter' plugin from Zynaptiq has been added before the final limiting process. See Fig. 158.

Figure 158: MORALIS, C. (2016) 'Layers 1 and 2 - group FX' [Screen Shot]

KICK SAMPLE 2: Audio Example 72

Since it is necessary to minimize the post-production process during a live performance, in Fig. 159 is shown the mastering process applied to the overall sample of the kick:

Figure 159: MORALIS, C. (2016) 'Layers 1 and 2 - mastering' [Screen Shot]

At this final stage, it is necessary to enrich the audio signal by creating harmonics with the harmonics shaping tool and applying the necessary compression. The spike removal tool will act as a treatment of the initial peak signal, creating a perceived warm sound, and minimizing the initial volume of the waveform allowing the limiter to work with more overhead. The necessary limiting process is applied again to control the overall output volume. It is important to mention that the 'Stereoizer' plugin by Nugen has been used to improve the total mono-to-stereo compatibility. See Fig. 160.

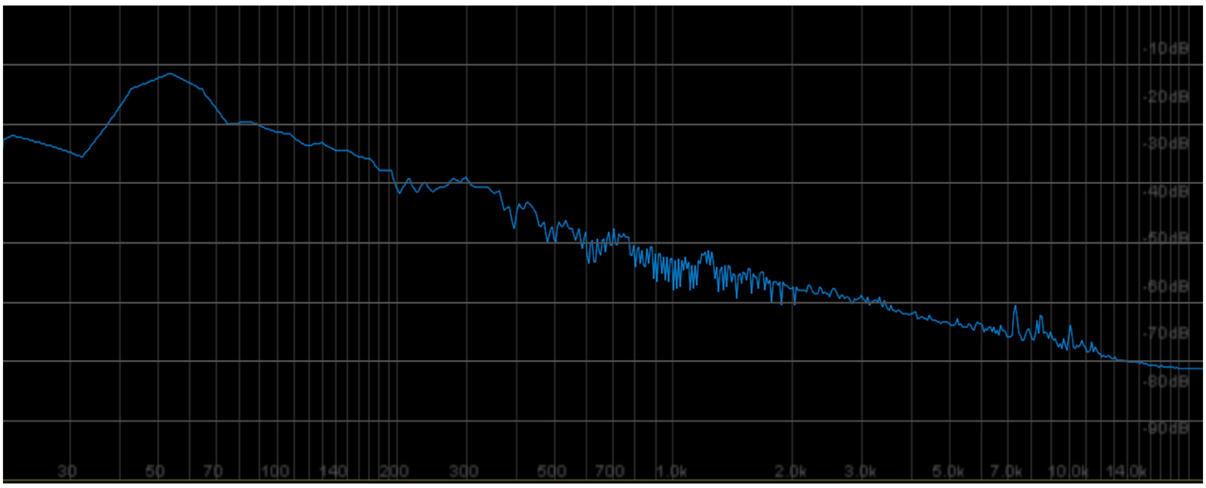


Figure 160: MORALIS, C. (2016) 'Layers 1 and 2 - spectrum' [Screen Shot]

KICK SAMPLE 3: [Audio Example 73]

Snare

For the creation of the snare's character sample, a previously produced snare has been used. However, it is necessary to sculpt the snare's sound to match the project's aesthetics and to blend well with the kick's timbre as if they were both samples of the same drum kit. Initially, compression equalization has been applied along with linearization of the frequency response to match that of the kick. Below are shown the effects used for designing and mastering the desired sound of the snare. See Fig. 161 and 162.

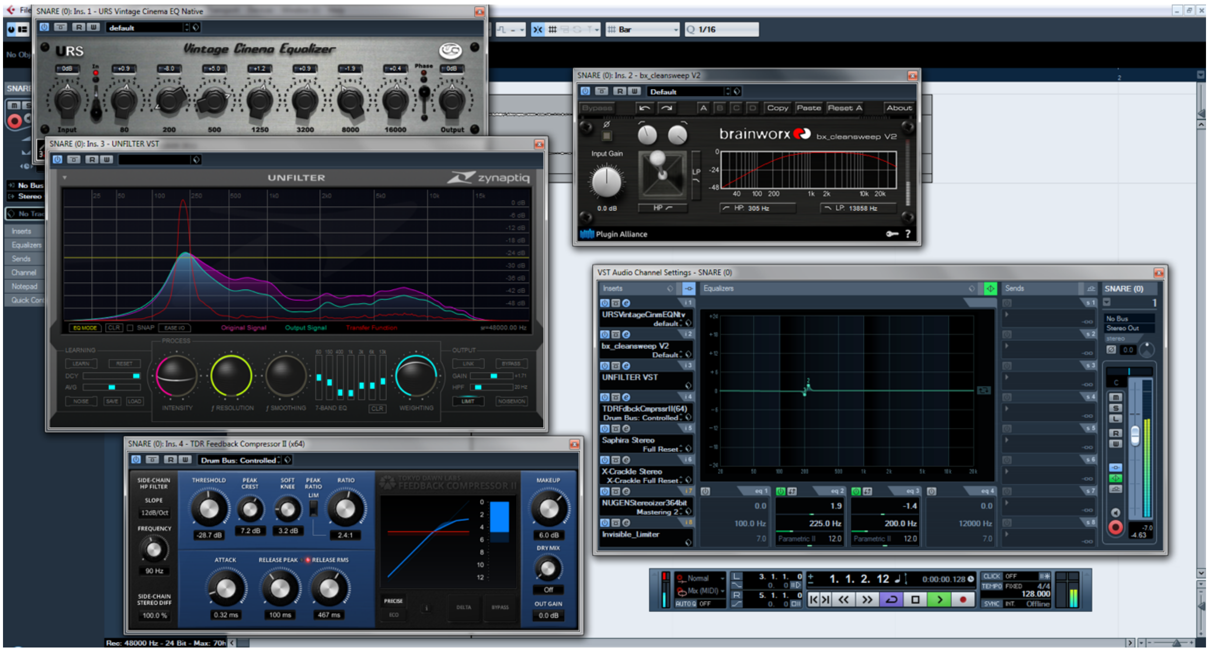


Figure 161: MORALIS, C. (2016) 'Snare Effects (1)' [Screen Shot]

Figure 162: MORALIS, C. (2016) 'Snare Effects (2)' [Screen Shot]

In Fig. 163 and 164 are demonstrated the spectrum curves of the two samples, the initial and the processed:

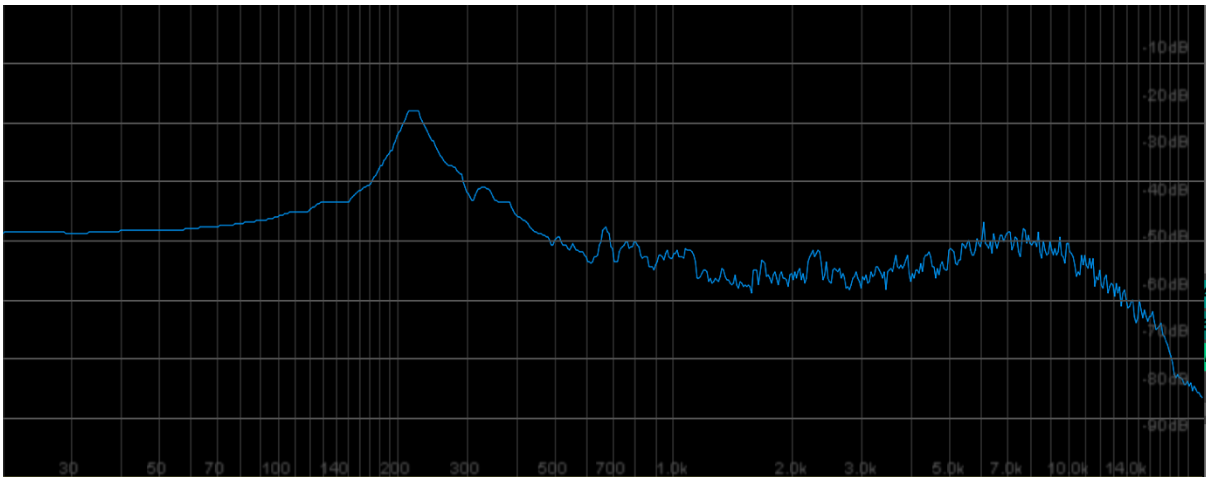


Figure 163: MORALIS, C. (2016) 'Initial Snare - spectrum' [Screen Shot]

- **SNARE SAMPLE 1:** Audio Example 74

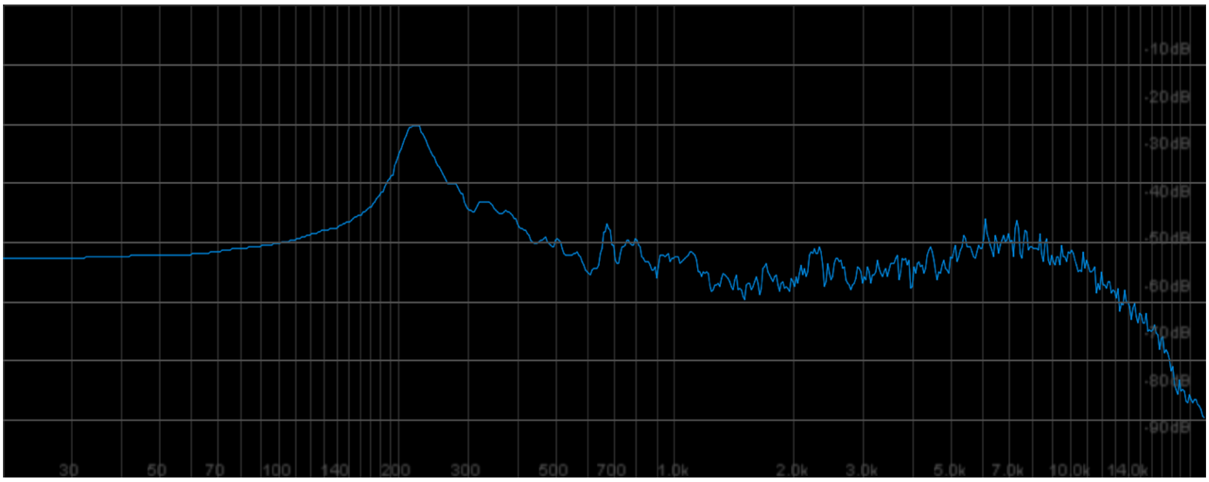


Figure 164: MORALIS, C. (2016) 'Processed Snare - spectrum' [Screen Shot]

- **SNARE SAMPLE 2:** Audio Example 75

At this stage, there are not many differences between them, as can be seen by comparing the two spectrum curves. However, after the processing of the snare's sound, the timbre is closer to the producer's aesthetics.

Furthermore, apart from the compressor and the limiter that have been applied to the kick sample, a mono-focus plugin that targets the frequencies below 250Hz has been used on the snare, both to match the volumes of these two samples and to achieve better mono compatibility. See Figs. 165 and 166.

Figure 165: MORALIS, C. (2016) 'Kick Mastering' [Screen Shot]

Figure 166: MORALIS, C. (2016) 'Snare Mastering' [Screen Shot]

The kick and snare samples are mixed in such a way, so far, that both reach -6.5db. Below is shown an audio example of these two samples playing together:

- **KICK AND SNARE 1:** [Audio Example 76]

However, to further match the snare's timbre to the kick sample, the waveform has been divided into different sections, and various effects have been applied to every channel. The procedure followed is shown more specifically in Figs. 167 and 168.

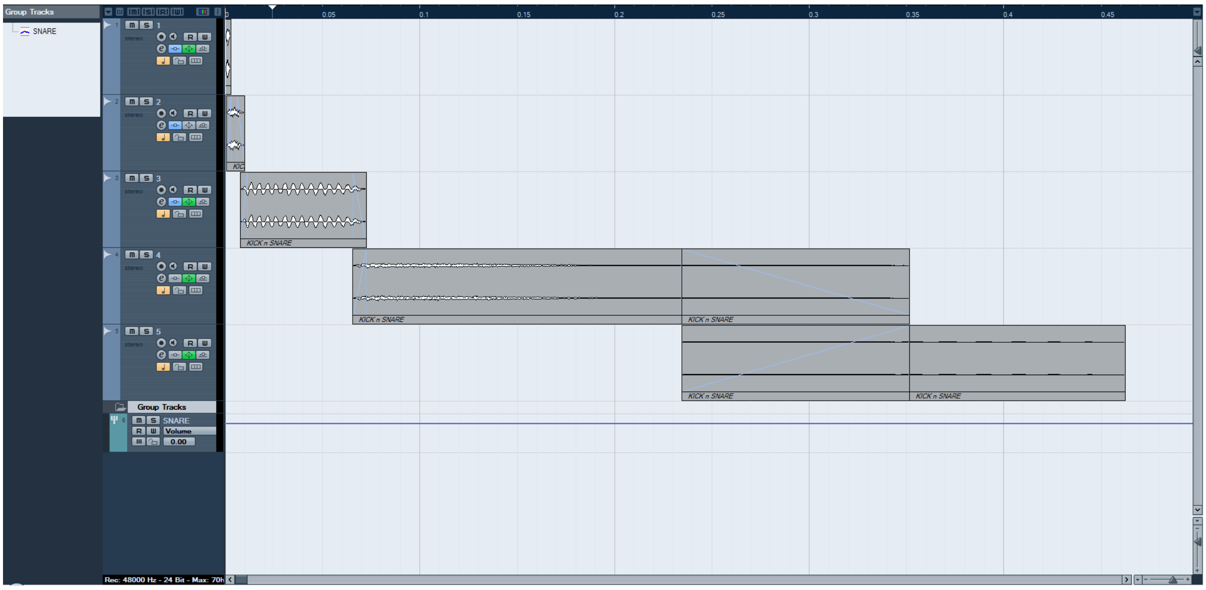


Figure 167: MORALIS, C. (2016) 'Snare Sections (1)' [Screen Shot]

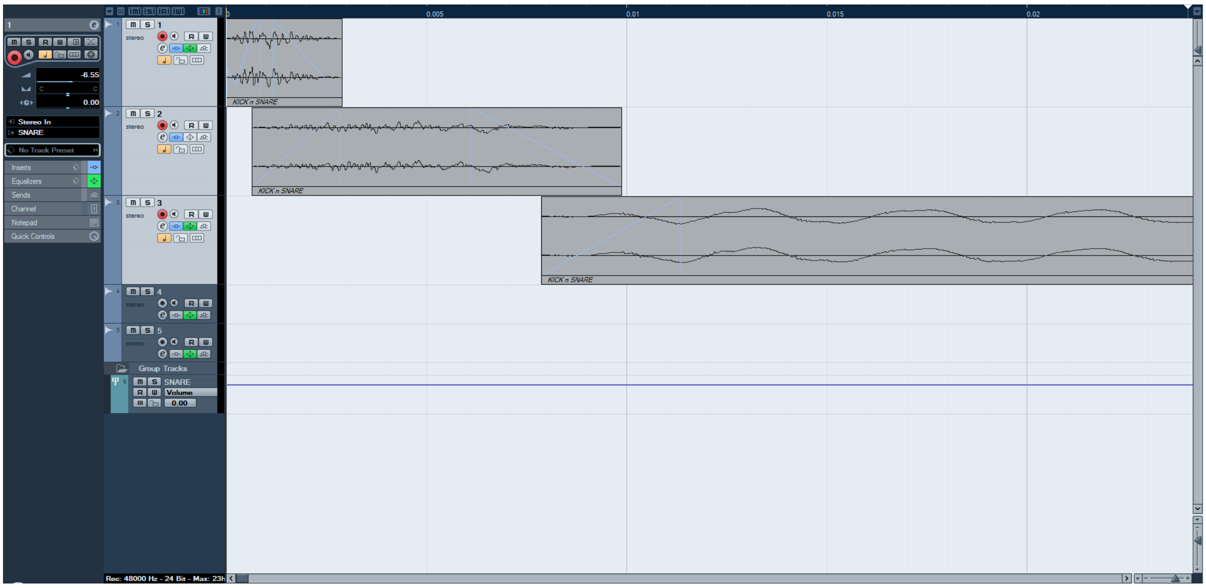


Figure 168: MORALIS, C. (2016) 'Snare Sections (2)' [Screen Shot]

The division of a sound sample into multiple sections permits the creation of a richer, more detailed sound by effectively controlling every part of the sample as the sound develops. The following pictures show the different effects applied in order to sculpt the sample and produce the desired sound.

In the first two channels, the de-crackle plugin has been applied in order to sculpt the initial transients and to achieve the desired equalization. See Figs. 169 and 170.

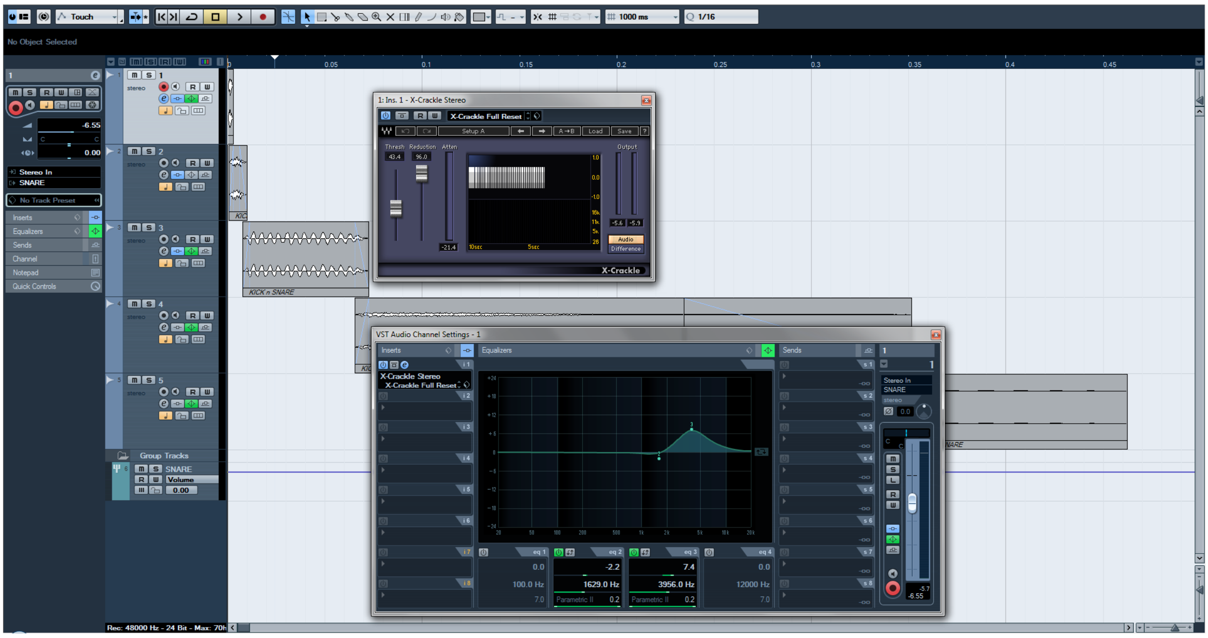


Figure 169: MORALIS, C. (2016) 'Snare Section 1' [Screen Shot]

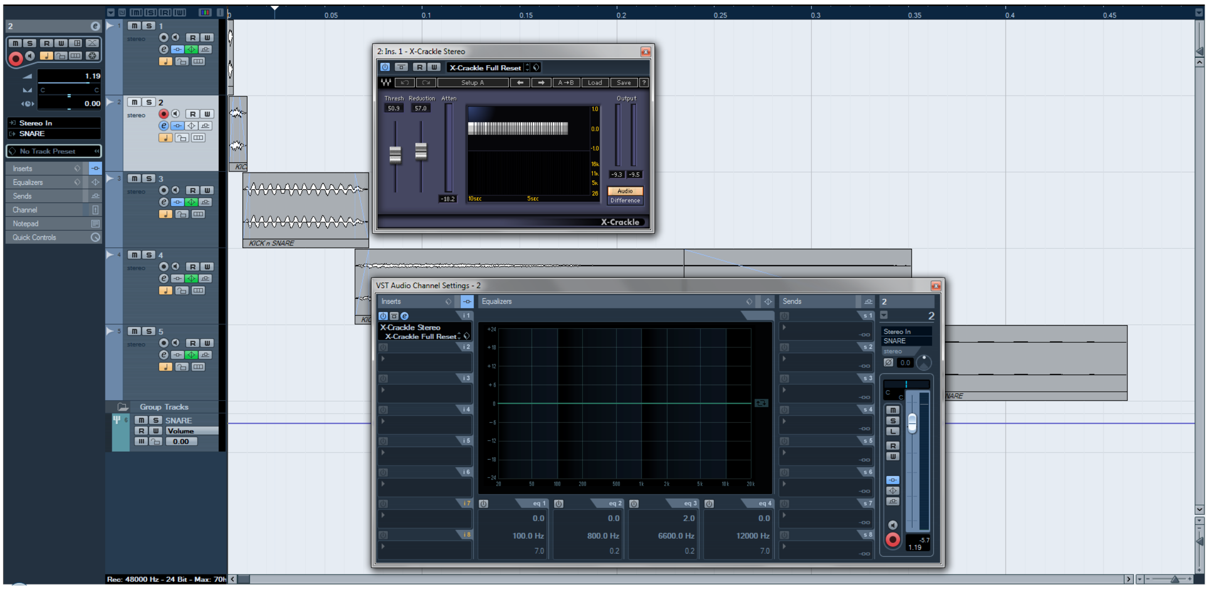


Figure 170: MORALIS, C. (2016) 'Snare Section 2' [Screen Shot]

In section 3, a compressor tightens the main sound of the sample, or body, along with equalization. See Fig. 171.

Figure 171: MORALIS, C. (2016) 'Snare Section 3' [Screen Shot]

In section 4, a low-cut filter is applied to remove any frequencies below 200Hz and make space for the kick. Also, in the group channel, a limiter is controlling the overall output signal.

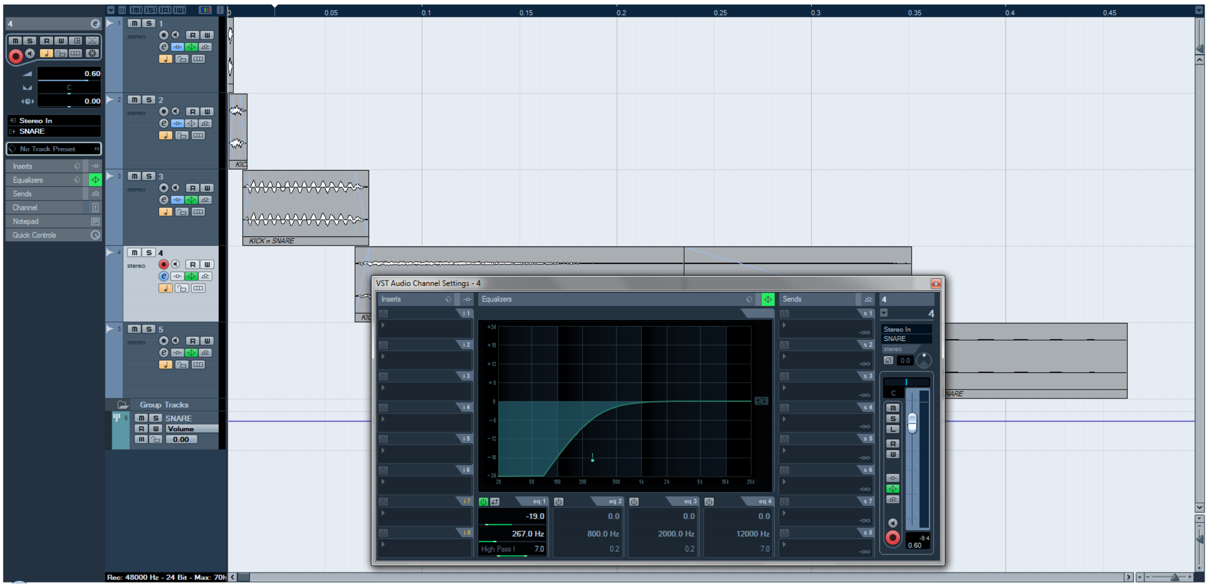


Figure 172: MORALIS, C. (2016) 'Snare Section 4' [Screen Shot]

After this procedure, the necessary adjustments to the kick and snare are made in order to match the volume and perceived loudness of the samples by applying compression, limiting and volume adjustments as shown in the following figures 173-176:

Figure 173: MORALIS, C. (2016) 'Snare Group Sections' [Screen Shot]

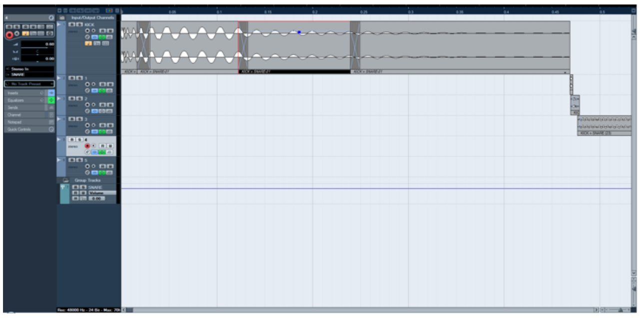


Figure 174: MORALIS, C. (2016) 'Kick final adjustments' [Screen Shot]

Figure 175: MORALIS, C. (2016) 'Kick final effects' [Screen Shot]

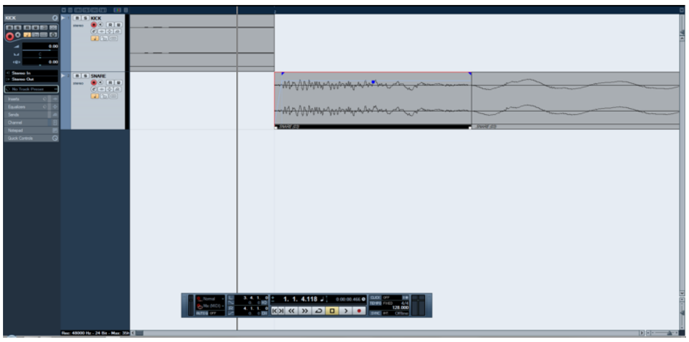


Figure 176: MORALIS, C. (2016) 'Snare final adjustments' [Screen Shot]

Figs. 177 and 178 show a comparison with the previous kick and snare balanced samples.

Before:

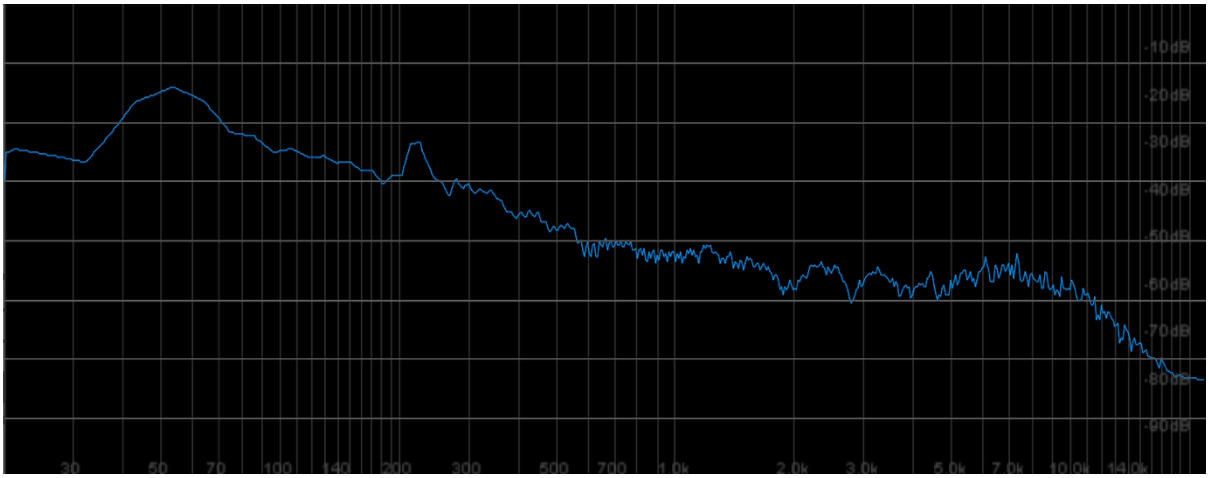


Figure 177: *MORALIS, C. (2016) 'Kick n Snare- spectrum' [Screen Shot]*

- **KICK AND SNARE 1:** Audio Example 77

After:

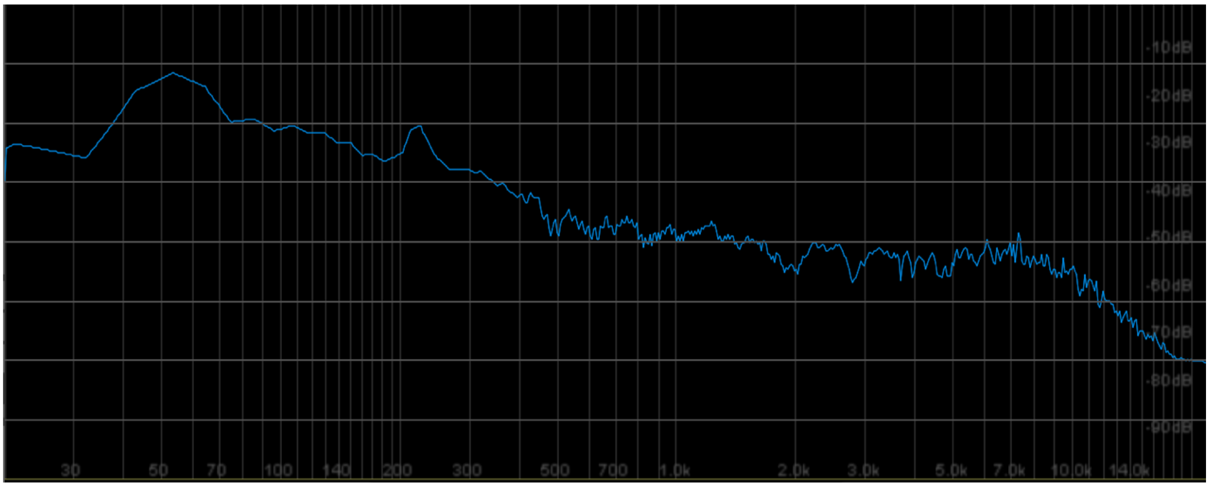


Figure 178: *MORALIS, C. (2016) 'Kick n Snare final - spectrum' [Screen Shot]*

- **KICK AND SNARE 2**: Audio Example 78

As shown above, the 54 and 220 Hz that respond to note A are emphasized, while there is space for other sounds to occur between 500-6500Hz. An emphasis has been made on 7Khz to improve the samples' presence.

Claps

For the creation of the 'claps' sample, the same production approach will be used. However, the blend of two samples with different stereo widths (one with wide and one closer to mono) is applied to improve the mono compatibility. Below are shown the effects used on the stereo clap sample to improve and match the sound quality and timbre to the existing kick and snare. See Fig. 179.

Figure 179: MORALIS, C. (2016) 'Claps adjustments' [Screen Shot]

Adding the mono clap sample and applying the dynamic phase reverse technique between the two channels, it is again necessary to use the same effects that were applied to the first sample to blend the two audio signals and timbres properly. See Fig. 180.

Figure 180: MORALIS, C. (2016) 'Claps group effects 1 [Screen Shot]

Further equalization, compression and limiting are applied to blend the timbre of the claps with the kick and snare sample. See Fig. 181.

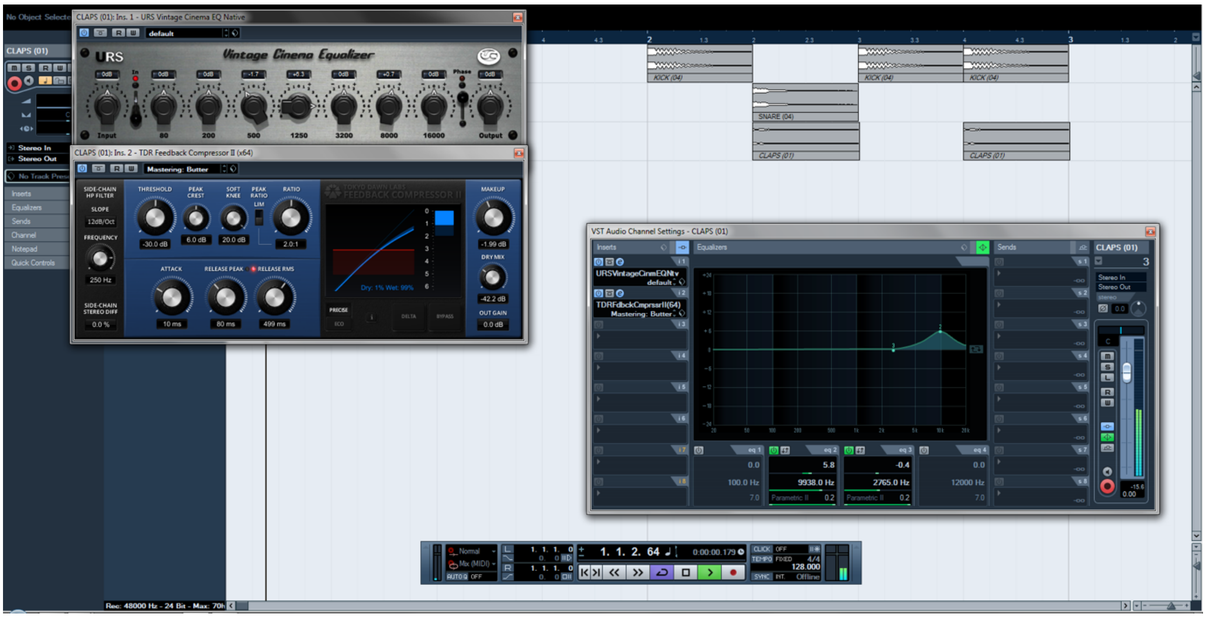


Figure 181: MORALIS, C. (2016) 'Claps group effects 2 [Screen Shot]

Fig. 182 shows the spectrum of the 'claps':

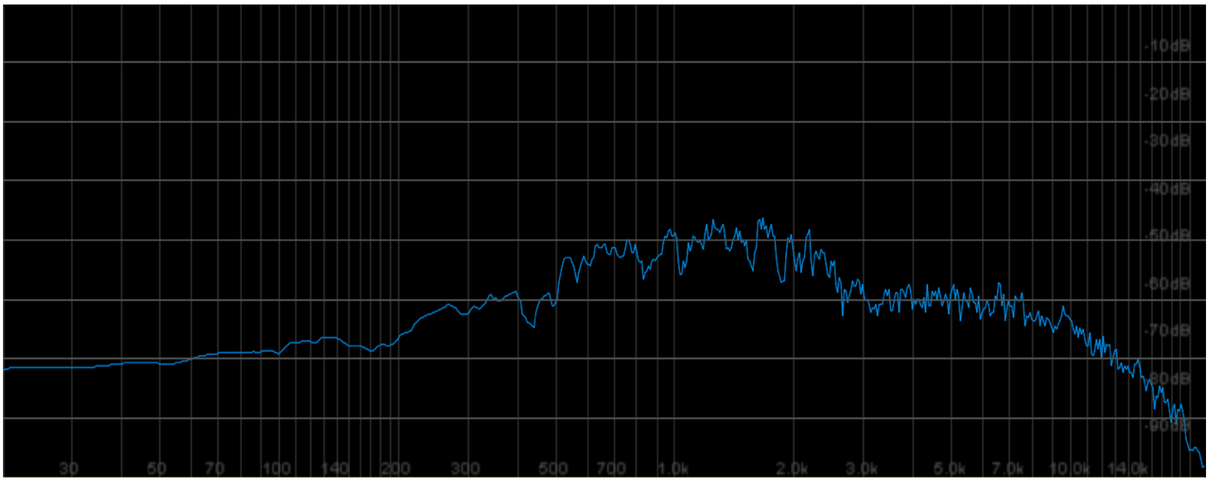


Figure 182: MORALIS, C. (2016) 'Claps - spectrum' [Screen Shot]

As shown above, the frequencies of the claps are focusing between 500 and 3000. This allows the claps sample to blend properly with the kick and snare.

- **CLAPS SAMPLE:** Audio Example 79

Soft Snare

The ‘soft snare’ sample has been created with the multi-section mixing process, applying a transient shaper at the first two sections, and compression and equalization to the next sections, along with stereo and mono compatibility improvement tool. See Fig. 183-187.

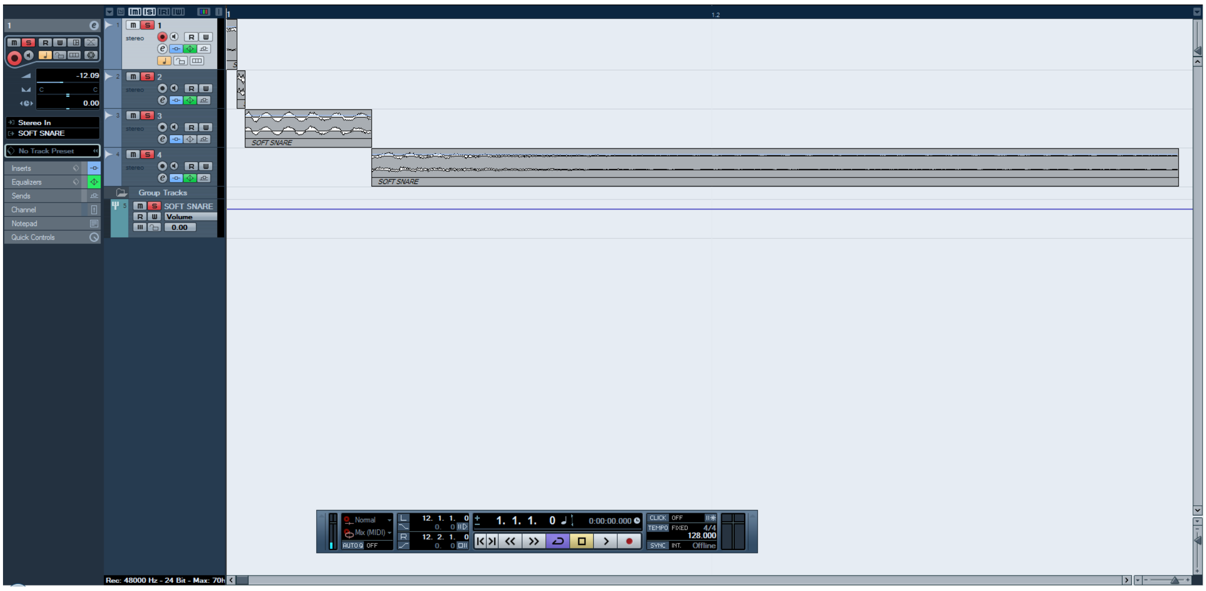


Figure 183: MORALIS, C. (2016) ‘Soft Snare Sections’ [Screen Shot]

Figure 184: MORALIS, C. (2016) ‘Soft Snare Section 1’ [Screen Shot]

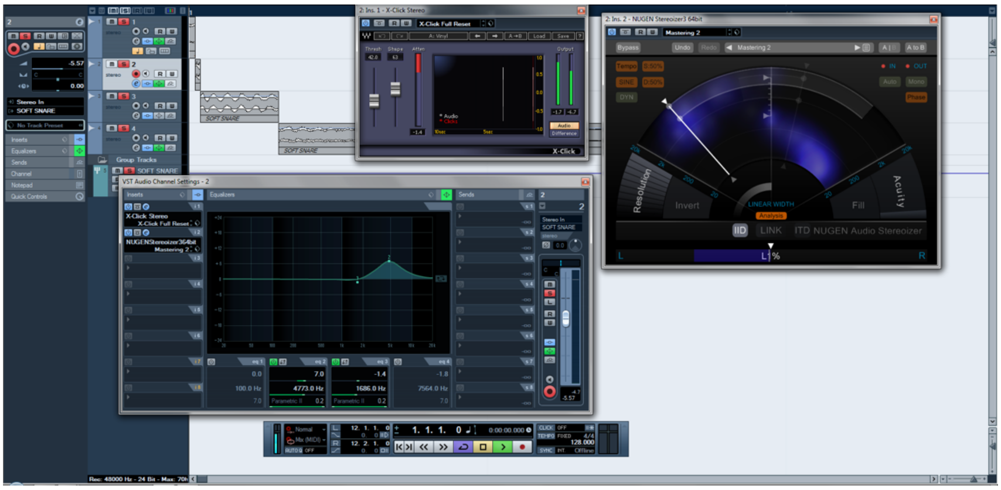


Figure 185 MORALIS, C. (2016) 'Soft Snare Section 2' [Screen Shot]

Figure 186: MORALIS, C. (2016) 'Soft Snare Section 3' [Screen Shot]

Figure 187: MORALIS, C. (2016) 'Soft Snare Section 4' [Screen Shot]

The final adjustments to the pitch and timbre of the sample are made as show in Fig. 188:

Figure 188: MORALIS, C. (2016) 'Soft Snare Group' [Screen Shot]

- **SOFT SNARE 1:** [Audio Example 80]

Further adjustments have been applied as shown in figures 189 and 190:

Figure 189: MORALIS, C. (2016) 'Soft Snare Further Adjustments 1 [Screen Shot]

- **SOFT SNARE 2:** [Audio Example 81]

Figure 190: MORALIS, C. (2016) 'Soft Snare Further Adjustments 2 [Screen Shot]

- **SOFT SNARE 3:** [Audio Example 82]

Fig. 191 shows the final curve of the 'Soft Snare' sample.

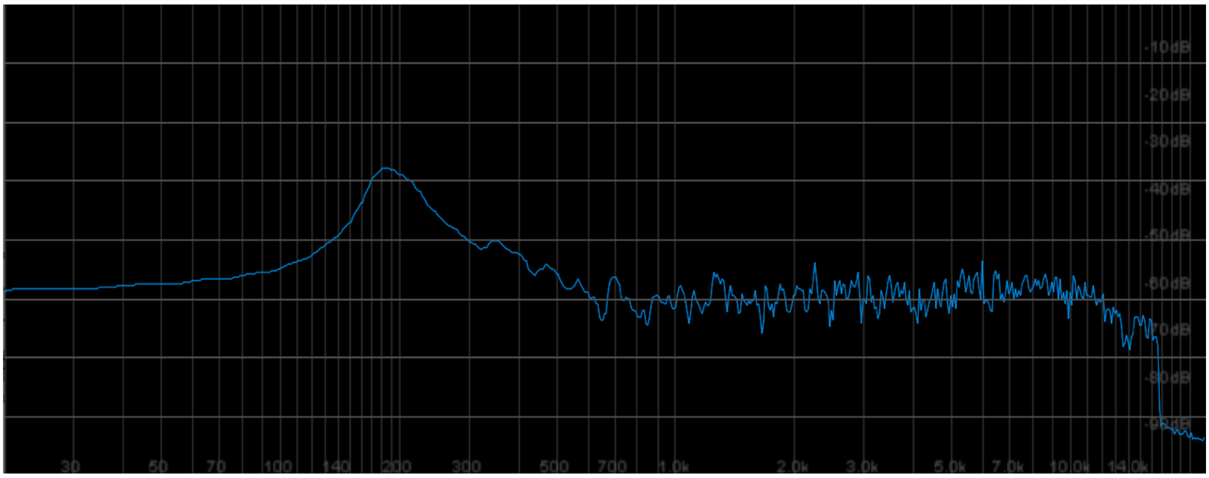


Figure 191: MORALIS, C. (2016) 'Claps Final- spectrum' [Screen Shot]

Reverse Reverb

Within the EDM style, a common effect is the reverse reverb sound effect on the snare before the groove starts. A reverb has been produced with the hit of the snare at 128bpm while the initial signal of the sample is the reversed reverb with a duration of one-quarter. The appropriate tonal linearization has been applied to match the drums' sound. This sound effect has been pre-produced, since the real-time application of it would consume enough CPU power to affect the overall performance of the computer. See Fig. 192 and 193.

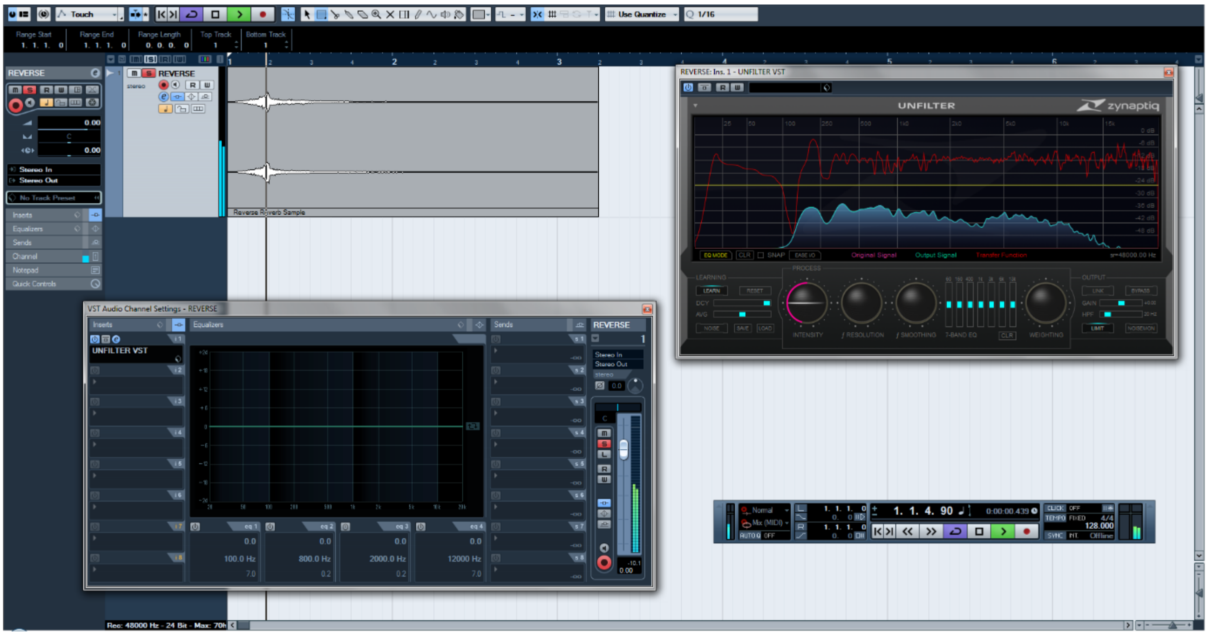


Figure 192: MORALIS, C. (2016) 'Reverse Reverb' [Screen Shot]

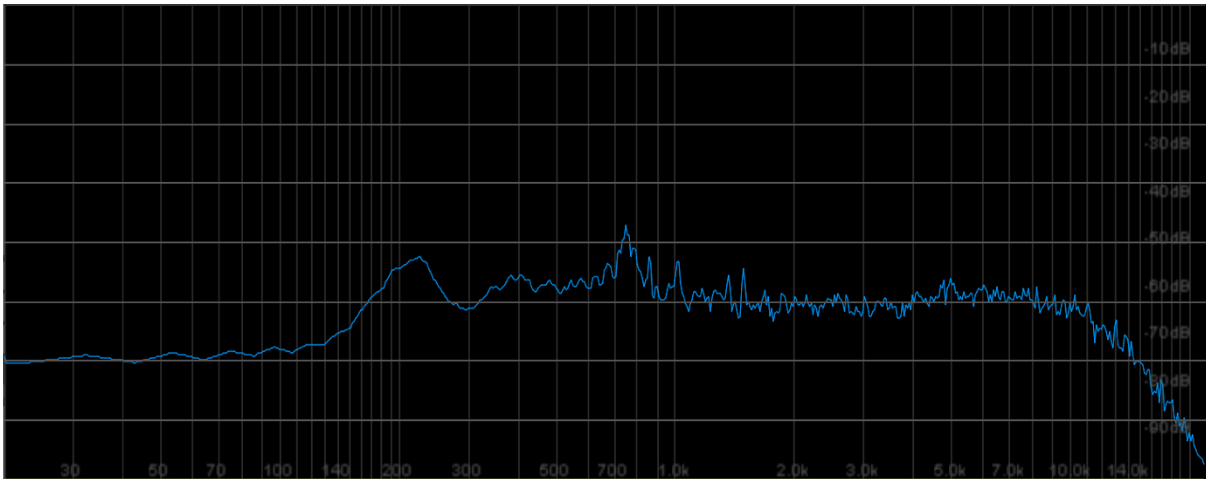


Figure 193: MORALIS, C. (2016) 'Reverse Reverb - spectrum' [Screen Shot]

Since this reverse reverb sound effect will be used along with the snare, the spectrum follows a curve similar to a snare's sample. However, since it is a sound sample that most of the time will be used on its own, emphasis has been put on the mid-range frequencies to improve the sound balance between the overall mix and this sound sample.

<u>Kick Clap</u>

In the style of Electro House, producers around the world add a second layer along with the kick sample. This sound is usually a clap sound. To improve the sound quality of the sample, the same production process has been followed, as shown above, with the Pi plugin applied to improve the phase relationship between the kick clap and the kick sample. Fig. 195 shows the frequency spectrum curve of this sample:

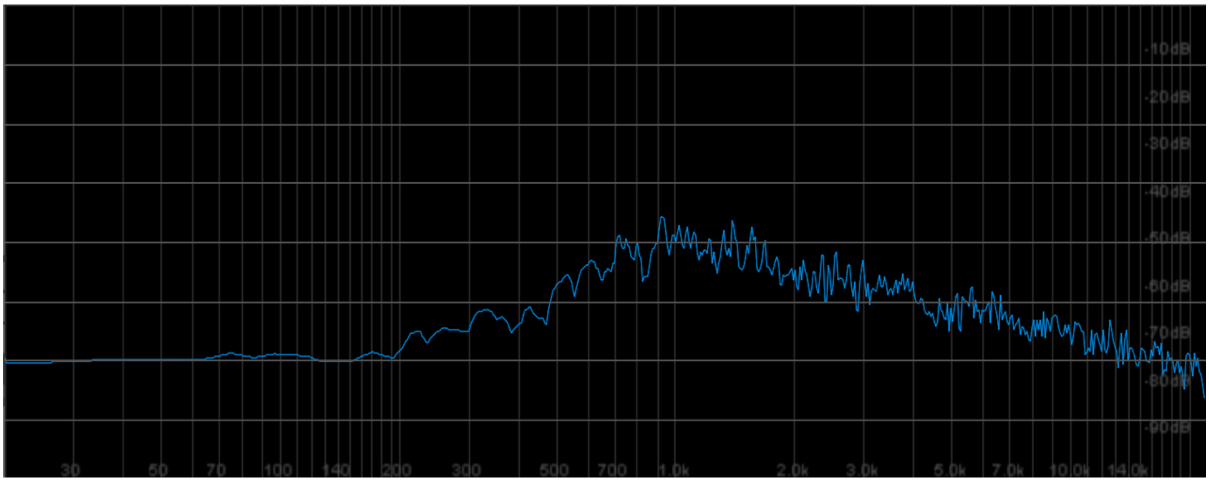


Figure 194: MORALIS, C. (2016) 'Kick Clap- spectrum' [Screen Shot]

Toms

The layering process for creating the cymbals samples has also been used for producing the tom samples. However, in this case, two different acoustic drum kits have been used to create the timbre of the toms. The mixing and mastering process follows the same concept of multi-compressing and limiting along with the manual editing of the transients to achieve the maximum sound compatibility with the rest samples of the kit. Again, three audio channels have been used: tom mic, overhead mics and room mics. However, in this case, five different dynamics have been selected. See Fig. 195 and 196.

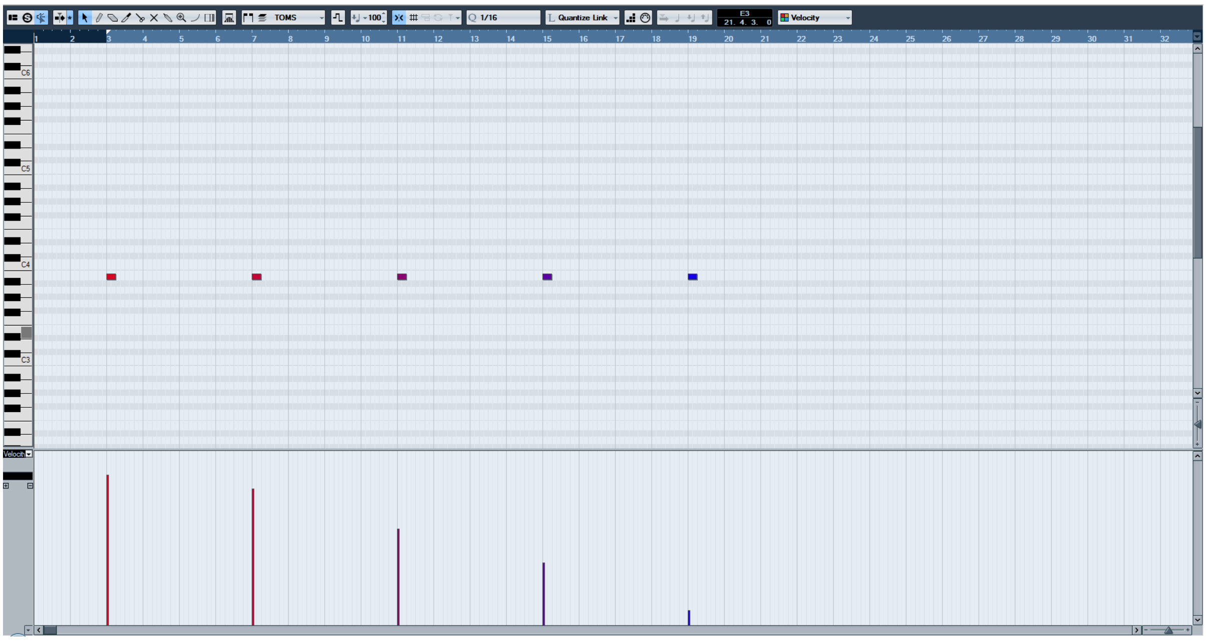


Figure 195: MORALIS, C. (2016) 'Toms - Dynamics' [Screen Shot]

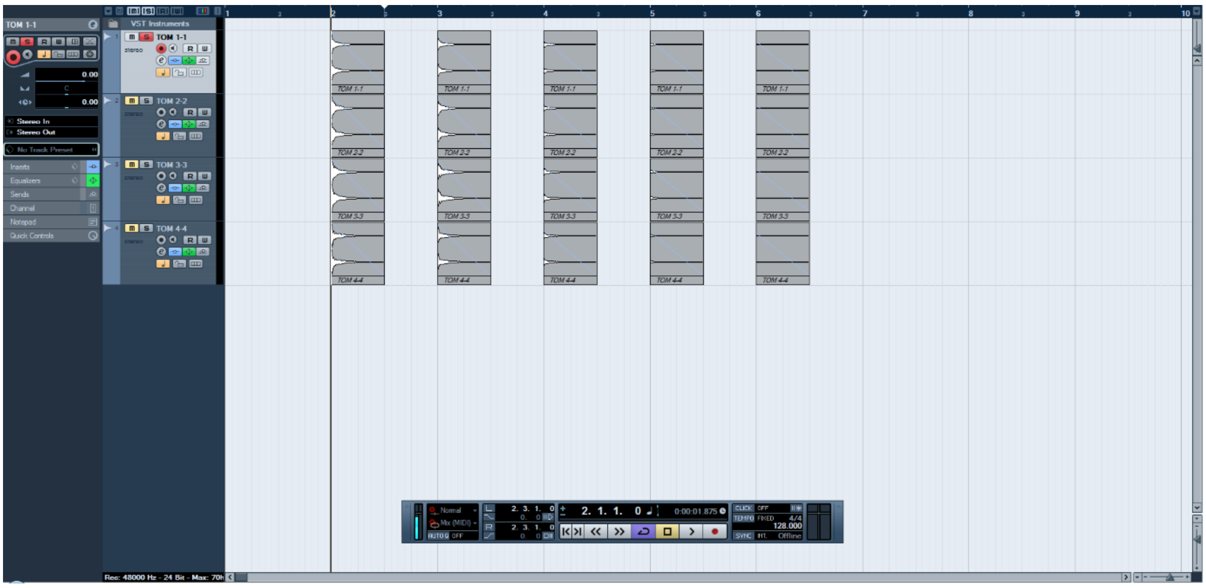


Figure 196: MORALIS, C. (2016) 'Final mixed Tom Samples' [Screen Shot]

Final Adjustments

To minimize the real-time mixing process and to match all the timbres of the samples created, final adjustments are made. See Fig. 197.

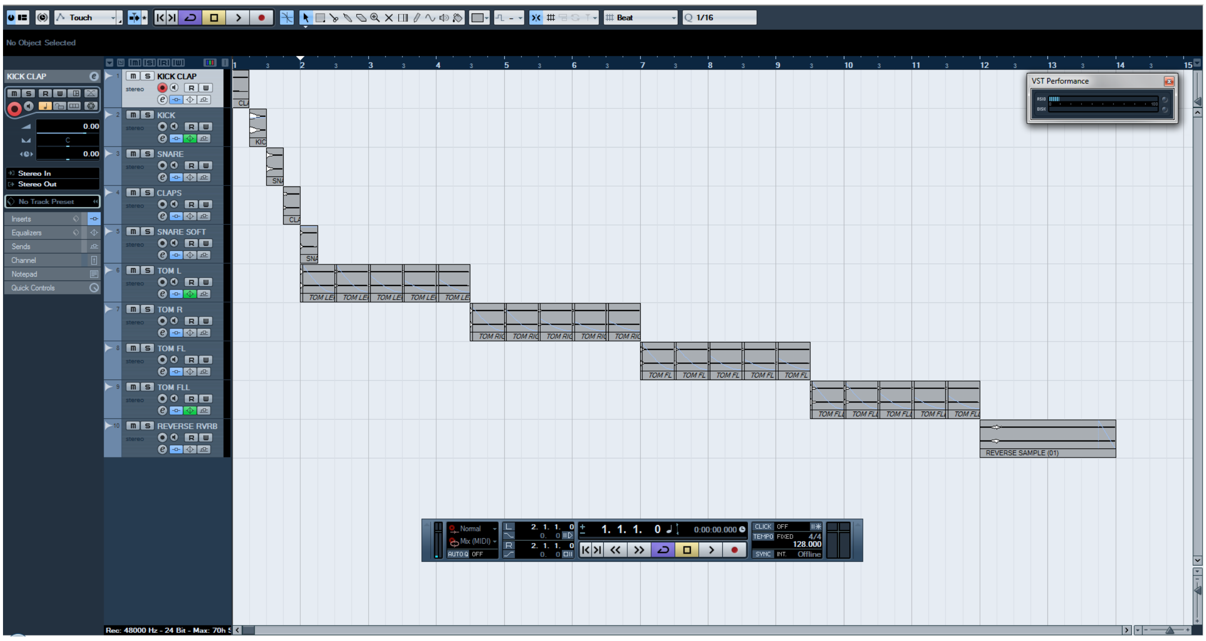


Figure 197: MORALIS, C. (2016) 'All Samples Overview' [Screen Shot]

In this stage, the 'Character' plugin has been used, a dynamic Eq and transient shaper tool by TC Electronic, along with a transparent limiter. In cases where additional equalization is needed, it has been applied. See Fig. 198.

Figure 198: MORALIS, C. (2016) 'All Samples Adjustments [Screen Shot]

Appendix 3

Kick

The sampler ‘Kick Synth’ by Sonic Academy has been used again to create the synthesized sound. Since it is about creating a synthesized sound, in practical terms it can be done with any sine or square wave generator. However, this sampler allows the user to shape the pitch envelope and the equalization and to add harmonics through the signature distortion embedded in the synthesizer. As shown in Fig. 199, the synthesized sound generator is used only when the initial punchy layer has been disabled:

Figure 199: MORALIS, C. (2016) ‘Kick - Synth’ [Screen Shot]

However, following the pitch selection, the appropriate envelope has been applied to help the punchy sound at the beginning of the sample by designing the volume envelope as shown in Fig. 200:

Figure 200: MORALIS, C. (2016) ‘Kick – Synth Volume Envelope’ [Screen Shot]

The volume follows an algorithmic curve with a sudden 90-degree change at the highest point. This creates a punchy sound while avoiding any unwanted spikes.

Regarding the character sample of the kick, a volume fade has been applied after the initial punchy sound. In this way, sub-bass sound that defines the pitch of the sample has been removed, helping to achieve a proper blend with the synthesized sound. See Fig.201.

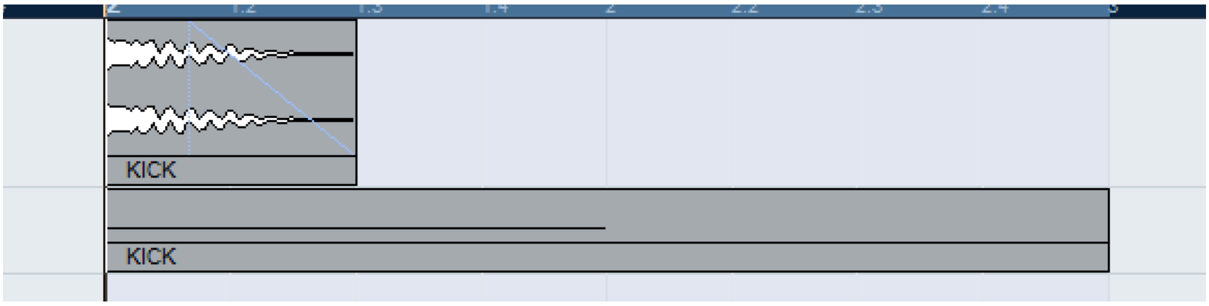


Figure 201: MORALIS, C. (2016) 'Kick –Character – Fade' [Screen Shot]

A further editing process between the two audio waveforms has been applied to improve the overall mix. See Fig. 202.

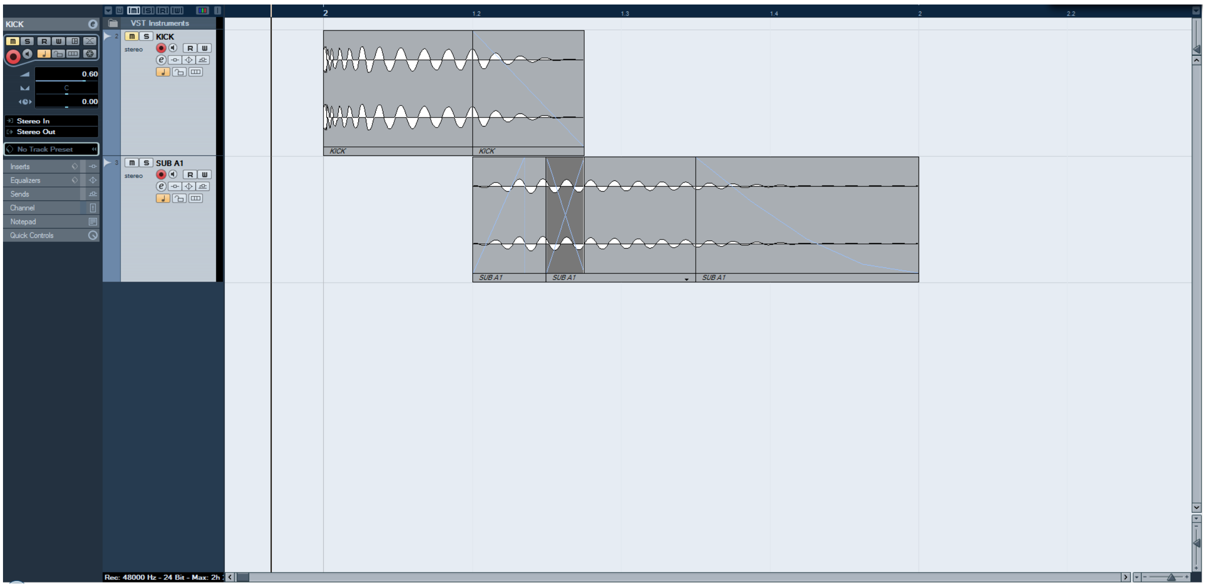


Figure 202: MORALIS, C. (2016) 'Kick Character and Pitch - Edits' [Screen Shot]

Following the same procedure, the 12 semitones of an octave have been produced and edited accordingly. As shown below, the volume of each synthesized wave differs. This is because the perceived volume of each frequency varies. Since it is a sound that does not play along with the initial character sample, it does not affect the overall volume of the kick. For the same perceived loudness levels, the samples are treated accordingly with compression, Eq and limiting. See Fig. 203.

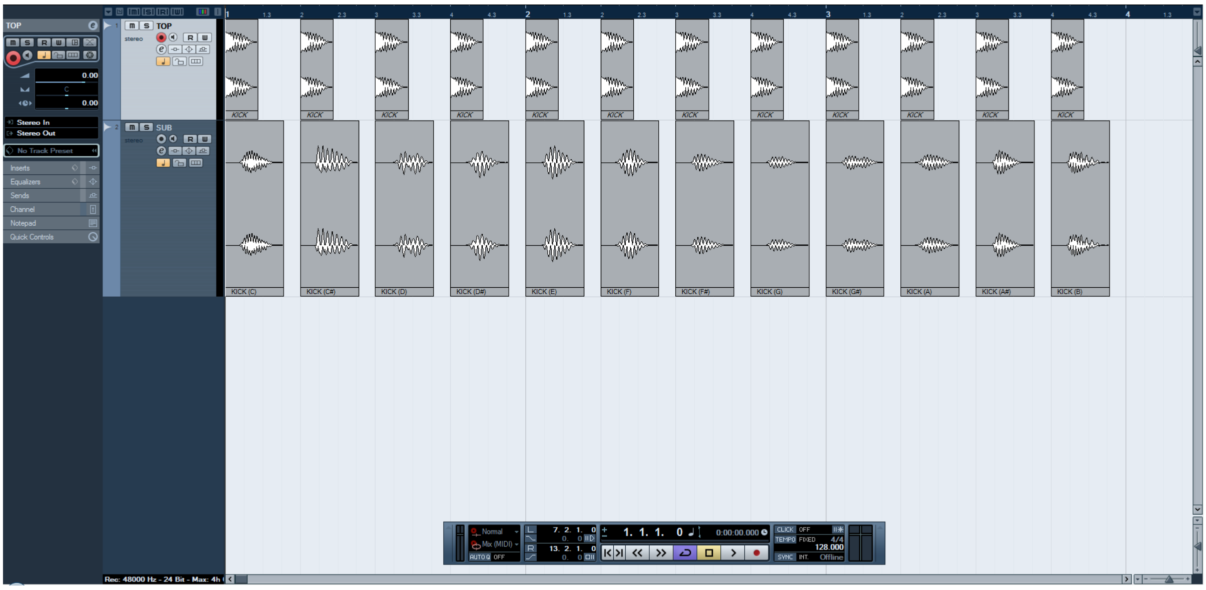


Figure 203: MORALIS, C. (2016) ‘Kick Character and Pitch – 12 semitones’ [Screen Shot]

As in Fig. 204, all the kick samples mixed together to reach a peak level of -6.5db regardless of the note played.

Figure 204: MORALIS, C. (2016) ‘All Kicks’ [Screen Shot]

Snare

For the final creation of the snare character sample, a slightly different procedure has been followed. To remove the tone or note from the sample while preserving the punchy sound of the sample, a multi-section editing approach has been applied, as shown in Fig. 205:

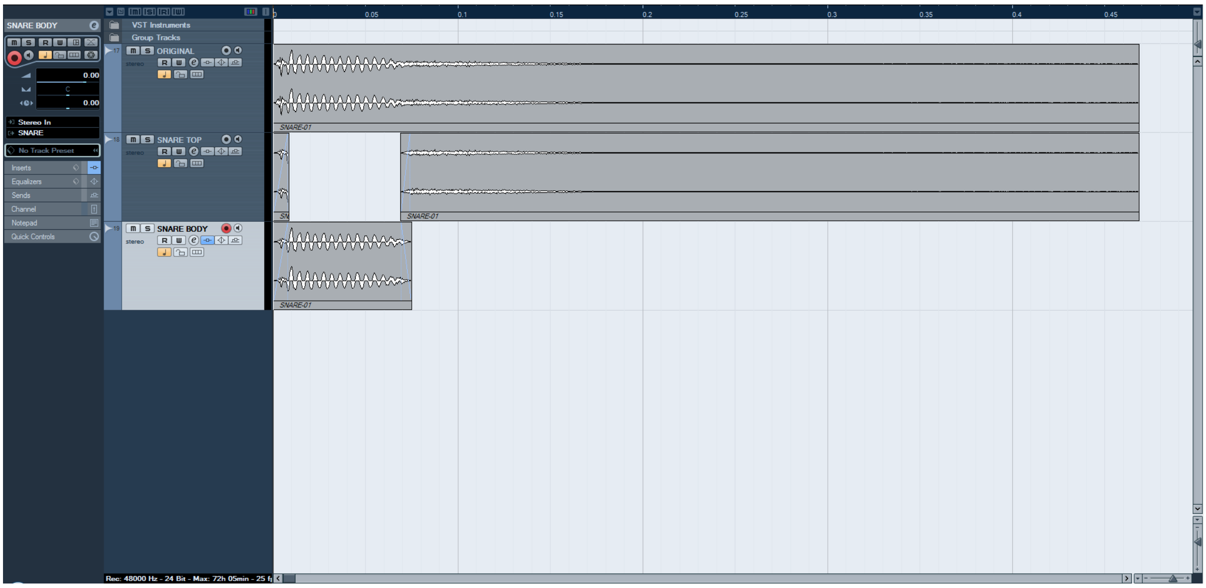


Figure 205: MORALIS, C. (2016) 'Snare Character Tone removal 1' [Screen Shot]

The sample has been divided into three parts: the transients; the main body, which contains the tonal information; and the tail. In the main body part, a linear phase equalizer has been applied to remove any information below 220Hz that defines the pitch of the sample, avoiding the creation of phase issues. Equalization adjustments are also made to improve the balance of the sound. Since it is not the initial part of the snare sample, affecting its peak level, it is not necessary to apply a limiter. See Fig. 206.

Figure 206: MORALIS, C. (2016) 'Snare Character Tone removal 2' [Screen Shot]

Furthermore, to improve the overall transition from one part to the other without any audible differences, white noise has been added to the overall snare's sound. See Fig. 207.

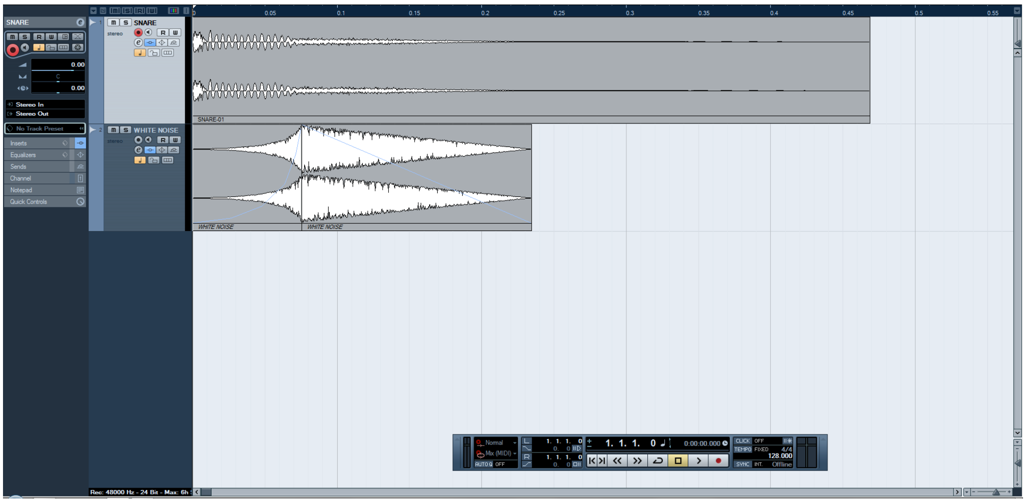


Figure 207: MORALIS, C. (2016) 'Snare Character Tone removal – White Noise 1' [Screen Shot]

Finally, a stereo-to-mono compatibility improvement plugin has been applied to the white noise sample to improve the stereo image of the white noise. See Fig. 208.

Figure 208: MORALIS, C. (2016) 'Snare Character Tone removal – White Noise 2' [Screen Shot]

Continuing with the creation of the pitch sample using the 'Synth Kick' by Sonic Academy, the synthesized 'A' note sound is shown in Fig. 209:

Figure 209: MORALIS, C. (2016) Snare - Synth' [Screen Shot]

However, in this case, the pitch is not static but changes from 216Hz to 221Hz. This slight movement in the pitch helps to distinguish the kick from the bass and at the same time work better in the mixing process as there is no need for further process.

As shown in Figs. 210 and 211, all of the snare's samples are mixed to reach a peak level of -6.7db regardless of the note played.

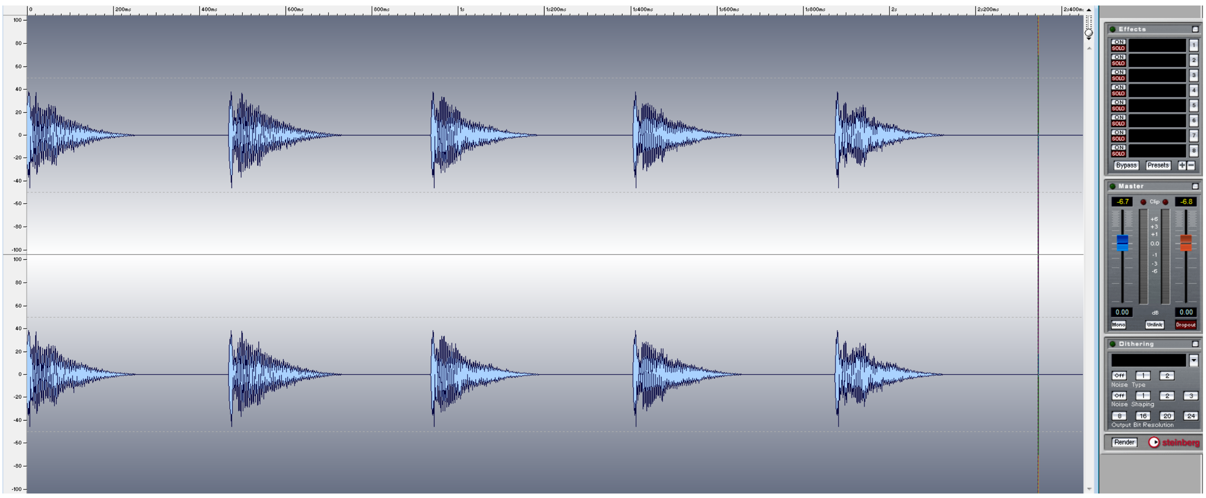


Figure 210 MORALIS, C. (2016) 'All Snares (1)' [Screen Shot]

Figure 211: MORALIS, C. (2016) 'All Snares (2)' [Screen Shot]

Toms

The snare's procedure has been followed also for the creation of the toms' samples. See Fig. 212.

Figure 212: MORALIS, C. (2016) Tom - Synth' [Screen Shot]

However, the pitch envelope curve, in this case, is slightly different. To emulate the movement of the drum heads, the pitch envelope changes between three different pitches: 66Hz, 137Hz, and 90Hz, the last of which is the tone of the sample. However, further nonlinear and linear equalization has been applied to improve the sample's timbre. See Fig. 213.

Figure 213: MORALIS, C. (2016) 'Tom Character Sample Creation' [Screen Shot]

Furthermore, the pitch samples are distributed over the four toms as shown below:

<u>TOM LEFT:</u>	B2	C 3	C#3	D3
<u>TOM RIGHT:</u>	G#2	A 2	A#2	
<u>TOM FLOOR LEFT:</u>	F2	F#2	G2	
<u>TOM FLOOR RIGHT:</u>	D2	D#2	E2	

All of the toms' samples are mixed to reach a peak level of -6.6db. See Fig. 214 and 215.

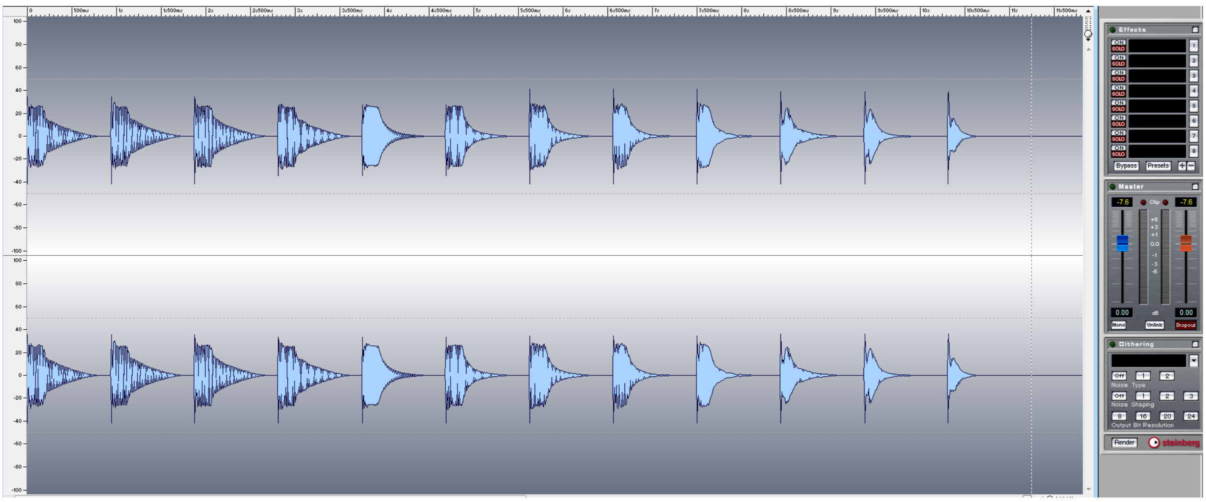


Figure 214: MORALIS, C. (2016) 'Al Toms (1)' [Screen Shot]

Figure 215: MORALIS, C. (2016) 'Al Toms' [Screen Shot]

The drum kit samples are shown below:

- KICK (1 Sample)
- KICK CLAP (1 Sample)
- SNARE (1 Sample)
- SOFT SNARE (1 Sample)
- REVERSE REVERB (1 Sample)
- CLAPS (1 Sample)
- TOM LEFT (5 Samples - Dynamics)
- TOM RIGHT (5 Samples - Dynamics)
- TOM FLOOR LEFT (5 Sample - Dynamics)
- TOM FLOOR RIGHT (5 Samples - Dynamics)
- CYMBAL LEFT (4 Sample – Dynamics)
- CYMBAL CENTER (4 Sample – Dynamics)
- CYMBAL RIGHT (4 Sample – Dynamics)
- HIHAT PEDAL (2 Samples)
- HIHAT CLOSED (5 Samples – Dynamics)
- HIHAT CLOSED (1 Sample – Parallel)
- HIHAT OPEN (2 Samples)
- HIHAT OPEN (1 Sample - Parallel)
- RIDE (2 Samples – Dynamics)
- RIDE BELL (3 Samples – Dynamics)

Appendix 4

In the case of hi-hats, a common approach is to layer different sample lengths with varying types of reverb. This approach serves to produce a multi-timbral sound rich in audio variances. However, in this instance, instead of this technique, another hi-hat layer has been created to act as reverberation for the primary sample, minimizing the necessity for extra reverb, and thus CPU power. This layer will be used in parallel with the hi-hats, leaving the main hi-hat sample unaffected. See Fig. 216.

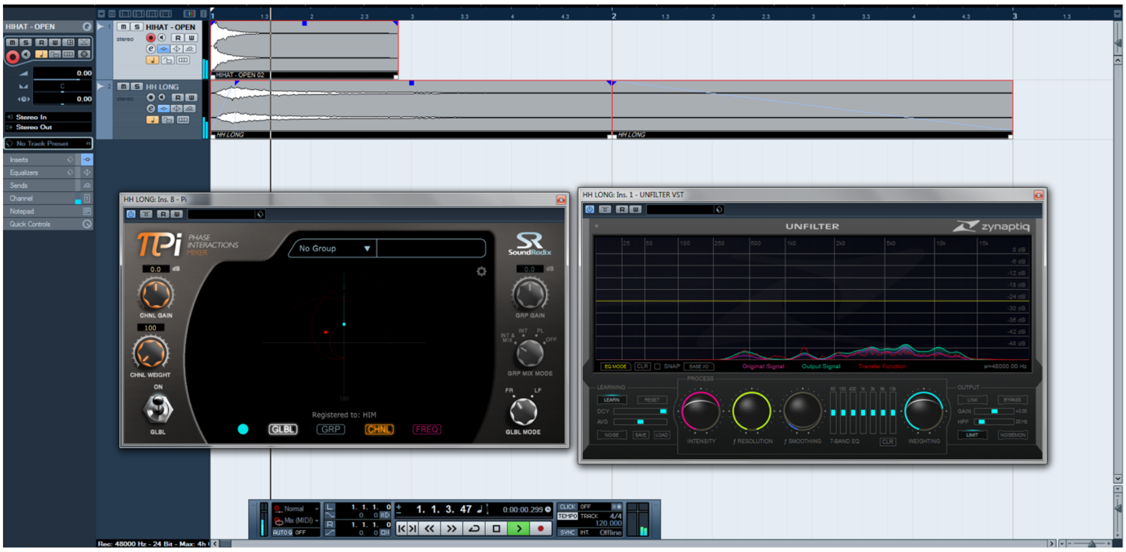


Figure 216: MORALIS, C. (2016) 'Extra HH – Reverb Layer' [Screen Shot]

At this stage, apart from the 'Pi' technique and the common effects that already have been used for the mixing and mastering process, it is necessary to align the different audio channels in order to avoid phasing issues. The plugin 'Auto Align' has been used as shown in Fig.217:

Figure 217: MORALIS, C. (2016) 'Cymbals - Alignment' [Screen Shot]

To also improve the natural response of the cymbals' timbre, three dynamics have been used as shown in Fig. 218, taken by the acoustic kit, and a fourth one, at a lower volume, created by using only the electronic sample.

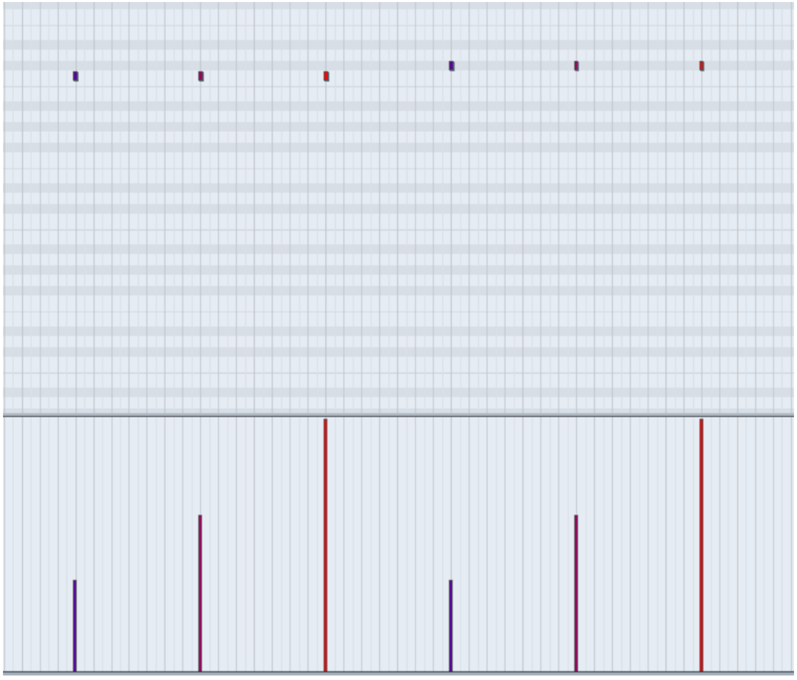


Figure 218: MORALIS, C. (2016) 'Cymbals dynamics'- spectrum' [Screen Shot]

However, regarding the hi-hats, five different dynamics have been selected. With the creation of the final samples, a group mastering treatment has been used to blend the overall sound of the cymbals as well as editing the sustain and release time of all samples to match each other. See Figs. 219 and 220.

Figure 219: MORALIS, C. (2016) 'Cymbals - 'Final Mastering' [Screen Shot]

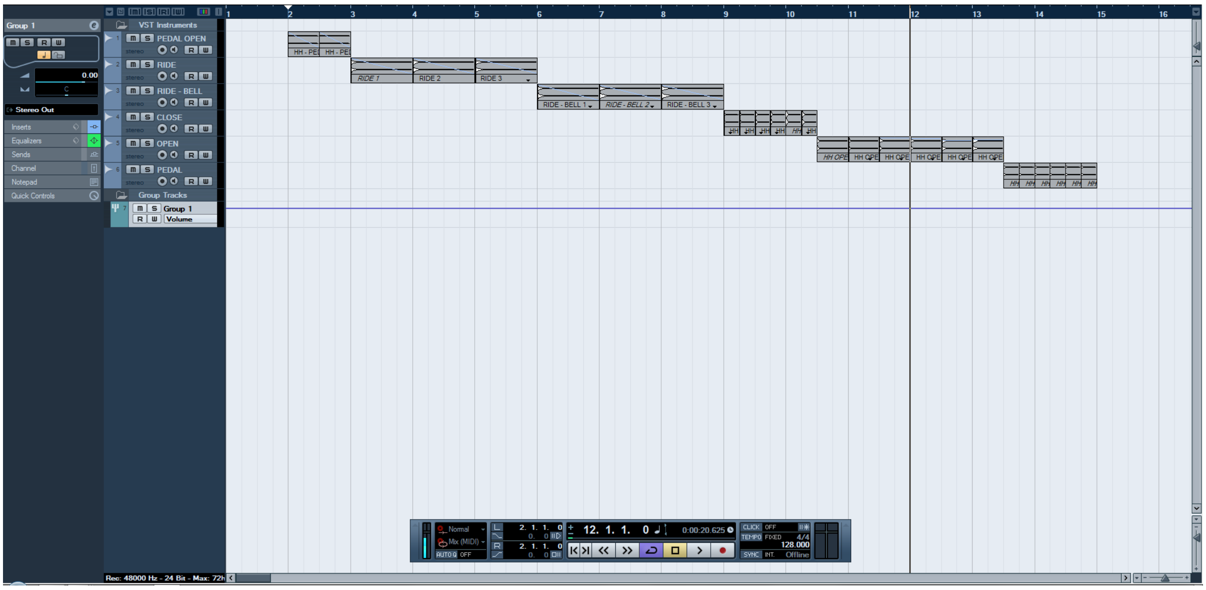


Figure 220: MORALIS, C. (2016) 'Cymbals – Sustain / Duration' [Screen Shot]

Appendix 5

Figs. 221-231 show the volume envelope curves used:

SNARE SUB COMPRESSED BY KICK

Figure 221: MORALIS, C. (2016) 'SC 1' [Screen Shot]

CLAPS COMPRESSED BY KICK

Figure 222: MORALIS, C. (2016) 'SC 2' [Screen Shot]

CYMBALS COMPRESSED BY KICK

Figure 223: MORALIS, C. (2016) 'SC 3' [Screen Shot]

CYMBALS COMPRESSED BY SNARE

Figure 224: MORALIS, C. (2016) 'SC 4' [Screen Shot]

HIHAT COMPRESSED BY KICK

Figure 225: MORALIS, C. (2016) 'SC 5' [Screen Shot]

HIHAT COMPRESSED BY SNARE

Figure 226: MORALIS, C. (2016) 'SC 6' [Screen Shot]

REVERSE – (Synchronized to Quarter Notes according to Ableton Live's BPM)

Figure 227: *MORALIS, C. (2016) 'SC 7' [Screen Shot]*

SNARE COMPRESSED BY CLAPS

Figure 228: *MORALIS, C. (2016) 'SC 8' [Screen Shot]*

TOMS COMPRESSED BY KICK

Figure 229: *MORALIS, C. (2016) 'SC 9' [Screen Shot]*

TOMS COMPRESSED BY SNARE

Figure 230: MORALIS, C. (2016) 'SC 10' [Screen Shot]

TOMS SUBS COMPRESSED BY KICK

Figure 231: MORALIS, C. (2016) 'SC 11' [Screen Shot]

Appendix 6

Fig. 232 shows an explanation of the effects according to the manufacturer's manual:

Double block / tab

Often referred to as "Doubling" or "Double Tracking", the Double effect mimics a singer recording multiple versions of the same vocal passages and playing them back simultaneously. The small differences in timing and pitch that result from the two recordings create a more full and "doubled" sound.
It's quite common in contemporary music for some sort of Double effect to be active during the entire song, albeit with varying intensity.

HardTune block / tab

This effect has become very, very common in recent years. Most people remember when Cher released the song "Believe", featuring the first commercially represented use of heavy and robotic-sounding pitch correction.
Since the release of "Believe", there have been many advances in pitch correction technology, allowing both extreme and subtle or transparent vocal correction. It's a misnomer to only equate vocal pitch correction with AutoTune™ and T-Pain™. Subtle use of pitch correction is a staple of almost every recording, and many live performances too.

Synth block / tab

Synth effects are created when a "carrier" sound modifies a signal (your voice) to create an interesting combination of the two elements.
Common use includes the classical guitar "talk box", where the notes played on an electrical guitar dictate the pitch and sound of the voice output, regardless of what you sing. You don't have to use a guitar though – it could also be sounds from a keyboard or other instrument.

Transducer block / tab

The Transducer effect is often referred to as "Megaphone" or "Distortion", but it really covers any manipulation of gain structure and EQ filters. Distortion and filter effects are common across Rock, Pop, Country, Hip Hop, EDM and other genres.

Vocal µMod block / tab

Pronounced "Micro Mod", the µMod block includes effects like micro-pitch shifting (hence Micro Mod), Flanger, Chorus, Rotor and more.
These effects can be subtle, like "Thicken", or quite extreme with "Tube Up" or "Alien Voiceover".

Choir block / tab

Formerly a part of the Harmony effect block, Choir was split into its own effect in the TC-Helicon VoiceLive Touch 2. Separating these effects gave us the opportunity to provide more styles and parameters for controlling the Choir sound. We have defined Choir as its own effect block in VoiceLive 3 too, so you have extra control over the way it sounds.
Choir makes a great companion to Harmony, especially when you are trying to create a "group" sound. You can use Choir by itself too, which can give its own unique flavor to the vocal.

Vocal Rhythmic block / tab

Rhythmic effects use VoiceLive 3's tempo to chop, break up, pan or otherwise manipulate your voice in time with the music. Depending on how you set the depth and target controls, Rhythmic can be mild or wild and apply to either your lead vocal or the Harmony voices.

Stutter block / tab

Stutter is essentially a small sampler, used to make a quick recording of your vocal and play it back repeatedly, in time with the music. Depending on the division setting, the sample used for the Stutter effect can be longer or shorter.
Stutter is great for Pop, Hip Hop and EDM genres, but can find a home in any style of music when used creatively.

Figure 232: TC Helicon (2017) 'Operating Manual' [Screen Shot]

Appendix 7

Wireless Transmission

Nowadays many guitarists use wireless systems to extend their range of movement on stage. However, wireless technology can have a lot of advantages and disadvantages at the same time. One of the most important advantages, whether through analog or digital transmission, is avoiding Galvanic isolation of the signal, i.e. avoiding ground loops between the transmitter and other electrical instruments on stage. However, since the Performable Recordings model is a production model that aims to bring the high-quality studio sound on stage, it is also necessary to compare analog to digital signal transmission.

Companding

According to Line 6 (2017), a dynamic range of 100db is considered to be a minimum for high-quality audio. However, to fit, for example, within 50db of dynamic range, the signal must be compressed by 2:1. Consequently, when the signal arrives at the receiver, it must be expanded to restore its original dynamic range. This means softer signals are made softer and loud are made louder. According to Line 6 (2017), *'The combination of these two processes is known as 'companding' (a combination of compressing and expanding).'*

However, companding the signal may introduce some sonic artifacts. Consequently, the better the manufacturer's accuracy in matching the time constants and gain control between the transmitter and receiver, the better the audio quality of the transmitted signal.

Fig. 233 shows two diagrams of the signal process from the transmitter to the receiver comparing the analog to the digital systems. Some digital systems may reach up to 115db:

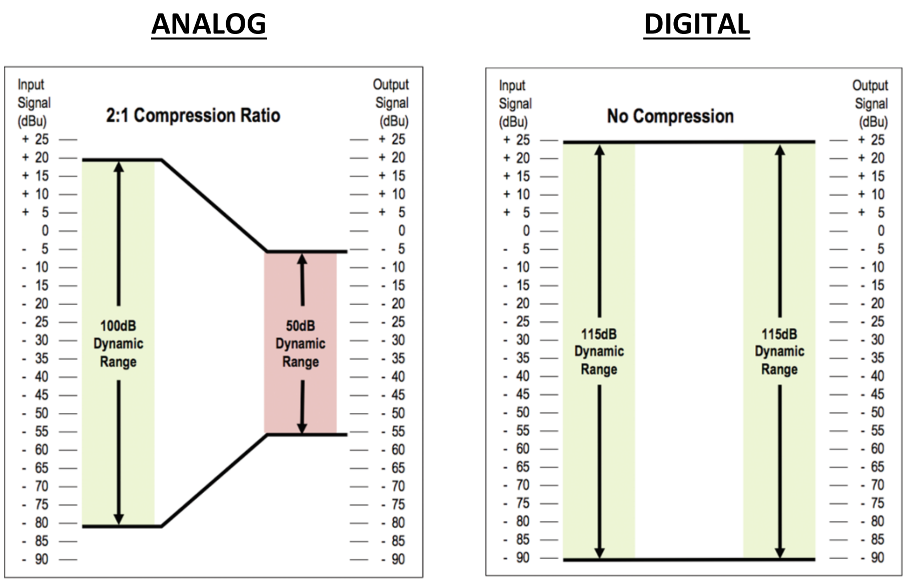


Figure 233: 'Wireless microphones Whitepaper UK' (1) (Line 6, 2017, p. 10)

It is clearly shown that the digital systems can deliver an uncompressed sound resulting in no sonic artefacts or sound interference. The dynamic range is higher, 115db.

Frequency Response

According to Line 6 (2017), '*The frequency response of an analog wireless system is limited at both the low and the high end. On the low end, it is necessary to roll off frequencies that would interfere with the companding circuitry. For example, a frequency of 20Hz is slow enough to cause the gain to change with each cycle of the waveform. Therefore, low frequencies are filtered out. The high frequencies are limited by the constraints of analog FM technology, which typically cannot produce frequencies above 15kHz*.'

Fig. 233 shows two diagrams of the signal process from the transmitter to the receiver comparing the analog to the digital systems. The following first graph shows a comparison between two popular brands.

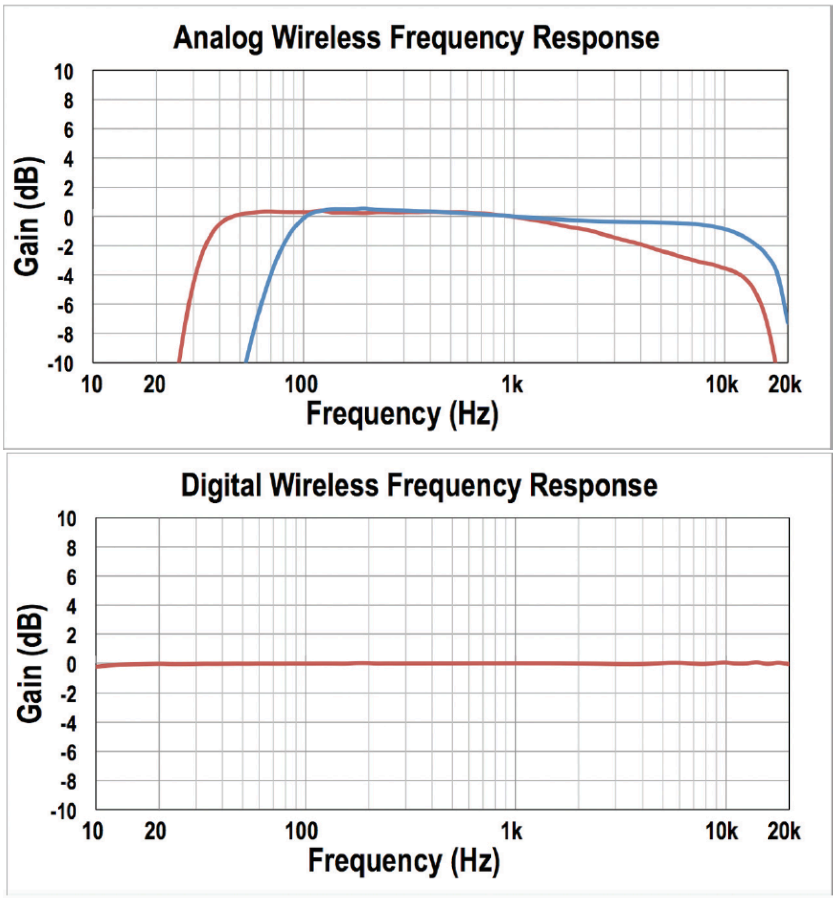


Figure 234: 'Wireless microphones Whitepaper UK' (2) (Line 6, 2017, p. 11)

Again, the digital transmission technology can deliver a better audio signal than the analog one. The frequency response curve is almost linear while the signal carried ranges from 10Hz up to 20,000 Hz.

Distortion

According to Line 6 (2017), '*Most analog wireless systems specify their Total Harmonic Distortion at a level in which the compander is steady, and no overmodulation can occur. In these conditions, the THD specification is typically 0.1% to 0.5%'* and continues, *'In a digital wireless system, the distortion is a function of the overall linearity of the system. There is no compander, nor any audio overmodulation possibility. The signal remains linear throughout the dynamic range, resulting in a typical Total Harmonic Distortion specification of 0.03%, an order of magnitude improvement*'.

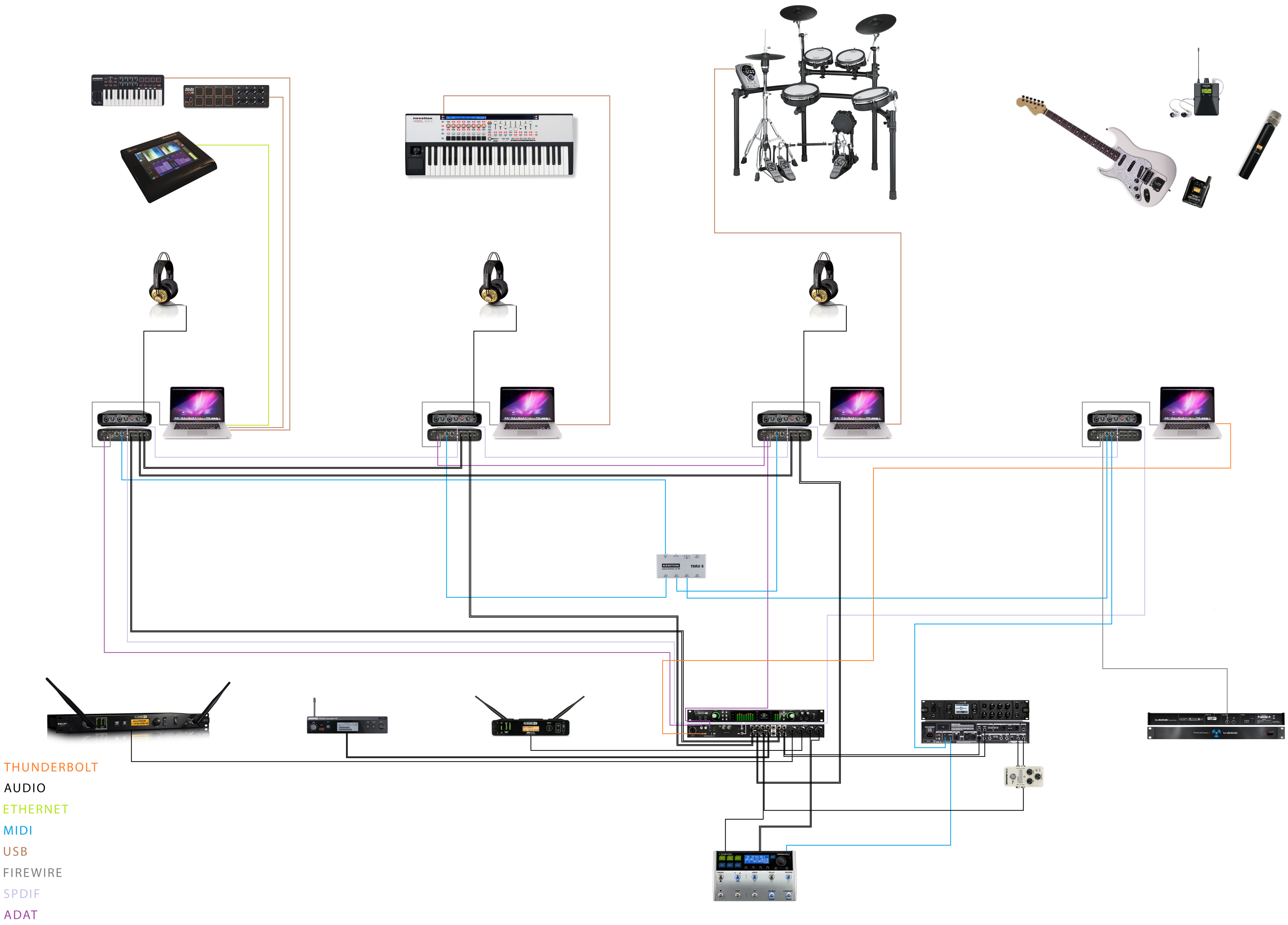
Conclusion

Comparing the digital to the analog transmission, it is clearly shown that the digital audio transmission can deliver high sound quality with no sonic artefacts or noise interfering with the signal. However, the digital signal also involves the further constraint of latency from the AD and DA conversion processes. This may vary from 1.5ms or less, depending on the manufacturer, up to 2 or 3ms.

Since the Performable Recordings model is a real-time process model, latency is one of the most critical factors in the decision making during the sound design and mixing process. The wireless equipment used in this project to send the guitar and vocal signals have the minimum possible latency, which is less than 2.9 ms (audio input to output).

Appendix8

PERFORMABLE RECORDINGS SETUP



Appendix 9

The ADAT (a registered trademark of Alesis) is an eight-track digital tape recorder that caught the recording industry by storm when it was first released in the early 1990s. Today, with over 100,000 ADATs in use in recording facilities around the world, it is the most widely used professional digital recording system. The ADAT was the first product in the category now known as modular digital multitracks (MDMs).

The ADAT system allows up to 16 ADAT units to be used in synchronization, enabling the user to build a very cost-effective multi-track recording environment. The transportability and modularity of the system makes it ideal for mobile recording and wherever space is limited.

Digital transfer between ADATs in a system uses an optical fibre digital communication standard pioneered by Alesis which has become known as Lightpipe. The Lightpipe digital interface has been adopted by other manufacturers as a means of transferring digital data from other types of audio devices, such as mixers, synthesizers, and effect processors.

(whatis.techtarget.com, 2005)

Appendix 10

SPDIF, also written as S/PDIF, stands for Sony/Phillips Digital Interface, and is an interface to transmit digital audio. In this tutorial, we will explain everything you need to know about this interface, including when and how to use it.

Digital audio means that the audio signal is transmitted encoded in a series of 0s and 1s instead of being transmitted in analog format. This makes audio have higher fidelity, because no noise will be added to the audio signal. Therefore, it is always better to transmit audio in digital format.

Currently, there are two consumer-level interfaces to transmit audio in digital format: SPDIF and HDMI (High-Definition Multimedia Interface). SPDIF transmits only audio, but HMDI also carries digital video signal.

(TORRES, G. 2011)

Appendix 11

Link is a technology that keeps Link enabled applications in time over a local network. Link synchronizes musical beat, tempo, and phase across multiple applications running on one or more devices. Applications on devices connected to a local network discover each other automatically and form a musical session in which each participant can perform independently: anyone can start or stop while still staying in time. Anyone can change the tempo; the others will follow. Anyone can join or leave without disrupting the session.

(Ableton.com, 2017*)*

Appendix 12

Potential Acoustic Gain or PAG is the maximum acoustic gain that can be obtained from the system before feedback occurs. For this simplified system (neglecting reverberation and echoes), PAG can be stated mathematically as (34-3) where, Ds is the distance between the talker and the microphone, D1 is the distance between the loudspeaker and the microphone, D2 is the distance between the loudspeaker and the farthest listener, D0 is the distance between the talker and the farthest listener.

(Ballou, G.M., 2008)

Appendix 13

'X-FDBK is a feedback elimination plugin that helps sound engineers to optimally prepare their stage monitors and PA prior to sound check. In the live sound" lingo, this process is called "ringing out." It's usually a lengthy and annoying process. It requires identifying feedback-sensitive frequencies and then cutting them from the stage monitors or PA … X-FDBK dramatically simplifies and speeds up this process by precisely identifying the offending frequencies and cutting them with the exact Q and amplitude. X-FDBK enables you to globally adjust the Q and amplitude, as well as manually add, delete, and adjust filters'.

(Waves.com, 2017)

Appendix 14

Smart: EQ live enables a streamlined workflow for (live) sound mixing. The high-precision adaptive filter of smart: EQ live analyses audio signals, interprets them musically and compensates spectral imbalances in real-time.

(Sonible, 2018)

Appendix 15

Auto-Tune® for Guitar

What is it?
Incorporating world-renowned Auto-Tune pitch detection and manipulation along with other proprietary technologies from Antares®, Auto-Tune for Guitar is an entirely DSP-based suite of functions that offer everything guitarists have always wanted from their guitars, along with capabilities never imagined possible.

From jawless intonation to astonishing tonal edibility to alternate tunings that open up entirely new areas of inspiration and creativity. Auto-Tune for Guitar technology expands the edibility and range of the electric guitar while letting players continue to leverage their own techniques and styles.

What does it do?
Auto-Tune for Guitar offers a variety of functions that greatly enhance the playability and capabilities of the electric guitar. They include:

The Solid TuneTM Intonation System
The Peavey AT-200's Solid-Tune intonation system addresses the eternal challenge of maintaining perfect intonation as a guitarist moves up and down the neck. When a guitar's intonation is even slightly off, nothing sounds quite right—and the effect can cause dissonance and muddy guitar tone. By using the Auto-Tune® for Guitar's Solid-TuneTM intonation system, the Peavey® AT-200TM constantly monitors the precise pitch of each individual string and makes any corrections necessary to ensure that every note of every chord and riff is always in tune, regardless of variables like nger position or pressure. As a result, listening to the Peavey AT-200 with Solid-Tune is a revelation, offering a purity of tone that has simply never before been possible.

Additionally, Solid-Tune is smart enough to know when players intend to manipulate the pitch, so they can play bends and vibrato exactly as they always have. Solid-Tune intonation even makes it easier to bend to the right pitch every time.

Instant String Tuning
With the Peavey AT-200, you can tune all six strings instantly with the push of a button (no motors or gears required). Simply strum the strings, trigger String Tune via the volume knob button and your guitar is instantly in tune.

(assets.peavey.com, 2018)