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Skweeel: towards a method for music production with an automated large-format analogue audio mixing console used as a sound-generating device through controlled Larsen effects

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Skweel – Towards a method for music production with an automated large-format analogue audio mixing console used as a sound-generating device through controlled Larsen effects

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Doctor of Philosophy
I certify that the work presented in this thesis is, to the best of my knowledge and belief, original, except as acknowledged in the text, and that the material has not been submitted, either in whole or in part, for a degree at this or any other university.

I acknowledge that I have read and understood the University's rules, requirements, procedures and policy relating to my higher degree research award and to my thesis. I certify that I have complied with the rules, requirements, procedures and policy of the University (as they may be from time to time).

Philippe Chambin – 22nd February 2019
Abstract

Large-format analogue consoles such as the Neve VR, VX and 88R, or the Solid State Logic SL 4000 G and XL 9000 K, are being relegated to history in all but the orchestral recording market. In studios around the planet, they are being replaced by more affordable all-digital solutions or much smaller format analogue "front-end" consoles.

Since the early 20th century, electronic musicians and producers have been striving for new sounds which set them apart from other artists. The variety of tones that can emerge from a console within a normal recording or mixing session, due to accidental feedback loops, is promising in that regard. Producing these feedback loops deliberately and using them in a totally controlled, creative sound production and live performance, enables exploration of the nature and potential of these phenomena. The console itself acts as a multi-timbral polyphonic synthesiser, with each of its signal paths delivering one sound. Whilst the console’s automation system effectively becomes a sequencer, controlling the occurrence of the sounds, as well as their changes in timbre and pitch.

Within the practice-based research paradigm framework, the purpose of this study is to explore and assess this degree of control over pitch and timbre when using a large-format automated analogue studio console with internally-generated feedback loops. Research was documented via social media blogging and a process diary. The research output includes a public performance and HD video of said performance, and a written exegesis.

The research outcomes suggest that using a large format automated analogue console to produce feedback loop-based music, has the potential to breach the confines in which No-Input-Mixer music has been constrained so far. Thus, not only would studio owners around the world potentially consider their investment in analogue music production equipment under a new light, but so would artists and producers be offered a viable tool in other genres of music production – this repurposing of large format analogue consoles, extending their life well into the 21st Century.
Acknowledgements

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Chapter One: Introduction

In studios around the planet, large-format analogue consoles are being replaced by more affordable all-digital solutions or much smaller format analogue “front-end” consoles (Kirby 2015; Pras, Guastavino & Lavoie 2013). Consoles such as the Neve VR, VX and 88R, or the Solid State Logic SL 4000 G and XL 9000 K, are being phased out in all but the orchestral recording market. Could there be a completely new way to consider these large format consoles, which would give them a new purpose in the music industry, and hopefully extend their life well into this new century? As electronic musicians and producers have been striving for new sounds which set them apart from other artists since the early 20th century (Théberge 1997), it seems that indeed, one very obvious piece of music production equipment has been overlooked as a possible sound source: could the ubiquitous mixing console itself be used as a new type of musical instrument and production tool? Can a large-format automated analogue console be used to produce music, by generating internal feedback loops – thereby transforming the console into a powerful multi-timbral polyphonic synthesizer, with its automation system acting as a sequencer?

I owe my career as an audio engineer/producer to my passion for experimenting with sound using various synthesizers in my early teens. I purchased my first synthesizer in 1984 – a Roland Juno 106 – and as the MIDI norm was just emerging, I was fortunate to have learned both subtractive synthesis components and their interactions, and the MIDI language at the same time (MIDI.org n.d.). I further developed my understanding of electronic music via the purchase of a Yamaha RX15 drum machine and Korg DW8000 synthesizer, alongside an Atari ST 1040, running Pro24 (which would become Cubase). With these, and two tape cassette decks, I set out to follow in the footsteps of my musical influences at the time, which were Saga, Rush, ELP, Marillion, Yes, Genesis, The Buggles, Kraftwerk, Electric Light Orchestra, Nick Kershaw, Duran Duran, Jean Michel Jarre, to name but a few: an eclectic who’s who of keyboardists and electronic musicians, with a predominance in the Progressive Rock genre. Applying for an internship in one of Paris’ music production studios, in the summer of 1985, was the beginning of my career as an audio engineer and producer, where my skills with MIDI-related gear complemented the traditional knowledge of the senior generation of engineers. Since 1990, I have worked on many different large format analogue consoles (SSL, Neve, Amek, Harrison, ADT,
Lafont, Soundcraft, Midas, Allen & Heath,…), allowing me to compare the pros and cons of the various automation systems available at the time: Flying Faders found on the Neve VR, NECAM found on earlier Neves, SSL’s SL 4000 VCA-based automation, SSL’s motorized fader Ultimation automation, GML, and Optifile. Those were also the days of complex synchronisation between the console, two or more multi-track machines, sequencers and drum machines, the 2-track master recorder(s), and the occasional video tape machine, via synchroniser systems such as Timeline’s Lynx, Adams-Smith’s Zeta 3, or the SSL SL 4000’s Motionworker. When one has to make all of these devices and concepts work seamlessly, a great attention for detail, and a strong understanding of control voltages and rudimentary programming and coding skills emerge. Considering such a background, not surprisingly, I have always been intrigued and fascinated by the variety of tones that can emerge from a console within a normal recording or mixing session, due to accidental feedback loops. My intention to produce these feedback loops deliberately and use them in a totally controlled, creative sound production, has allowed me to explore the nature and potential of these phenomena.

The purpose of this study was to explore and assess the degree of control over pitch and timbre when using a high-end analogue studio console and its automation system as a new tool, by preparing then performing a musical piece in front of industry guests and the general public. “New materials and tools lead to fresh compositional strategies…” (Robindoré 2005). The literature review therefore encompasses themes such as deliberate misuse in electronic music, from its pioneers (for example, Pierre Schaeffer, Daphne Oram) to its feedback loop specialists, such as David Tudor or David Lee Myers. Can the control of feedback loop-based electronic music be refined to a point where these techniques can be applied to mainstream music production? Just as DJs in hip-hop extended the life of the vinyl record player, by repurposing it (Daniel, M 2014), I was hoping to provide studio owners with another potential application for their investment in large-format analogue consoles.

1.1 Larsen effects vs. Feedback loops - disambiguation

Whenever the output of a signal path is accidentally or intentionally connected to its own input, it forms a loop known as a feedback loop. Provided the overall gain structure of the signal path produces even the slightest amount of positive gain, the
signal will be amplified each time it passes through the loop, theoretically to infinity. In reality, the signal will quickly stabilise at a level determined by the electronic circuitry having reached the peak of its ability to amplify/handle the signal (Boner & Boner 1965).

Applying the concept to audio signal paths, such as when a vocalist brings their microphone too close to a stage monitor speaker, the resulting feedback loop produces a distinctive loud howling or squeal which, if allowed to develop to its fully-stabilised maximum level, can potentially damage equipment and most importantly, human hearing (Davis & Jones 1989). This particular type of sound resulting from audio signal feedback loops is known as a Larsen effect, named after Søren Absalon Larsen (1871 – 1957), the Danish scientist who first studied the phenomenon (Larousse Encyclopedia 2011; Talc n.d.; Troxel 2005). The entire audio industry in France, French-speaking Belgium and Switzerland, and French-speaking former African colonies, all use the term “Larsen” to designate a Larsen effect produced by a feedback loop – except in Quebec (J Obadia 2018, pers. comm., 13 July). The use of the term Larsen in English literature is more anecdotal; for instance, I could find no mention of that word in any of David Tudor’s or David Lee Myers’ documented work, both being pioneers in the field of producing music using Larsen effects inside electronic equipment. One of the few scholars who specialises in research based on Larsen effects, Agostino Di Scipio, refers to these sounds either as “Larsen tones” (Placidi 2010) or “Larsen effects” in his 2003-4 Audible Ecosystemics sound installation (Di Scipio 2003).

Artists such as Terry Riley in Music for the Gift (Riley 1963) or Brian Eno and Robert Fripp in No Pussyfooting (Eno & Fripp 1973) having popularised the use of feedback loops incorporating a tape recorder (later replaced by a digital delay line) to produce complex layered polyphonic and rhythmical patterns, the term ‘feedback loop’ has become ambiguous over time. A Larsen effect, on the other hand, strictly refers to the direct, non-delayed manifestation of a feedback loop: the loud, and most often unpleasant howl or squeal described by Søren Larsen. As my research focuses on that particular type of feedback loop, to avoid any ambiguity, the term ‘Larsen effect’ will be used instead throughout this research.
1.2 Research question

This research asks: what potentials exist for producing music using a large-format automated analogue audio console as a sound-generating device, via controlled Larsen effects triggered with internal feedback loops? The outcome of the research may have implications for the artists, music producers and studio owners around the world, who are using or have invested in expensive large-format automated analogue consoles. The following sub-questions were derived:

- How would the hardware and software limit/restrict the creative outcome and its aesthetics?
- Which signal paths produce which kind of tone?
- How much control of that tone could be obtained using the console's circuitry and on-board processing (EQ and Dynamics)?
- Would the console’s automation system time base be precise-enough to generate rapid, steady rhythmical patterns, akin to using a sequencer or drum machine to produce 16th or 32nd notes?
- Would the console’s automation system allow precise-enough control of pitch, or would I only be able to compose a piece using noise and textures, much akin to some of Merzbow’s work in the past decade (Doornen 2006)?
- Would the console's Total Recall™ capabilities be precise enough to provide repeatability, or would a considerable amount of manual tweaking be necessary before each performance of the piece?
- Would I have to develop my own symbolism to compose and sketch ideas on sheet music, or could I rely on existing notation systems, such as the ones used by Pierre Schaeffer as early as 1948 for his Musique Concrète (Hodgkinson 1986), Stockhausen’s and Xenakis’ as seen in the literature review, or more recently, Aaron Cassidy for his String Quartet work (Rutherford-Johnson 2011)?

* Total Recall can store all the positions of the console's knobs and switches, so that these settings can be retrieved later on, for another session.
1.3 Organisation of the study

Chapter two reviews the available literature pertaining to the research, focusing on relevant composition and process precursors. Chapter three presents the research design and methodology: how are the research question and sub-questions addressed, what are the boundaries of the research, and what are the various stages necessary to produce the research output? Chapter four presents the data collected during the investigative stages of the research. The findings are organised into sub-chapters, each dedicated to one of the research stages, compiled from the research diary found in Appendix A. Chapter five presents the data collected during the compositional stages of the exemplar musical piece. This covers the aspects of scoring, preparing the studio for a performance, and the performance itself. This information is also compiled from the information gathered in the research diary, found in Appendix A. Chapter six presents the study’s conclusions on timbre exploration, the console’s ability to store and recall timbres, and its effectiveness when used as a sequencer. The composition of the piece is discussed, in terms of whether or not it has met the benchmarks set in the research methodology. The efficiency of the peer feedback system is weighed using online statistics. Areas for further study are outlined, along with directions and implications.

Appendix A is the research project’s diary: it presents data collection in chronological order, each chapter covering one research session. The purpose of the session, its processes and an analysis of the findings, are presented along with graphs and diagrams. Appendix B is a glossary of technical terms found in the exegesis. Appendix C, D and E are the questionnaires and consent form handed out to participants in the public performances of the project’s output. Appendix F contains the operator manuals for both the SSL XL 9000 K and the SSL Duality SE consoles, used in the research project. Appendix G presents a set of tables listing all of the feedback loop signal flow configurations achievable on the SSL XL 9000 K and SSL Duality SE consoles. Appendix H is the research project’s timeline, as it was originally drafted in the early stages of the project’s methodology design. Appendix I and J are the files that were used to set up the studio before the performance of the piece, and the actual performance files, respectively. Appendix K is a set of spreadsheets linking timecode formats to various ideal tempo values. Appendix L contains the research project’s survey results.
The Research Video Files folder contains all of the videos which were produced as part of the research output – totalling more than 3 hours and 40 minutes. Episodes are numbered in chronological order. The main project output – the performance of the project’s composition I Pressed Play – is highlighted in green. Important note: in October 2015, the candidature was transferred from a Masters by thesis to a PhD. Hence, up to Episode 03 included (which was produced prior to the transfer), you will hear me refer to my work as a Masters research project.
Chapter Two: Literature review

This chapter explores the relevant literature in terms of the following themes: deliberate misuse of electronic devices, composing with Larsen effects, scoring, performance techniques, and repurposing. Each of those themes is investigated via a few of its key protagonists, starting with Luigi Russolo, Pierre Schaeffer and Daphne Oram. When undertaking research which involves purposefully using a piece of music production equipment in a way that the manufacturer had not intended, reviewing the work of notable precursors can provide valuable insight into the process. Indeed, this deliberate misuse can be found throughout music history (Taylor 2001). As the study focuses on the use of an electronic piece of audio production equipment, the following sections examine the electronic era of music production, from the early 20th century onwards. Likewise, available knowledge on the aspects of performance, repurposing, and scoring, focus on contemporary methods pertinent to the research subject’s epoch.

2.1 Luigi Russolo: an introduction to noise-based music

In a letter to a friend entitled “Dear Balila Pratella, great Futurist composer” (1913), Russolo lays the foundations for music based not on pitch produced by the traditional orchestral instruments, but on noises emerging from the 19th century industrial revolution. In it he describes how human hearing has evolved to appreciate noise and its musicality:

To excite and exalt our sensibilities, music developed towards the most complex polyphony and the maximum variety, seeking the most complicated successions of dissonant chords and vaguely preparing the creation of musical noise. This evolution towards “noise sound” was not possible before now. The ear of an eighteenth-century man could never have endured the discordant intensity of certain chords produced by our orchestras (whose members have trebled in number since then). To our ears, on the other hand, they sound pleasant, since our hearing has already been educated by modern life, so teeming with variegated noises. But our ears are not satisfied merely with this, and demand an abundance of acoustic emotions (Russolo 1913).

He goes on to describe the “6 families of noises” of his proposed “Futurist orchestra” and concludes on this final note: “We therefore invite young musicians of talent to conduct a sustained observation of all noises, in order to understand the various rhythms of which they are composed, their principal and secondary tones” (Russolo 1913). A book developing these ideas was published in 1916, The Art of Noises
(Russolo 1916). The artist then set out to design and build musical instruments that would allow the generation and performance of noises. By the 1930s, Russolo had built twenty-seven different varieties of his *Intonarumori* (noise-makers), which were acoustical contraptions based on a resonant cavity coupled with a horn. Inside the resonant cavities, various systems would create friction or impact against strings, membranes, etc. (Brown 1981). Russolo’s concept on noise-based music and his work were an inspiration for Schaeffer and Henry (Warner & Cox 2004), as well as Trevor Horn and Paul Morley’s sample-based band *Art of Noise* – to the point of naming the band after Russolo’s manifesto (Morley 2002).

### 2.2 Pierre Schaeffer and Musique Concrète

When national radio broadcasting organisations around the world invested in new technology (tape recorders in the ‘50s for example), this encouraged new experimentation by their engineering teams. Pierre Schaeffer is regarded as the pioneer in that field (Battier 2007). In 1936, Schaeffer joined the French national broadcasting organisation, which would become the ORTF (Office de Radiodiffusion Télévision Française). His experimentation with the equipment led to the publication of various articles and theoretical works, most notably, *A la Recherche d’une Musique Concrète* (In Search of a Concrete Music) in 1952 and *Traité des Objets Musicaux* (Treatise on Musical Objects) in 1966.

With *Etude aux Chemins de Fer* (Railroad Study) in 1948, Schaeffer laid the foundations of Musique Concrète by taking a radio sound truck to the Batignolles train station to record locomotives and assemble a collage. In an interview led by Hodgkinson (1986), Schaeffer explains one of the “three circumstances that compelled me to experiment in music”:

> Thirdly, after the war, in the ‘45 to ‘48 period, we had driven back the German invasion but we hadn’t driven back the invasion of Austrian music, 12-tone music. We had liberated ourselves politically, but music was still under an occupying foreign power, the music of the Vienna school.

With *Etude aux Chemins de Fer*, the main focus was to use existing sounds, or noises, but in a way that would strip them of their original signification or implied emotion, returning those audio signals to a purer, musical form. This was mostly
done with manipulation of acetate discs*. Soon after, the techniques would be transposed to tape recording and/or intentional misuse of the tape recorder itself (Teruggi 2007).

Schaeffer’s investigations into Musique Concrète would later be consolidated in 1951. Joined by another pioneer, Pierre Henry, Schaeffer founded the *Groupe de Recherche de Musique Concrète*. The research group would later be re-baptised *GRM* (Groupe de Recherches Musicales) in 1958, with contributions from Pierre Boulez, Edgar Varese and Iannis Xenakis (Battier 2007). Ample evidence can be found regarding their use of tape machine feedback to loop particular sound patterns, with devices such as the morphophone which was designed specifically for that purpose, as described by Teruggi (2007):

The *morphophone* permitted formal-type modifications by accumulation of events, filtering and reinjection. It consisted of a turntable on which was set a loop of magnetic tape. Around the table was a unit comprising several heads, whose positions were adjustable. The heads were as follows: an erasing head, a recording head, and ten playback heads. A sound was recorded on the tape, either in a loop or continuously, and the ten playback heads could be adjusted to provide different delays. The sounds could be filtered and re-injected to obtain artificial reverberations or continuous threads.

The literature, however, holds no evidence of pure electronic Larsen effects as a means to produce sound in the few available sources that truly describe Schaeffer’s (and his team) techniques in detail.

### 2.3 Daphne Oram and Musique Concrète

Following a visit to the (O)RTF in the early ‘50s, British composer Daphne Oram set out to develop similar artistic investigations into electronic manipulations at the BBC which would lead to the creation of its Radiophonic Workshop in 1958 (Worby 2008). Less than a year after the department’s inception, Oram resigned from the BBC in January 1959, “unhappy with the music department's lack of interest” (Davies 2003). Oram’s focus was to develop a graphical interface, “with which the composer could ‘convert graphic information into sound’, bypassing the more laborious methods of cutting and splicing tapes used in electronic music studios” (ibid). This developed into the Oramics system in 1962, where “the composer draws onto a synchronised set of

* See the glossary in Appendix B for the definition of “acetate disc” and other technical terms.
ten 35mm film strips which overlay a series of photo-electric cells, generating electrical charges to control amplitude, timbre, frequency and duration” (Daphne Oram.org 2014). Once again, the literature provides no definitive evidence of pure electronic Larsen effects as a means to produce sound in the few available sources that truly describe Oram’s (and her team) techniques in detail.

Although the original approach of “internally-generated Larsen-based” music follows the tradition of Luigi Russolo’s “noise-sound” (Russolo 1913) in that the sounds produced are mostly noise-based, part of my research’s goal is to explore whether or not the tool, if advanced enough, can successfully be used to produce pitch-based music.

2.4 Max Neuhaus and Robert Ashley’s pioneering use of acoustic Larsen effects in music

1964 marks the earliest documented works of electronic music based on deliberately setting off acoustic Larsen effects. In the liner notes that accompany the *Fontana Mix – Feed* CD from Alga Marghen, Max Neuhaus explains that a year earlier, in 1963:

“While exploring ways of changing the timbre of percussion instruments through amplification, I had discovered a means of generating sound which I found fascinating – the creation of an acoustic feedback loop with a percussion instrument inserted inside it. Instead of the usual single screeching tones of acoustic feedback, this created a complex multi-timbed system of oscillation (Neuhaus 2003).

He began touring in the USA and Europe with a system based on four feedback loops, using four contact microphones resting on percussive instruments. For instance, as reported by Theodore Strongin of the New York Times, Timpanis were used for performances in 1966 (ibid.). The microphones’ signals pass through a sound mixer controlled by Neuhaus, varying the amount of amplification for each of the channels using John Cage’s *Fontana Mix* scoring system*. The outputs are sent to loudspeakers in the room. Instead of randomly altering the score as per the original concept of Cage’s *Fontana Mix* system, during a series of performances,

* Ten pages covered with six curved lines each, and ten sheets of transparent film covered with randomly-placed points. In accordance with a specific system, and using the intersecting points of a raster screen, two of the pages produce connecting lines and measurements that can be freely assigned to musical occurrences such as volume, tone colour, and pitch (Sauerwald n.d.).
Neuhaus would use the exact same score to control the volume changes (Reddy brown objects 2015). This would still yield different results as the many remaining variables would be enough to drastically change the feedback loops’ interactions: different venue acoustics, instruments, speaker position and even each audience members’ seating arrangement. As noted by Neuhaus though, even if the exact same score is used twice in a row in the same room, with the same instruments, speakers and audience, the result would be different, as the loops interacting with each other form an incredibly complex and unpredictable system; not only because the vibrations would sometimes cause the microphones to change their position on the percussive instruments. “It seems something alive” (Neuhaus 2003).

Early in that same year of 1964, Robert Ashley composed *Wolfman*, which was then performed for the first time in the fall on Charlotte Moorman’s festival of the avant-garde (Lovely.com n.d.). The piece is based on a magnetic tape collage of found sounds and AM radio recordings (Padua 2015)– *The Wolfman Tape* – which is then played during the performance through a loudspeaker, through which Ashley’s vocals are also sent. The amplification of the voice coming through the loudspeaker is such that a Larsen effect develops. “Ashley’s notes clarify that the vocalist in fact had to perform softly, otherwise the feedback would be blocked out” (ibid.). The sounds on the tape are selected and edited to form “a blizzard of very short sounds across the whole frequency range-so that the illusion of the sounds coming from all parts of the room is preserved” (Lovely.com n.d.).

### 2.5 Steve Reich and acoustical Larsen effects

Another early and very well-documented example of the use of Larsen effects in noise-based music, is a 1966 piece by American composer Steve Reich: *Pendulum Music (For Microphones, Amplifiers Speakers and Performers)* (Reich 1966). Three or more microphones are hung from various heights using their own cables. Each microphone is connected to an amplifier and speaker, the latter being positioned on its back, under its respective microphone, therefore pointing straight up towards it. During the setup, before the audience arrives, a Larsen effect is triggered between each mic/speaker tandem, by setting the amplifier at a sufficient gain level. The amplifiers are switched off. To begin the piece, performers pull back the microphones (at which point an operator switches on the amplifiers) and intentionally release them slightly out of sync. One should note that this is also a good example of misuse. This
asynchronous release, along with the different cable lengths, guarantees a random evolution of Larsen effects, as the microphones swing by their respective speakers. The entire system takes between three to ten minutes to stabilise, depending on the initial setup parameters (Van Eck 2017, p. 88). "The piece is ended sometime shortly after all mikes have come to rest and are feeding back a continuous tone by performers pulling the power cords of the amplifiers" (Reich 1974).

This particular piece makes full use of the link between vision and hearing - listening through it purely on speakers or headphones, without seeing the preparation and the microphones swinging, would arguably take away more than half of the entire experience. In Reich’s own words, "If it's done right, it's kind of funny" (2000). Just as a classical music aficionado attends concerts in the hall in part because of the added visual stimuli, how much of an audience’s ability to engage with noise-based music comes from the actual performance: seeing the setup and the artists' interaction with the instrument(s)? As one of my goals was to spread awareness to the general public on the possibilities of Larsen effect-based music, this is an aspect which I have kept in mind when staging the performances of the piece. Contrary to Reich’s approach, I aimed to develop consistency and precision in the generated sounds, by eliminating as many random aspects as possible in the performance, letting the console’s automation computer handle the changes whenever possible.

2.6 Other examples of acoustical Larsen effects in music

A review of literature focusing on Larsen effects would not be complete without the mention of Jimi Hendrix’s immense contribution to its popularity in music-making, most notably via his rendition of *The Star Spangled Banner* (Hendrix 1969) at the Woodstock Music Festival. As early as the 1950s, various other artists had used acoustic Larsen between their guitar and its amplifier, but none in such an openly deliberate and exaggerated way, as Hendrix pushes the effect to a point where the guitar tone has infinite sustain, and harmonics are generated that would otherwise be impossible to play. This has opened the door to an entire new range of guitar sounds and guitar playing. As noted by the Rock & Roll Hall of Fame Museum: "Jimi Hendrix expanded the range and vocabulary of the electric guitar into areas where no musician had ever ventured before" (Rockhall.com n.d.).
In 1969, Alvin Lucier devised *I am Sitting in a Room*, where a tape loop of human speech is fed to a loudspeaker, the resulting sound recorded onto a second machine, feeding back into the first. After fifteen generations for the 1970 version of the piece, and thirty-two generations in its 1980 version, the room’s natural resonant frequencies have completely overpowered the speech. Although not a true Larsen effect as per its definition, the concept of sound accumulation using a speaker and microphone is being used (Collins, N 1990). In 1975, Lucier designed another piece using acoustic Larsen effects: *Bird and Person Dyning* – for performer, binaural microphone, amplifiers, loudspeakers and electronic birdcall (Van Eck 2017, p. 91). Lucier meant it spelt that way, as an abbreviation for heterodyning – a phenomenon where a sum and difference signal are generated from two signals mixing together. It all started when he received an electronic gadget in the mail: a Christmas tree ornament shaped like a bird, which would emit an electronic birdcall via a speaker inside it. Donning a pair of binaural microphones (the type that have the capsules resting just outside of the ear canal, kept in place between the tragus and antitragus), he had the birdcall playing, and would walk around the room with the microphones connected to amplifiers and loudspeakers. By controlling his head movements and evolution around the room, he could control various aspects of the generated Larsen effects, including how those extra sum and difference frequencies would appear. “I heard ‘heterodyning’ – difference between the two frequencies – between the pitches of the feedback and the bird call. It was fascinating, seemed to happen in my ears” (Viola & Harder 2013).

Likewise, some of Nicholas Collins’ early works use the room acoustics itself as an additional component of the resulting sound, in what he calls “improvising with architecture”. This is exemplified in his 1974 piece, *Pea Soup*, where acoustic Larsen effects between microphones and speakers have their pitch constantly altered by three phase shifter circuits. The performer can move around the space to influence the build-up of the Larsen effects, or play various instruments to supply new audio material for them. (Collins, N n.d.). In 1978’s *Niche*, Collins expanded his concept of Larsen effects interacting with architecture, in the form of a large tent pitched above the performer, covering as much of the performance area as possible. A system of several pulleys and ropes allow the performer to pull various sections of the canvas up towards the ceiling as he/she moves under it. A contact microphone on one of the pulleys provides the initial audio signal to form the Larsen effect. This microphone
signal is combined with that of a directional microphone placed on the floor, pointing up to the tent “ceiling”. Both microphones feed into compressors and limiters, then into a tape machine to provide an element of delay in the feedback loop. A high-frequency driver placed on the floor, facing up towards the tent “ceiling” broadcasts the tape machine’s output signal. “The response of the tweeter limits the feedback to mid-range and high frequencies only. At these frequencies canvas acts like a solid wall which reflects and absorbs sound”. In the piece’s score, the performer is given a variety of instructions regarding how they should move, and interact with the ropes and sounds they hear (Collins, N 1978). A more straightforward piece from 1975, Feedback, has brass or woodwind instruments equipped with microphones in the mouthpiece, feeding speakers placed on the floor. The score indicates that the long microphone cables should have their excess coiled “neatly in the center and leave the ends unmarked, so that none of the performers can tell which cable leads to which speaker”. The performance involves another element of playfulness, as the musicians must take a step towards their speaker, and try to cancel their Larsen effect by any means, but only if no other member of the group is feeding back. The performance ends when stepping forward is no longer possible for anyone (Collins, N 1975).

One last example of the use of acoustic Larsen effects is Pouvoir d'Achat's Larsen Forever, where vocals, guitars and even turntables are triggering acoustic Larsen, contributing to the final sound (Talc n.d.). Incidentally, the name of that particular song highlights the use of the word ‘Larsen’ in nearly all French-speaking communities, to depict the sounds produced by audio feedback loops whose overall gain is greater than 0 dB.

2.7 David Tudor and early works in electronic Larsen-effect-based music

David Tudor was a pianist, composer, and long-time collaborator with John Cage (1961). “His collaborations with Cage for the latter’s Variations II (1962) and Variations V (1964) would also nourish Tudor’s later experiments with amplified small sounds, acoustical resonances, and feedback processes” (Holzaepfel n.d.). Through these collaborations, including Cage’s Project for Music and Magnetic Tape in the early ’50s, Tudor became interested in the potential for live electronic music. In the mid ’60s, he teamed up with Lowell Cross (ibid.), for whom he participated in the elaboration and performance of Bandonon!, performed as part of 9 Evenings:
Theatre and Engineering, in the 69th Regiment Armory in New York, from October 14 to 18, 1966 (Bonin 2006). This piece uses a bandoneon fitted with contact microphones, played by David Tudor, whose signals are then sent to audio processors and video/light processing apparatus. The electronically-modified audio signals were sent to twelve speakers spread around the large armoury, some of which are mounted on remote-controlled carriages which are promenaded around the hall by their human operators, varying the way in which the sounds reverberate around the room. To enhance these effects, other remote-controlled carriages support reflectors which intersect with the speakers (SensoProjekt 2011). Additionally, Tudor’s bandoneon is fitted with various switches, one of which mutes the sound to allow the audience to better hear the reflections off the room’s architecture (Bonin 2006). Through this experience, Tudor came to the realisation that loudspeakers “[could] become instruments, ‘sound sculptures’ with individual characters and personalities like conventional instruments.” More importantly for this research, he also realised “the intriguing possibility of generating sound without an external source of input” (Holzaepfel n.d.). Tudor went on to explore his first idea, when he was commissioned along with Robert Breer, Frosty Myers and Robert Whitman, to design the Pepsi Pavilion at Osaka’s 1970 World Expo. With Gordon Mumma, he conceived a thirty-seven-channel sound system, with speakers arranged around the pavilion’s dome ceiling (Composers Inside Electronics n.d.). “So there were actually thirty-seven loudspeakers, which could be programmed to have eight different spatial patterns, if you wanted to do that” (Fullemann 1984). Tudor experimented further. Connecting them in various ways, he managed to get the system to generate sound with no input signal; “So I chained them together in various ways an lo and behold, there they were, oscillating” (ibid.).

In 1973, Tudor founded the Composers Inside Electronics group with John Driscoll and Phil Edelstein (Driscoll n.d.). The group’s focus, starting with Tudor’s Rainforest project, was to promote the composition and collaborative performances of electronic music. Matt Rogalsky, one of its current members, has had access to Tudor’s collection of electronic devices, and set out to catalogue them in July 1996* (Rogalsky n.d.). In that article, he goes on to describe some of the units: “Most boxes had numerous connection points and knobs, switches, etc, but if any of

* The article states July 1966 erroneously in the first sentence. The article makes it clear that this was weeks before Tudor’s death, meaning it should read 1996.
these was labelled it was usually with some cryptic mnemonic code that it seemed only Tudor could interpret”. Rogalsky also notes that not only was the labelling obscure, but so was the actual way in which Tudor might use the units:

It seems that Tudor’s use of electronics did not depend on rules of ‘proper’ use but rather on intuition informed by deep background study in the subject; and there are many accounts of how he employed circuits ‘backwards’ or made use of the chaotic behaviour of failing components (ibid.).

Rogalsky traces back the first use of electronic feedback loops to generate Larsen effects in Tudor’s music, as early as 1969, implying that these techniques would have been used in the Pepsi Pavilion at the 1970 Osaka Expo. The davidtudor.org website has Rainforest as composed in 1968 in its Chronology of works page (Adams n.d.). “Possibly as early as 1969 Tudor was employing principles of electronic feedback to generate sound ‘spontaneously’, i.e. without any external input”. Definite proof of purely electronic Larsen-based signal generation can be established in 1971, as Rogalsky had been able to identify some of the units used in Tudor’s Rainforest setup as being “… little amplifiers; Tudor made use of them as sound generation devices by connecting outputs back into inputs, allowing the noise in the system to blossom into controllable chaos: whoops, chirps and irregular rhythms”. The first official composition for what would later be known as “no-input mixer”, was 1972’s Untitled, his last collaboration with John Cage. Tudor’s system diagram in Figure 1 shows that a tape machine is part of a complex arrangement of amplifiers, equalisers, and high-pass and low-pass filters, combining their outputs in multiple ways, to achieve what Tudor called “formant shifting” (Kuvila 2004). By controlling the filters’ cutoff frequencies, resonant frequencies could be formed which would dictate the pitch of the feedback signals. Rogalsky’s analysis of Tudor’s notes provides further explanation of the tape machine’s purpose. To provide a greater variety of sounds, and minimize the amount of equipment needed when touring (as there are sixty components in total), the tape machine could be used to store the sounds produced by the “Source Generation” system from Figure 1, which could then be played back at random, to excite the exact same setup, which would now become a “Performance Processing” stage, during the actual performance (Rogalsky n.d.). Tudor provides further explanation as to why he resorted to using a tape machine for Untitled instead of generating all of the sounds in real time, in an interview he did in Stockholm in 1984:
I mean the two oscillations coming there, each with their associated controls, and trying to handle that and then trying to handle the output... it was not that it was too complicated, it was simple [sic] too difficult. I mean, after all, feedback is feedback. And even though I’m good at isolating the output, in the end there is always the possibility that it will take off (Fullemann 1984).

In 1975, Tudor simplified the setup he used for Untitled, eliminating the tape machine altogether to form a piece which was 100% based on internal feedback loops: Toneburst. By adding a few components to the output of the system, he was able to increase the variety of sounds produced. “it added output processing so that the same material was being processed in different ways” (ibid.).

David Tudor would go on to design the equipment and perform many more Larsen-based pieces such as Pulsers (1976), Dialect (1984), various additional versions of Rainforest (1972, 1973 & 1976), … and his career as an electronic musician took him around the world, from the Royal Albert Hall to Tokyo and Australia (Adams n.d.; Fullemann 1984).
2.8 David Lee Myers and the Feedback Workstation

Working under the name Arcane Device, David Lee Myers is perhaps the most famous earliest experimentalist in the art of generating Larsen effects via feedback loops in electronic circuitry, before Nakamura (Meyer, W 2003) and the Japanese noise-music scene coined the “no-input mixer” terminology to identify that specific sub-genre (Myers n.d.-b).

I will concede to being a pioneer in this area, but certainly David Tudor beat me to the punch as it were by seventeen years. In 1970 he used feedback principles in producing his music, although this played a relatively small part (DL Myers 2018, pers. comm., 18 July).

A review of Myers’ 1988 album, Engines of Myth, by Curtis Karnow (2005) states that “Myers makes closed electronic feedback loops, and so even the Moog keyboard violates the implied rule: No outside input”. “Myers fabricates machines, including the Feedback Workstation, which he used to make most of the Arcane Devices’ tracks”. The liner notes on the CD re-issue of Engines of Myth and Improvisations for Feedback indicate: “Feedback Music recordings produced on the inputless “Feedback Machines”” (Arcane Device 1988). Myers’ website, Pulsewidth.com (n.d.-b), has a page dedicated to the ‘devices’ he used since 1987, and allows us to see the progress from conceptualisation, as seen in Figure 2, to having designed and built his own apparatus – the pinnacle of which is the “Feedback Workstation”, seen in Figure 3. “Almost everyone taking the feedback road has used mixing boards alone. I differ in that I have always focused on forcing audio processors to feed back via simple mixing arrangements” (DL Myers 2018, pers. comm., 18 July). The exact list of elements inside the racks is not available, but the website has some information about the content: “Based on 4 digital delays. Included are custom mixer with feedback matrix, a smaller submixer, two DIY ring modulators, a nutty timeclock generator/enveloper/delay modulator, etc.” (Myers n.d.-b). The digital delays are RDS 7.6 Time Machine units made by Digitech, which offer a maximum of 7.6 seconds delay time, as well as sampling capabilities (Loopers-Delight.com n.d.). Another description of the machine, found on Starkland.com, indicates:

The exceptionally elaborate unit typifies his method of expanding the functions of off-the-rack units by tearing them apart, routing controls for each parameter of each device to a front panel potentiometer, routing the output of each device to every other device, as well as to itself, and routing all controls to his own front panel concoction of patch bays, switches, and faders. (Visconti n.d.)
The use of delay lines as part of the feedback loop, will not generate Larsen effects, allowing instead for the looping of sections of sound to form patterns, as can be heard in *Implicate Order*, on the Ether Music album (Myers 2017). One can suspect that should the delay time be reduced to its minimum value, which is usually one millisecond on most digital delay units to mask the fact that the A/D and D/A conversion stages are not instantaneous, then this would generate timbres close to the Larsen effects of No-Input Mixer music, via the cumulative build-up of sound. Such timbres can be heard with added reverberation, on *Arabic Science* or as drones and varying delay-timed loops, on *Dorsal Streaming* – also from *Ether Music*. 
One can also imagine that the delay units can be removed from the feedback loops, to allow true Larsen effects to be used.

The “About” section of his website, gives us an insight into Myers’ creative philosophy:

Essentially, I do not create sounds or compose, but allow latent or unseen forces and processes to present themselves via simple technologies. I select the methods, set the stage, and as the phenomena emerge I of course introduce my own aesthetic judgements to the mix (Myers n.d.-b).

A philosophy to which he has adhered to this day:

I watched your video* and must say that we certainly see this musical approach differently! It would never occur to me to try and control feedback with such technical precision. In fact, its natural uncontrollability is its biggest charm for me - I couldn’t have done this for thirty years otherwise (DL Myers 2018, pers. comm., 18 July).

2.9 1990s Japanoise and No-Input Mixer music

Japan has been at the forefront of noise-music since the '90s (Novak 2013). The Japanoise movement has inspired artists to experiment in many different directions, such as Merzbow’s explorations into distortion presented in 2012 at Brisbane’s Centre of Contemporary Arts (Bneart.org 2012). One of the by-products of this diversified noise-music exploration was the creation of a new sub-genre consisting in generating Larsen effects inside an audio mixer’s circuitry. This particular type of method and performance became known as “No-Input Mixer” music (Meyer, W 2003).

Toshimaru Nakamura, who produced No-Input Mixing Board and other albums based on the console being the instrument, is one of the genre’s most influential artists. As William Meyer (2003, p. 1) gathered in his interview:

Nakamura started out playing guitar, but in 1997 he put it aside and began playing the mixing board. He jacks the board's output into its input to create a feedback loop which, by judiciously twisting a knob here and there, he tweaks into sounds ranging from piercing high tones and shimmering whistles to galumphing, crackle-spattered bass patterns.

* Myers is referring here to the video of my PhD project's main output which is the musical performance: I Pressed Play.mp4.
† The Brisbane Art Guide has the event located at the IMA (Institute of Modern Art). I was at that concert; it was definitely at the Centre of Contemporary Arts.
A video interview (SamadhiSound 2010) as well as a few liner notes on his CDs (2014), imply a rather small mixer - eight to twelve channels at the most - as the sound source for his compositions. For instance, in *The Improvisation Meeting at Bar Aoyama*’s liner notes (1999), Nakamura states:

> I play the no-input mixing board which allows me to control internal feedback, produce loops, melodies and so on. Basically, these three and a couple of guest performers play at the bar once a month. Five altogether seems to be the maximum due to the size of the room.

Also, on the web page dedicated to the *Egrets* CD (Nakamura 2010), he explains that “I work with analogue audio feedback, utilising inexpensive gears, so it is very difficult or impossible to control perfectly.” The use of a small-sized console is a key aspect of what has defined, and limited, No-Input Mixer music thus far. The live performance imperative requires having equipment that is easily transportable. A portable analogue sound mixer will have limited features, yielding limited control over pitch and timbre.

### 2.10 Other examples of No-Input Mixer music

Another key artist in the No-Input Mixer music stage, is Croatian composer Marko Ciciliani (n.d.), who turned to performances based on a small no-input mixer in the late 2000s. The section regarding the performance side of his work on his website reveals that:

> For more than 10 years Marko Ciciliani's primary performance medium was the mixing board 'sans instruments', using only feedback from the machine itself. He has made a reputation of being the virtuoso of the abstruse practice of playing a no-input mixer. Unlike the minimalist layerings that the no-input mixer became known for, Ciciliani gives very textured performances, allowing the cross-interference of feedback from the circuits grow into complex patterns.

A particular execution in 2004, of a composition for no-input mixer called “Mask” (Ciciliani, Marco 2007) shows a little more control over pitch and timbre than Nakamura’s pieces, as the artist demonstrates repeatability, and has managed to organise the various noises into well-controlled and performed patterns.

These performances are very small-scale in comparison to my project, both in the number of sounds being controlled, and the degree of that control. Indeed, one major limitation prevails: as audio mixers were never designed to be played as one would a
piano keyboard, the various controls are positioned in such a way that even with the
best eye-to-hand coordination, a No-Input Mixer music virtuoso would at best be able
to produce and control ten sounds: one per finger. If one were to use a large-format
analogue console equipped with an automation system, the automation computer
could be left in charge of most, if not all of the performance’s manipulations, ensuring
consistency in timing and fader positioning.

At the opposite end of the No-Input Mixer music spectrum, Christian Carrière set out
to perform Fratres, originally written by Arvo Pärt for a string quartet, with a 40-
channel Allen&Heath GL2400 (Carrière 2011). This is set in a recording studio’s
control room instead of a live concert venue. Due to the size, weight and wiring
complexity of a large-format automated analogue mixing console, the budget
involved in using one within a live concert touring environment is prohibitive. For his
1988 London outdoor concerts at the Docklands (8th and 9th October), Jean-Michel
Jarre borrowed an SSL to handle the audio for the show and its eight 24-track Studer
tape machines: some playing music backing tracks, some playing cues for the
fireworks and lighting technicians, and some recording the show in its entirety. As
reported by Denis Vanzetto, the head sound engineer for this one-off event, due to
heavy rain and wind, the SSL got wet which caused it to malfunction. Thanks to
SSL’s technicians present on site, the issue was addressed in time for the shows
(Jacob 1990). This is an example which demonstrates how, even with an appropriate
budget, other issues such as weather conditions suggest that these studio consoles
may not be adequate for touring live sound use.

Although Carrière’s performance demonstrates the application of No-Input Mixer
music techniques to a perfectly tonal rendering of ‘conventional’ music, the video also
highlights many of the current limitations of the No-Input Mixer music genre. Each of
the console’s channels has previously been tuned to a specific note. Due to the very
unstable and somewhat unpredictable nature of Larsen effects, the artist would not
dare, even with extensive preliminary testing, tweak the settings during the
performance. Which means the timbres are set once and for all and will not evolve
throughout the piece. And once again, although some of the faders remain in a
position as to produce a background drone, the artist is limited in the number of notes
he can control simultaneously. At best, due to the spacing of the faders, one could
expect Carrière could bring up between three to five notes, along with the continuous
drones. This being a rather slow-paced, hypnotic piece, this is not an issue for his interpretation but this is also an area where traditional non-automated consoles quickly reveal their limitation in regards to more ambitious No-Input Mixer music-making. With even a simple VCA-based automation fitted on the same console, one could envision musical pieces where forty sounds can be produced and controlled at the same time.

2.11 John Cage and intentional misuse

Another relevant precursor to this research is American composer and writer John Cage. If the manipulations required of acetate discs to produce *Etude aux Chemins de Fer* may seem incomprehensible to the novice, one need not be a pianist or sound engineer to predict the sounds that could be produced by one of Cage’s prepared pianos. In 1940, for a performance of Syvilla Fort’s *Bacchanale*, in an effort to expand on the instrument’s original possibilities, and to “place in the hands of a single pianist the equivalent of an entire percussion orchestra”, Cage set out to insert insulation strips under the strings of twelve specific notes (Cage & Charles 1981). In later compositions, the instrument’s preparation and sonic characteristics would be expanded via the use of other materials, such as screws, bolts, bamboo strips, and wool.

Inspired by Henry Cowell (Nicholls 1998), Cage’s exploration of a specific instrument’s possibilities, beyond the original intent of the luthier/manufacturer, represents one of the key aspects of noise-music production. With *Imaginary Landscapes No. 1* (Cage 1995), along with a muted piano and cymbal, turntables are used to play back two discs containing pre-recorded frequencies, which are modulated by having their variable speed adjusted by the performers. Composed in 1951, *Imaginary Landscapes No. 4* (ibid) uses twelve radios, each manipulated by two performers. One dials the radio stations and the other controls amplitude and timbre (Johncage.org n.d.). As described by Cage in *Silence: Lectures and Writings* (1961), “in the radio piece, numbers on a tuning dial are written instead of sounds, whatever happens being acceptable (station, static, silence)”. This deliberate misuse of playback media, be it the playback mechanism of turntables, tape machines, CD players, or the medium itself (vinyl records, tape or compact discs), is referred to as ‘cracked media’ by Caleb Kelly (2009).
The apotheosis of intentional misuse, where the entire concept of an audience’s expectation of a performance is turned upside down, is 1952’s 4’33”, his most famous piece: the musician is instructed not to play, for the entire duration of the composition’s three movements. Cage came up with the idea after having spent some time in an anechoic chamber at Harvard University. This “shattered the belief that silence was obtainable and revealed that the state of ‘nothing’ was a condition filled with everything we filtered out” (Cascone 2000). The purpose of 4’33” is to make the audience aware of the many sounds around them, to let those sounds become the musical experience, on that particular day and time. “Concerts and records standardize our responses, but no two people will ever hear 4’33” the same way” (Gutmann 1999).

2.12 Conlon Nancarrow and prepared player pianos

Since the late ’40s, Nancarrow had focused his compositional intentions on the possibilities of player pianos, due to the limitations of human performers. Annette Margolis, Nancarrow’s second wife, recalls that “when Conlon first tried to have a concert of his music in Mexico, …, the pianist called it off. He said the music [Sonatina] was too complicated, rhythmically, for human hands to play” (Hocker 2002). Although his childhood home had a player piano (Gann 1995), according to a letter sent in 1991 to Dr. Juergen Hocker by Margolis, he was introduced to the idea of replacing the human element with a player piano by one of their friends, Arthur Gregor.

Conlon discussed his musical problems with Arthur, and about the technical difficulties involved with two human hands playing his compositions on a piano. Arthur suggested that since his music was too difficult, technically, for human hands to play, he could punch holes, representing notes from his compositions, on a roll of paper, and then insert the roll into a player piano which would then play the music mechanically (Hocker 2002).

Having purchased a hole-punching device and two player pianos, Nancarrow slowly drifted away from his early jazz influences, to explore complex compositional aspects, now that the human factor was taken out of the equation. This included pieces where notes or canons, as well as their rhythm and acceleration, follow mathematical progressions, in which the intervals, quantity and speed at which the notes are played are well beyond human performance (Gann 1995). My intended use of an audio console’s computer automation system to expand the boundaries of
Larsen-based musical performance is in many ways similar to Nancarrow’s work with player pianos.

At one point in his body of work involving player pianos, Nancarrow entertained the possibility of incorporating elements of Cage’s prepared piano compositions, having met John Cage and Henry Cowell, and having experimented with the concept (Hocker 2002). This led to Study No. 30.

This is the only study written for prepared player piano until the Contraption of 1992-93; Nancarrow abandoned the piece because the screws, washers, and so on would not stay in the strings of an upright piano; an early tape of the piece apparently exists; Like No. 29 only more complicated, it intercuts among five different textures in an accelerative process (Gann 1997).

Indeed, with the advancements in computerised music brought forth by MIDI equipment, Nancarrow would return to prepared piano composition with the help of German musician/sculptor/inventor extraordinaire, Trimpin. *Contraption No. 1 for IPP* (Instant Player Piano) was implemented via a MIDI controlled grand piano, where wedges, dampers and dials can be lowered onto the piano strings, effectively controlling and altering the preparations in real-time. Together with expanding the limits of manual performance, the concept of preparing a musical instrument (mechanical or electronic) so it behaves in ways not originally intended by the manufacturer, are both key elements in my research.

**2.13 Trimpin and prepared player pianos**

Born Gehard Trimpin in 1951, Germany, the Seattle-based artist developed a passion at an early age for cutting apart and recombining elements of musical instruments (Gann 1993). He was classically-trained, “but I didn’t always like classical music. The youth orchestra played very standard and boring semi-classical music. I was more interested in experimental and improvisational music” (Leitman 2011). This inventiveness, coupled with his musical instrument-related health issues, prompted him to shift his focus to mechanised means of producing music.

That’s where I was starting to transition into building physical objects, kinetic objects, because I myself couldn’t touch these instruments anymore with my lips. I was building instruments where I was physically detached, but had control over parameters, like composing for it (ibid.)
He became particularly proficient with MIDI-controlled solenoids, combined with various acoustic instrument parts and other resonators, to form unique sound sculptures which quickly garnered worldwide attention. In 1987, he was asked by Nancarrow to demonstrate a new MIDI-controlled system with which a Yamaha concert grand piano had been equipped (Yamaha.com n.d.). Impressed by how quickly note durations and other elements could be changed, Nancarrow invited Trimpin to Mexico for what was to be a long collaboration between 1987 and 1991. An example of the output of that collaboration, was the aforementioned *Contraption N0. 1 for IPP*. The equipment and installation used to compose and perform *Contraption IPP 71512* was also used for Nancarrow’s *Contraption N0. 1 for IPP*. Resting on top of the piano’s frame, was “a quartet of precision-cut metal bars, from which dangle golf tees, bronze discs, rubber wedges and other trinkets” (Gann 1993). Laid atop the keyboard, is Trimpin’s own version of the Disklavier system, which he called the Force Sensor: “a bar that could be attached to any piano keyboard. There was a solenoid for each key that could play whatever MIDI told it to play” (Leitman 2011).

Trimpin has since built improved versions of the concept, such as the Red Hot which is a grand piano frame held upright so the audience can see the various actions performed on the strings (Mighty Tieton Warehouse n.d.-b). The mechanism is firmly attached to the soundboard’s trusses. “Visitors can also participate by stepping up to the podium and giving the downbeat” (Savery 2014). Also stored at the Mighty Tieton, one can find Trimpin’s construction to realise Conlon Nancarrow’s *Percussion Orchestra*: a large assembly of various drums, cymbals and other percussion instruments, ready to be truck by solenoid-activated mallets or other forms of exciters (Mighty Tieton Warehouse n.d.-a).
2.14 Music scoring and performing

When one composes music where timbre is the prime focus, as opposed to pitch intervals, chords, etc., how does one write instructions for other artists to be able to perform the musical piece as the composer intended? There are many precursors in here, such as Iannis Xenakis, György Ligeti, Edgar Varèse or Karlheinz Stockhausen, who have developed their own method for identifying when and how each sound in their complex pieces must be produced. However, I most likely would not need a score – at least in the traditional paper form – with my approach consisting in having as much of the performance automated as possible. Just as a player piano piece does not need sheet music since the performance is entirely driven by the perforated roll of paper. The console and its automation would be generating and controlling most if not all of the sounds. I have therefore kept this section on scoring succinct, exploring in depth only a few examples which are relevant to my research project’s concept, deliberately omitting descriptive scores (Seeger 1958).

Fig. 4: Stockhausen’s *Spiral* score
Stockhausen scholar Ian Parsons kindly suggested (2018, pers. comm., 17 July) a few iconic examples of scores where the German composer used symbolism. Of these, I have first selected *Spiral*, to illustrate how one might indicate to artists how they should perform, without having any pre-conceived idea of how the final piece should or would sound like (from the notation alone). With his 1968 composition *Spiral* seen in Figure 4, Stockhausen explored what he called “processed compositions”, where the performers are asked to apply 206 transformations grouped into 10 sections, to an initial “Phrase” comprising a soloist (an oboe was used for the first performance), voice and a shortwave radio receiver (Stockhausen 1991). “The notation is in plus-minus-equals signs which instruct the soloist to increase, decrease, or imitate various musical parameters” (I Parsons 2018, pers. comm., 17 July). The parameters are: duration of the event, number of internal subdivisions, dynamic level and pitch. The artist chooses which parameter to change, unless noted by vertically-stacked plus/minus signs, in which case several are altered.

Also of interest is Stockhausen’s score for his *Helikopter Streichquartett* (1992-93). In this piece, which forms part of his *Licht* cycle, each artist from a string quartet boards a separate helicopter, and follows the score aided by an audio click track and pre-recorded voice prompts. As seen in Figure 5, Stockhausen has had to use colour in the notation. “Since the four string players usually tremolo in criss-crossing glissandi, I had to draw their pitch lines and curves on top of one another in four colours, so that the melody trajectories could be followed” (Stockhausen n.d.). The first violin is shown in red, the second violin in blue, the viola in green and the cello in orange. Inside their helicopter, the musicians would wear headphones, in which a click track and voice prompts identifying specific bars or beats, assisted in following the set of instructions laid in front of them. They could also hear the three other musicians’ audio signals. A microphone on the instrument and another one mounted outside of the helicopter to capture the rotor blade sound, was broadcast to the audience via a column of speakers, with an image relayed by a camera onto a video monitor. The four audiovisual columns (one for each musician) are arranged on stage as a quartet would be, with the cello on the left, the viola at half-left, second violin at half-right and first violin on the right. A key element in the dream Stockhausen had in 1991, which led to this composition, is that “the string players played tremoli which blended so well with the timbres and the rhythms of the rotor blades that the helicopters sounded like musical instruments” (ibid.). This is reflected by the glissandi and tremolo
instructions in the notation. Oddly enough, although the players’ glissandi and tremoli are scored down “to the tenth of a second”, the helicopter pilots had a rather wide latitude in terms of how they were to fly.

[once off the ground] the four helicopters circle within a radius of circa 6 km above the performance venue, individually varying their flying altitudes. They should fly so high that the direct sound of the rotor blades is much softer than the sound coming from the loudspeakers, or even better, inaudible (ibid.).

Iannis Xenakis being primarily a mathematician and architect, has a different approach to scoring. In his most famous compositions, the purpose of the score is reversed: the music is not first imagined in the composer’s head, to which the set of performance instructions are then written on paper. Rather, the artist starts off with a visual idea, which once transposed into performance instructions via a set of rules, will generate the various sounds of the piece (Xenakis 1971). One of those early works, where Xenakis’ application of mathematical principles is evident, is *Pithoprakta* (1955-56). Xenakis would go on to conceive a system where the score in the form of various drawings could be interpreted directly by a computer to produce the corresponding sounds of the piece. The UPIC (Unité Polyagogique Informatique CEMAMu) was imagined and designed in the ‘50s, and completed in 1977. It uses a tablet for the performer to draw the various shapes and patterns, linked to a sixty-four-oscillator synthesizer (Thiebaut, Healey & Kinns 2008).
A perfect illustration of that system’s potential is Mycenae Alpha (1978), which is the first work entirely done on the UPIC. Drawings on a landscape spanning from left to right, are then fed to the computer which interprets the ink into sounds, where the vertical axis corresponds to pitch (Figures 6 and 7 show examples of what the original drawn instructions fed to the UPIC would have looked like).

Another example of a graphical score for electronic music, is the score for Wanhfried (1975), composed and performed by Klaus Schulze. Lines and dots are drawn with the vertical axis representing pitch; extra lines indicate various expressions. As seen in Figure 8, ample annotations provide indications as to what those lines mean. That particular piece had eight synthesizers, listed on the left, contributing to the final mix.
Having had a look at several other artists’ output in terms of scoring, it would appear that overall, artists in timbre-driven electronic music are adhering to the same principles as the conventional Western music notation, where the horizontal axis read from left to right represents time and the vertical axis indicates pitch (Gaare 1997).

Closer to the subject of my research, is computer-generated scoring. The example that comes to mind of such a method of producing a score, is that of a person playing notes on a MIDI keyboard, which are recorded into a DAW, and displayed in quasi-real time as either a piano roll-style layout or on staves. A popular example of such a software is Avid’s Sibelius 7 (Avid 2011). One of the interesting developments in the field of computer-driven scoring in recent years, is that of Live Coding, where the score is changed in real time: the performer instructs the computer what it is to do next, with a mirror of the computer screen usually video-projected for the audience to see. The code represents the majority of the score (it may not indicate what the

* Early experimentation of real-time algorithmic composition in performance stretches back as far as the ‘70s, led by artists such as Iannis Xenakis, Peter Zinoviev or Joel Chadabe but the advent of MIDI (and furthermore, OSC), has popularised the concept (Roads 1996; Xenakis 1971).
original input audio signals are), but since it is constantly changing, unless a complete log is kept of the various command lines as they were input before each alteration, there is no lasting trace of the instructions necessary to perform the same piece again. The log would additionally need to contain precise timing references, to indicate exactly when a command line should be altered if the performance was to be repeated. Hence, there is often no score per say; the score only exists in the moment. Furthermore, unless audience members are particularly well-versed in the programming language and various platforms used for the performance, they would be hard pressed to foretell the resulting audio signals from the instructions they see projected on the screen, as per Figure 9. To that extent, efforts have been made to present the coding (or its results) in various graphical forms, to complement or replace the alphanumerical commands, as seen in Figure 10 (Magnusson 2011; McLean et al. 2010).

Fig. 9: Example of raw Live Coding video projection data

Fig. 10: Example of Live Coding graphical interface (video still)
When I conceptualised my composition, at first, I thought I would not need additional instructions for the performance. The files produced by the console’s automation system would constitute the score, just as a player piano roll provides all the information the player piano needs to perform a piece exactly as the composer envisioned it. However, It became apparent in the compositional process, that extra documents were required and that I too would have to “write” a score. This focused literature review on scoring has provided guidance in that matter. Finally, scores not only allow performers to execute a piece according to the composer’s intent, but also allow analysis and study of the piece by scholars. Further discussion of scoring and performance aspects pertaining to the composition and execution of the project’s musical piece, can be found in Chapter Five.

2.15 Repurposing audio equipment

Occasionally, throughout music production history, specific pieces of audio equipment having missed their intended market base and fallen into obsolescence have benefited from a sudden regain of interest via intentional misuse, effectively being offered a second life. This section reviews examples that are relevant to the misuse of electronic audio production equipment.

The turntable, which followed the slow decline of vinyl record sales when CD technology was made public in 1982 (Browne 1991), has since then seen a boost in its popularity. This is due in great part to turntablism, introduced in the early days of the hip-hop genre. The regain of interest has reached a point where manufacturers such as Technics are pressured to re-open their production lines of the acclaimed SL-1200 (Daniel, M 2014). Pierre Schaeffer’s experimentations with acetate discs, or John Cage’s Imaginary Landscapes No. 1 (Cage 1995) are very early examples of turntable repurposing.

The Roland TB-303 bass line synthesizer was introduced in 1982. “It was originally made to accompany a drum machine, the TR-606 specifically, and provide bass-line accompaniment to guitarists, keyboard players, etc.” (VintageSynth.com n.d.). Having failed quite significantly in its originally intended market, it was easily purchased by aspiring musicians and DJs on a budget, who brought it to the foreground as the lead instrument of several electronic music styles, most notably Acid House and Techno (Taylor 2001).
Within the repurposing template, another common musical art form that often involves Larsen effect-based sound generation is “circuit-bending”, where electronic devices are reconfigured by adventurous DIY (Do It Yourself) enthusiasts, to make them produce sounds that the original manufacturer did not intend them to make. Larsen effects are only one of many ingredients at play here instead of being the centrepiece, as the circuit board tracks are often shorted in the circuit-bending process. This is even at times done by exposing the circuit board to allow the performer’s fingers to do the shorting (BentElectronics 2008). The Texas Instruments Speak &Spell, introduced in 1978 (Shu 2012), is a very popular device for circuit benders, as the speech synthesis chip is capable of producing a wider variety of sounds other than the original letters of the alphabet and pre-programmed words, by looping individual phonemes, gliding between them, or distorting them. Several websites offer modified units for sale with various combinations of pre-fitted switches and knobs, allowing a more predictable sonic behaviour (Pill Shovel Productions n.d.).

Unfortunately for budding No-Input Mixer composers, a large-format analogue console such as the one used by Christian Carrière is no longer what the music production industry wants in their studios (Pras, Guastavino & Lavoie 2013). They are being sent by studio owners to online second-hand audio stores such as Funky Junk and Vintage King, hoping to find a buyer even at a fraction of the original cost. When one considers that these consoles could easily reach the $1,000,000 mark, and can now be found for under $50,000, this could be a sound investment for someone looking into using such a console for no-input mixing. For example, a Neve VR60 (i.e., fitted with 60 I/O modules) based in South Korea was on offer for $39,000 on eBay. As noted by the seller, the console cost US$ 700,000 when it was purchased in the ‘90s (Ebay 2014). Can enough interest be generated via my research project, into the many possibilities of large-format analogue, automation-controlled No-Input Mixer music, to encourage these musicians to make such a purchase?

### 2.16 Gaps in current knowledge

In this chapter, I have provided a brief overview of noise-based music landmarks and their key protagonists. I proceeded to highlight various uses of Larsen effects in the field, to then focus on purely-electronic implementations of the concept. Also relevant
to the research were the themes of scoring and repurposing. Throughout this preliminary investigation, control over pitch and timbre and to a lesser degree, volume, whenever Larsen effects are used by musicians, was found to be quite minimal. I believe this is mostly due to the costs involved in accessing or owning a large-format analogue mixing console, restricting the artists to much smaller analogue consoles, which are not computer-controlled. Also, there is the issue of mobility: most of these musicians are live performers, which rules out a traditional large-format automated analogue console, weighing in at nearly one tonne (including two metre-tall power supply and computer rack). With the appropriate approach, the originally unpredictable nature of Larsen effects could be controlled, leading to more applications within the music composition and performance fields than an occasional spot effect within a broader traditional orchestration. A survey of the existing literature suggests that the full potential for Larsen effect-induced music production was yet to be explored.

Figure 11 summarises where my research sits within the broader scope of electronic music – it is read from left to right, top to bottom. As my study focuses on the use of Larsen effects to produce music, I am investigating its potential within the noise-based music category, and in so doing, I am hoping to break the boundaries of that niche to produce pitch-based music if enough control is achievable with my method. Larsen effects are generated in real time, which is itself distinguishable from any noise-based music composed and performed using pre-recorded material or “found sound”. Within the category of “generated sounds” in noise-based music, one would find all of the music produced using microphones such as in Reich’s *Pendulum Music* (1966) or using synthesizers and samplers. No-Input Mixer music itself, is just a small fraction of that particular segment of noise-based music.
Contrary to Reich’s approach, I aimed to develop consistency and precision in the real-time generated Larsen effects, by eliminating as many random aspects as possible in the performance, letting the console’s automation computer handle the changes whenever possible. Also, performances reviewed are very small-scale (with the exception of Carrière’s *Fratres*), both in the number of sounds being controlled, and the degree of that control. By using a large-format automated analogue console, the automation computer could be left in charge of most, if not all of the performance’s manipulations, ensuring consistency in timing and fader positioning. This is one of the areas where existing literature seems to indicate a major gap, which I have explored with this research.

* An early example of music achieved entirely via real-time synthesized noises, in 1956, is the soundtrack for *Forbidden Planet*. The song entitled *Battle with Invisible Monster* (Barron & Barron 1956) combines long pitch-drifting Theremin-like notes, with various bursts of noises, some evocating a heart-beat and/or heavy footsteps, others, the havoc about to be unleashed on the protagonists.
Chapter Three: Research design and methodology

3.1 Research question

Having assessed the gaps in current knowledge regarding No-Input Mixer music, the available literature directed the research project’s main question: what potentials exist for producing music using a large-format automated analogue audio console as a sound-generating device, via controlled Larsen effects triggered with internal feedback loops? The outcome of the research may have implications for the artists, music producers and studio owners around the world, who are using or have invested in expensive large-format automated analogue consoles.

To address the main research question, the following sub-questions were assessed:

- How will the hardware and software limit/restrict the creative outcome and its aesthetics?

- Which signal paths produce which kind of tone?

- How much control of that tone can be obtained using the console’s circuitry and on-board processing (EQ and Dynamics)?

- Will the console’s automation system time base be precise-enough to generate rapid, steady rhythmical patterns, akin to using a sequencer or drum machine to produce 16th or 32nd notes?

- Will the console’s automation system allow precise-enough control of pitch, or will I only be able to compose a piece using noise and textures, much akin to some of Merzbow’s work in the past decade (Doornen 2006)?

- Will the console’s Total Recall\(^\ast\) capabilities be precise enough to provide repeatability, or will a considerable amount of manual tweaking be necessary before each performance of the piece?

\(^\ast\) Total Recall can store all the positions of the console's knobs and switches, so that these settings can be retrieved later on, for another session.
- Will I have to develop my own symbolism to compose and sketch ideas on sheet music, or can I rely on existing notation systems, such as the ones used by Pierre Schaeffer as early as 1948 for his Musique Concrète (Hodgkinson 1986), or the other examples noted in the literature review?

My key research question and its associated sub-questions, indicated that the method and practice leading to the creative output is the focus of the research, not the creative output itself. This research and its associated composition, is akin to Bach’s two books, the *Well-Tempered Clavier* (1722 and ca. 1740), which “were designed to demonstrate the possibilities of playing in all keys on an instrument tuned in near-equal temperament” (Burkholder JP, Grout JG & Palisca 2006). I was investigating and developing a new tool for producing music, not as an “outsider, visitor or participant in scientific practice”, but as “an undertaking in which artistic practices contribute as research to what we know and understand” (Borgdorff 2012).

When investigating how this initial premise fits within the creative practice research umbrella, the acceptance of creative practice research within the broader academic community being quite recent (Frayling 1993), it seems that depending on the scholar, the terminology to be used has yet to be fully agreed upon.

The categories of research described by Hilary Collins (2010, p. 15), would indicate that my research fits the model of her second category, which is “…where the research has a practical requirement and involves collaboration between researchers. Such research would result in the generation of knowledge, which would then be applied to a practical situation”; albeit one would here substitute “researchers” with “fellow mixing engineers/producers”. This category is summarised in her work as: Question → Answer → Use.
As can be seen in Figure 12, Smith and Dean use the terms “practice-led research” and “research-led practice” as two components of the research process within the creative arts.

The term practice-led research and its affiliates (practice-based research, practice as research) are employed to make two arguments about practice which are often overlapping and interlinked: firstly, as just indicated, that creative work in itself is a form of research and generates detectable research outputs; secondly, to suggest that creative practice – the training and specialised knowledge that creative practitioners have and the processes they engage in when they are making art – can lead to specialised research insights which can then be generalised and written up as research (Smith & Dean 2009, p. 5).

Linda Candy (2006) suggests a distinction between Practice-based and Practice-led research. As per my research question, I was exploring tools that had yet to be documented in order to produce a creative work, but the validity of that work can only be assessed with the adjunction of the creative output. According to Candy’s own definitions, my research fit within both of her paradigms.

Furthermore, as it would be irrational to quantify creative artwork, or the adequacy of a tool to generate it, my analysis and reflection on practice was qualitative as opposed to quantitative (Patton 2002). My research therefore followed the qualitative,
practice-based paradigm (Barrett & Bolt 2007). Within that framework, drawing inspiration from Smith & Dean's diagram in Figure 13, I developed my own research workflow diagram, below in Figure 13:

![Research Workflow Diagram](image)

The following paragraph outlines the various research components that make up this research workflow diagram – each component is detailed in full under Chapter Four. The Research Question was investigated via the means of a Composition & Production (a piece of music), which led to a Performance of said composition and production. To develop the elements that make up the composition and production, a Signal Flow Exploration was led, through which various timbres were catalogued.

Critical Reflection took place at every stage of the research (Fook 2010). Early stages of critical reflection involve a Diary, which includes notes and findings from every studio session, whether the session was dedicated to timbre exploration or composition and production of the musical piece. During the early stage of signal flow exploration, this critical reflection focused on the prospective usability within the musical piece and nomenclature, and in turn, informed further exploration and variations into available timbres. Progress and results of the signal flow exploration were published on social media (research-dedicated Facebook page and YouTube Channel) in the form of pedagogical videos, as a means to collect additional feedback from peers and the general public; this was also done to generate interest in the research, with an expectation that this would increase the amount of feedback collected (Hookway 2008; Olive 2013). Another reason to use an audio-visual medium to document the research progress was that the posted videos would be
invaluable when comparing the differences between various timbres. As much as a spectrogram and written description can help in analysing and cataloguing sounds, being able to hear those sounds and see in which context they were obtained (including through which gestures), provides an important complementary perspective. Additionally, this provided statistics via Google Analytics. During the composition and production phase, videos of the sessions were again produced with a pedagogical perspective and posted on the research project’s social media outlets. It was envisioned that critical reflection would influence the compositional process: both my personal findings and comments from social media could influence the direction in which the piece was heading. However, through the course of the research, this avenue of feedback proved to be insufficient and therefore useless. This is discussed in more detail in 6.5.

Once the composition was produced and rehearsed, a public performance was organised to demonstrate the capabilities of the music production tool and method. In-situ interviews and a survey, collected feedback from audience participants. Video footage was also shot to produce a Performance Video which was then posted on the project’s social media outlets; this, again, with the ambition to collect as much qualitative feedback as possible. An online survey accompanied the video post of this major element of the research output. Last but not least, the diary findings were organised into chapters to form Chapter Four and Five of this exegesis. These sections are each dedicated to: experimentation with signal flow and timbre exploration, cataloguing of said timbres using the console’s automation system for storage and recall of settings, using the console’s automation system as a sequencer, composing the piece, scoring and aspects of live performance.

3.2 Research boundaries

In Focus:
- The ability to accurately and repeatedly perform a composition using controlled Larsen effects.
- Signal flow and processing techniques utilised on a large-format console.
- Sequencing of notes and rhythmic patterns using the console’s automation system.
- Installation and public performance of the piece.
- Production of a documentary video of the performance.
- Documentation of research via blogging.

Out of Focus:
- The artistic quality of the final musical piece in terms of composition and arrangement.
- Larsen effect-generation involving outboard processing via the console’s insert points. There were enough variations to explore already with just the on-board signal processing capabilities.
- Larsen effect-generating techniques on consoles other than the one(s) selected for the research.
- How outboard time-domain effects (reverberation and delay/echo units) participate in the final production and performance.
- The technical quality of the 1080P HD video documenting the performance.
- The technical quality of the videos posted on YouTube and Facebook.

3.3 Data collection and analysis

The data collection was broken down into several stages, which address the research project’s sub-questions. First, the timbres that could be generated by the console had to be explored and catalogued. This gave direction as to whether the hardware and software would limit/restrict the creative outcome and its aesthetics. This also addressed which signal paths would produce which tone and assess the degree of control offered by the console's circuitry and on-board processing (EQ and Dynamics).

A systematic approach was needed to guarantee that the timbres found during that exploration stage, were the result of the actual settings dialled in on the console’s module, and not the result of a random fault in a particular module’s electronics. Whenever a potential new timbre was found and the settings causing it had been noted, several modules were picked at random and the settings were replicated. Only if the resulting timbre was easily replicated in a significant proportion of randomly-selected modules, would the new timbre be officially declared as having been identified, and given a name.

* Composition and production are usually two separate stages of the music-making process. Here, more than in most other situation, both processes are so intertwined that production becomes the main focus.
When it came to nomenclature, I already knew that I wanted to keep things simple, by deliberately using a system where the timbre’s name would evoke the same reaction or association in the collective subconscious. If I name a timbre “Sawtooth”, it is fair to assume that a majority of knowledgeable readers will have an immediate idea of what the timbre sounds like. We are definitely in Pierre Schaeffer’s “reduced listening” scenario, in that it is useless in that case to refer to the actual cause of the produced timbres, in an attempt to describe each sound (Chion 1994, pp. 25-34). At times, it felt more appropriate to revert to a “causal listening” name, examples of which are the “Flatula”, “Gecko” or “Flipper” timbres.

To better document the various timbres I would encounter along the timbre exploration phase, the audio and text descriptions of each timbre are complemented with a spectrogram, in order to assist the reader in an analysis of the sounds’ frequency content (Couprie 2004; Smalley 1997). After having tested a few spectral analysis software, such as Sonic Visualiser 2 (Sonic Visualiser 2017), the spectrum analyser embedded in Apple’s Soundtrack Pro 3 – which was part of the Final Cut Suite, with FCP 7 (Apple n.d.) – or Metric Halo’s SpectraFoo 1.2 (Metric Halo 2016), I settled on using iZotope RX 1 because I found the colour choices of various shades of orange over a black background to be most legible in comparison (Izotope 2012). Since Larsen effects are generated via near-infinite gain inside the signal path, this leads to its nature being primarily based on harmonic distortion, and as such, visualising frequencies on a spectrogram would benefit from the frequency scale being able to best display harmonic relationships on the vertical axis. This therefore ruled out iZotope RX 1’s Log or Extended Log scales, as these put far too much emphasis on the lower frequency spectrum, whereas most Larsen effects encountered so far cover the entire audible spectrum (and beyond). The Linear scale, has the opposite inadequacy, in that it places too much emphasis on the upper frequencies (the last octave occupying half the screen) and as such, does not as clearly identify harmonic relationships as what one would hope. This left two possible scales with this particular audio processing and analysis software: Mel and Bark. Although the Bark scale seemed more appropriate in correlating the visualisation of the audio signals and its frequencies, as it factors perceptual loudness in its measurements and visual presentation of each band (Zwicker 1961), I found the Mel scale (Stevens, Volkmann & Newman 1937) was best at distributing the bands of the broad frequency range of Larsen effects, with less emphasis on the low and low-mid
frequencies, as noticed on the Bark scale, with particular emphasis on perceived pitch. Therefore, all spectrograms in this research – unless otherwise noted – were produced using iZotope RX 1’s Mel scale for its vertical, frequencies axis. See Figures 14 to 18 below, showing the same Larsen effect sample, visualised with iZotope RX 1’s five different frequency scales: Linear, Mel, Bark, Log and Extended Log.

Finally, each spectrogram has been imported into a Photoshop template for consistency in the vertical scale and size of the spectrogram diagrams. Furthermore, iZotope RX 1’s light grey scale lettering (frequencies and time) not being easily legible once reduced to an A4 printout, the scale information has been made white, and have had their font size increased by three points. A test printout of this spectrogram template was made to verify that this was easily legible. The horizontal scale has been adjusted for each spectrogram when necessary, to present a pertinent section of audio, for timbre and time-based visual analysis. The default unit is seconds and milliseconds. For example, a particular section facing 5.35 on the horizontal axis, means that moment within the sound occurs 5 seconds and 350 milliseconds from the start of the analysed audio sample.

Fig. 14: Spectrogram of typical Larsen effect visualised with iZotope RX 1’s Linear frequency scale
Fig. 15: Spectrogram of typical Larsen effect visualised with iZotope RX 1’s Mel frequency scale

Fig. 16: Spectrogram of typical Larsen effect visualised with iZotope RX 1’s Bark frequency scale
Fig. 17: Spectrogram of typical Larsen effect visualised with iZotope RX 1’s Log frequency scale

Fig. 18: Spectrogram of typical Larsen effect visualised with iZotope RX 1’s Extended Log frequency scale
Production process data was collected primarily using a written logbook, in diarized form. This has then been compiled for inclusion in this exegesis. The diary contains:
- spectrograms of particular timbre settings
- diagrams showing connections used to help reconfigure sessions
  - patch bay settings
  - outboard processor settings (if applicable)
- commentary collected from the project’s social media sites
- analysis of said commentary and reflection on appropriate action
- analysis of industry professional interviews following the piece’s public performance

During the timbre investigation and other explorations, production of the musical piece, and preparation of the performance, videos of the research progress were produced and posted on social media. According to Rebecca Olive (2013), blogging seems to be a very engaging and responsive way to position oneself within a particular sub-culture, and to refine one’s language to better correspond to the target audience. This would provide feedback and possible avenues for improvement, via online comments in response to the videos and short articles.

Two online resources have been created for that purpose:
- A Facebook page: Skweeel
  
  https://www.facebook.com/pages/Skweeel/547098688723098
- A Youtube channel
  
  https://www.youtube.com/channel/UCEN9Hb-qCk6A9_Jnr92qAdg

The name of the Facebook page (its availability) has had the immediate consequence of providing a title for my research: Skweeel. The “triple e” phonetically symbolises the sound of most Larsen effects in peoples’ collective minds. The public performances provided another avenue for the qualitative data collection, in the form of a questionnaire for all participants, and interviews with industry professionals – See Appendix C, D and E for related documents. The performance was edited and refined into a 1080P HD video, which represents the research project’s main output. I found it was the best way to make both the audio soundtrack of the composition, and the visual aspects of its performance, accessible to the largest possible audience.
Video interviews were conducted with industry guests, prior to the public performance of the piece, and shortly after the performance, to collect impressions regarding the overall concept of the project as well as artistic and technical opinions regarding the production. Analysis of these interviews and results were part of the critical reflection. Said critical reflection also took place during the production process, once the performances had taken place and once the HD 1080P video of the performance was produced. As previously stated, peer feedback was deemed insufficient overall, for a relevant discussion in this document. One can find the survey results in Appendix L, and further information at the end of each diary entry, in Appendix A.

This written exegesis documents the findings following the “commentary model” described by Milech and Schilo (2004). I found it also had to incorporate an analysis following Milech and Schilo’s “research-question model”, as to form the best possible complement to the 1080P HD video of the performance, its viewing experience, and the blog and posted video content.

### 3.4 Benchmark for composition

To investigate and validate the research question and its associated sub-questions, a possible approach would have been to produce a series of short studies: several demonstration videos, each focusing on one particular sub-question. I preferred to showcase all of the research project’s potential into a single musical composition, to better demonstrate how these techniques could be applied to music production. This would be more appealing to the viewers on the project’s social media outlets, generating more interest in the concept and thus, hopefully more feedback. The musicianship of the composition itself – its artistic qualities and appeal – is confined to illustrating the functionality of the established creative system.

When George Lucas’ company THX commissioned Dr James ‘Andy’ Moorer to produce the sound that would accompany the famous logo’s first on-screen appearance (Whitwell 2005), he was not just composing a piece that had to be pleasant, he had to make a statement on the new technology. As noted by John Dahl from THX (2006): “The first THX trailer had to deliver a sound and visual experience that left an impression on the audience, getting them primed for the feature...
presentation and making them realize that they weren't in just any typical movie theatre”.

Sound designer and re-recording mixer Gary Rydstrom adds: “The goal of producing a THX trailer is to create something that lets the audience experience the capacity of the sound system without being overwhelming” (ibid). Hence, the trailer goes from whisper quiet to impressively loud, demonstrating the improved acoustics of the Theater Alignment Program and the amplifier/speaker’s ability to handle loud SPL. It also incorporates room-shaking low frequencies and shimmering, distortion-free upper harmonics, to showcase the electroacoustic chain’s capabilities. Last but not least, the initial sounds gently swing around the room to highlight the accuracy of the surround sound channel allocation.

Similarly, the piece composed for the purpose of this study should address the shortcomings identified in the literature review, in terms of sound synthesis and performance. It should demonstrate the timbre generation possibilities of the proposed method, as well as an advanced degree of control over those timbres. It should also showcase the ability to produce and control more sounds than could be achieved by a single-person’s manual performance. Therefore, the final piece should (in no particular order):

1. Include several different types of timbres with a variety of nuances
2. Demonstrate a high accuracy over pitch control and other parameters such as modulation via various triggers
3. Incorporate sections where a large number of sounds are generated and controlled simultaneously – far more than ten, which would be impossible to achieve by hand by a single artist
4. Incorporate sections where sounds are generated in rapid sequence, which would be impossible to achieve by hand
5. Incorporate sections where the rhythmic precision of particular patterns would be impossible to achieve by hand
6. Incorporate an element of surprise (if tonal control is achievable) by composing a piece which starts off as noise, but progressively tends towards melodic and harmonious content.
3.5 Ethical considerations

The ethical considerations revolved around academic scrutiny, honesty (referencing to the best of my knowledge according to the Harvard standard), securing necessary permissions for re-use of images, graphics, etc. An Ethics Application was lodged with Southern Cross University on the 4th of December 2014, and approved on the 12th of December 2014: ECN-14-291. The ethics approval extends until the 21st of December 2018.

I have been in touch with SSL – the manufacturer of the consoles I used – on a regular basis from 2006 to 2013, regarding the XL 9000 K’s maintenance, so have developed a good working relationship with them. However, I chose not to involve the company in the research, to avoid bias.

Industry professionals and general audience members were asked for permission to publish their statements in my research. In a similar attention to privacy, all people attending the performances were asked for permission to film them, and use the footage in the edit for the final HD 1080P video. They filled out a release form before the show. Industry professionals and other possible interviewees were also asked for permission to use their footage (and comments herein) in the video of the performance. See Appendix E.

Having contacted two former colleagues, which are amongst Australia’s leading SSL and Neve console maintenance engineers, they have both guaranteed - one in writing, one in a casual phone conversation - that my Larsen effects would not damage the console considered for the project: an SSL XL 9000 K (S Crane 2013, pers. comm., 29 November; B McBean 2013, pers. comm., September). This was a key factor in determining the feasibility of this research altogether, and SAE Institute Byron Bay’s willingness to provide access to their studio. I have thanked SAE Institute Byron Bay for their contribution to my project in my exegesis, on the blogging website and in the credits and opening commentaries of each video blog.

The purpose of my blogging being to inspire budding No-Input Mixer artists, ample warning was posted on key sections of the web site, and within every video, that if they are to experiment with no-input mixing, there could be potential damage to their
console, potential damage to their speaker system, and most importantly, potential
damage to their hearing.

3.6 Potential issues

To meet the deadlines I had set for each step of my project, I needed access to the
console several times per month on average. I was the Head of Audio at SAE
Institute Byron Bay from 2008 to 2013, and having kept a good rapport with the local
administrative and pedagogical team, I was able to secure unsupervised access to
the SSL XL 9000 K studio in the evenings after hours, and during the week-ends, as
long as other staff members were not using it. This was an agreement with the
I had planned some headroom within each research phase deadline, should I have
needed extra sessions to accomplish my goals.

What I did have to always consider, was that the console could develop a fault so
expensive to fix that SAE Institute Byron Bay decides to sell it. This risk was
evaluated with SAE’s Head of Audio, Dirk Terrill, and myself, before undertaking this
particular research project. Were this to happen, the initial contingency plan was to
relocate the research and performance to SCU Lismore’s Studio A in Block D. It
featured an Amek Big by Langley (Amek 1993), which is a medium-format analogue
console with VCA-based automation, EQ on every channel and virtual dynamics
sections on every channel via its automation. Aside from having to diminish the
overall scale of the performance, which is not vital to my key research question and
sub-questions, I would not have been betraying the original intent of my study in any
other way.

As explained below, though, another solution became available and much more
suitable, as my initial apprehension did happen: the console I intended to use
became inoperable approximately one third into the data collection phase of the
research.

3.7 Application of research method

Having refined the concept, benchmark and limits of the research, it was then
possible to implement the research method: applying it to a suitable audio production
console: the SSL XL 9000 K at SAE Institute Byron Bay. This is the largest and most complex in-line analogue console, in terms of signal flow capabilities and audio quality, ever designed by Solid State Logic (Peterson 2002). With more than four decades of console manufacturing history, along with Neve and to a slightly lesser degree Harrison and API, SSL are regarded by the music production industry as the quintessence of analogue audio mixing consoles (Gearslutz.com 2008; Rotondi 2011). According to Anthony Garvin, manager at Studios 301 in Australia (2013):

I usually refer to our Neve 88R and SSL 9000k consoles as the “Ferrari and Lamborghini of mixing consoles”.... They are both the best cars that money can buy, but it’s up to the driver as to which one they prefer to drive.

3.7.1 Using SSL’s XL 9000 K

The frame in SAE Institute Byron Bay's main studio is fitted with fifty-six I/O modules with the following features as standard, all of which are key to my project and research questions:

- A four-band parametric equaliser, which allows me to shape the frequency spectrum and manipulate the generated Larsen effects.
- A set of high-cut and low-cut filters which offer additional means of tone control.
- A dynamics section with compressor/limiter and expander/gate, which can provide tone control, but also melodic and rhythmic pattern-generating control.
- A complete patch bay providing access to multiple points within the signal path.
- The Total Recall system, which can store all the positions of the console's knobs and switches, so that these settings can be retrieved later on, for another session.
- A system which allows the Channels to be named, albeit only on the computer screen. This might however prove useful in assisting with aspects of the live performance.
- The SSL 9000 K automation system, which will act as the sequencer for the performance. This allows automation of the Large Fader (motorised) and Small Fader (VCA only) with their associated Cut switches. Additionally, the following switches can also be toggled: EQ On/Off, Insert In or not, and all Aux Sends On/Off (Cue St and FX 1-6).
As each I/O module has two audio paths, potentially, one hundred and twelve audio signals can be produced simultaneously. Their processing power, along with a very flexible signal path, had the potential to provide the equivalent of a well-developed 112-oscillator subtractive synthesiser, such as the Buchla, Moog or PPG modular synthesizers of the ‘60s (Buchla.com n.d.; Moogmusic.com n.d.; ppg.synth.net n.d.). The Larsen effects play the role that a traditional modular synthesiser’s VCO would play, generating various waveforms. These can then be shaped by the EQ and filters, which are akin to the VCF modules in subtractive synthesis. The faders and to some extent the dynamics section, act as VCAs. The on-board Total Recall system would facilitate performance reset and preparation. The on-board automation can move the large faders, mutes, and toggle a few other switches such as the Insert Point activation, Aux Sends on/off, and EQ on/off. These two capabilities combined provide a rather flexible sequencer, similar to what would be achievable on a traditional CV/Gate sequencer. External effects processors, such as reverb and delay units, were used to enhance the final product, on the grounds that any other music production using external sound sources passing through a mixing console would also use those effects.

3.7.2 Using SSL’s Duality SE

After having completed most of the data collection for timbre investigation and just as I was about to start the in-depth study of the Total Recall and Automation system, the computer on the SSL XL 9000 K console I originally set out to use for the project, suffered from an irreparable failure – a crucial component, as it is responsible for storing and retrieving settings, as well as automation. With no maintenance scheduled by SAE Institute Bayforeseen in a reasonable time frame, I was about to put my contingency plan in action, of using Southern Cross University’s Amek Big console in Studio A. However, not long before the XL 9000 K computer irremediably broke down, a forty-eight-channel SSL Duality SE was installed at SAE Institute Byron Bay. It was not available in 2014 when the research commenced. This console qualified in every aspect as a replacement for the research. The SSL Duality SE in SAE Institute Byron Bay’s Studio One is fitted with forty-eight channels with the following features as standard:

- A four-band parametric equaliser, identical to that of the XL 9000 K.
- A set of high-cut and low-cut, identical to that of the XL 9000 K.
- A dynamics section with compressor/limiter and expander/gate, identical to that of the XL 9000 K.
- A Total Recall system, which works in a similar way to that of the XL 9000 K, via multiple built-in video screens depicting channel settings.
- Channels can be named and displayed on an electronic scribble strip (six characters max.)
- The same parameters that can be automated on the XL 9000 K using its on-board computer system, can here be automated via a plug-in running in your DAW. These are automation of the Large Fader (motorised) and its Cut switch. Additionally, the following switches can also be toggled: EQ On/Off, Insert In or not, and all Aux Sends On/Off (Cue St and FX 1-4).

Although most of the timbre investigation had to be carried out again, this change was a blessing in disguise, as it offered the following: pitch control was not achievable with the original console, i.e. I could not get the automation system to precisely change from a particular note, to another, even just a semitone apart. Whereas with this replacement console, not only could notes be replicated precisely, but a significant range of semitones could be accurately and repeatedly obtained. The timbres themselves seemed overall similar, but with a perceptible lack of high-frequency content, i.e. upper harmonics being generated. This slight drawback was more than compensated by the aforementioned ability to control pitch and therefore construct melody and chords, but even more so by the fact that the replacement console’s automation system relies on an external DAW, and not on an antiquated proprietary computer. Programming music with the hereby proposed method became far easier and much more powerful, due to the user-friendly GUI of today’s DAWs. A consequence of this, and an additional advantage, was that sequences and other material could be pre-programmed at home, to be later ported onto a similar DAW controlling the replacement console. Most significantly, this has broadened the scope of the research, as I have decided to present the data and analysis from the old console (XL 9000 K) alongside that of the new one (Duality SE) as I saw this as an opportunity to discuss both technologies and associated workflows of Automated No-Input Mixer Music production – using both console-based and DAW-based automation systems. This would allow the reader to apply the techniques to the many consoles around the world that still rely on their own built-in computer for automation.
Each sub-question could now have specific processes put in place to address them, starting with “Which signal paths produce which kind of tone?” and “How much control (variations) of that tone can be obtained using the console’s circuitry and onboard processing (EQ and Dynamics)?” Even though the question of timbre was initially thought to be investigated into these two stages, it made more sense to encompass both in the same process. There are many different ways in which a Larsen effect can be triggered within either the In-Line module of an XL 9000 K or the Channel of a Duality SE, each of which could produce a different timbre:

- Insert Send back into the Line Input
- Insert Send back into the Mic Input
- Insert Send back into the Group Input
- Direct Out back into either Line, Mic or Group Inputs
- Aux Send back into either Line, Mic or Group Inputs
  - Pre Fader
  - Post Fader
- Direct Out back into the Insert Return
- Routing Matrix source set to ‘Channel’ with Channel Source set to Group and Channel routed to same number in Routing Matrix (XL 9000 K only)
- Routing Matrix source set to ‘Monitor’ with Monitor Source set to ‘Group’ and Monitor routed to same number in Routing Matrix (XL 9000 K only)
- Plus many more, such as on the XL 9000 K, having the Channel feed the Monitor which then feeds back into the Channel, or the contrary, with Monitor to Channel to Monitor

Within these basic configurations, systematic exploration of various interactions between the audio path’s settings was necessary, as any control applying even the slightest gain change will have a repercussion on the Larsen effect’s pitch and timbre.

**3.7.3 Addressing sub question – tone generation and degree of control**

On the SSL XL 9000 K, the possible combinations of EQ, Filters and Dynamics Processing sections are numerous due to SSL’s complex signal processing routing

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* For clarity, in the rest of this document, physical controls on the console (switches and pots), paths and access points are denoted using capital letters: e.g. Mic Input Gain.
on their large-format consoles. Figure 19 shows a simplified signal flow diagram featuring the XL 9000 K I/O module’s Channel and Monitor paths, with the various routing possibilities of the Insert Point, EQ, Filter and Dynamics sections. SSL has dedicated an entire chapter of the console’s manual to cover signal processor routing (see Appendix F – SSL XL 9000 K Manuals / Console / Section 04 Signal Processor Routing). Additionally, as with any console equipped with an Insert Point, said Insert Point can be used as an input into the Channel path if one wants to shorten the signal flow to the output fader.

To list the numerous available configurations, several tables were drafted and compiled in Appendix G. The following abbreviations were used to designate the relevant electronic circuitry blocks: Mic, Line, Group, Pre LF (for Channel Output Pre fader) or Pst LF (for Channel Output Post fader), EQ, Filter, Dyn (for Dynamics section), Ins (for Insert Point) or Ins Pre (for Insert Point when set to Pre), Aux and Aux Pre (for Auxiliary sends Post or Pre fader) and Fader. Each row in the tables is
read from left to right, indicating the order in which the various signal flow elements are found within the audio path of that particular configuration. Furthermore, the feedback loop’s entry point is indicated in yellow, and the exit point, in green. With such a high level of signal flow flexibility, some configurations would most likely be redundant. An example of a redundant configuration would be selecting the Insert Point as Pre or not (i.e. Post), whilst the Dynamics section is the sole signal processor inserted in the path, using Ch In: in both cases, the resulting order is Dyn then Ins. During “normal” use of the console, when producing music, the two different sets of copper tracks and electronic components on the circuit board that the signal would go through in the previous example, between the Insert Point set to Pre or Post, should have no noticeable (and even measurable) difference on the audio signal. However, when using the console for No-Input Mixer music production, with the circuitry driven to overload by the feedback loops’ Larsen effects, it could be that the resulting sound would differ in a noticeable way. This was to be tested during the timbre exploration stage of the research’s data collection. Therefore, instead of eliminating redundant configurations, for clarity, they have been kept in the tables, but struck through. Last but not least, the Auxiliary Sends can also be used as exit points for the feedback loop, either Pre or Post Fader. In order to reduce the clutter in the tables and maximise their legibility, Aux Pre or Aux (as in Post) will appear in selected tables, where relevant.

I first investigated the Input to the Channel selected as Line and patched to be the feedback loop’s entry point, and the Insert Point selected as the feedback loop’s exit point; the patch bay points used are Channel Line Input (row D) and Channel Insert Send (row E). The signal processing configurations shown in Table 1 are then available. As the Insert point is being used as the exit point, the Fader in all of these configurations controls output level only, as it is not part of the feedback loop. Likewise, any processing occurring downstream of the Insert Point, does not participate in the feedback loop, and therefore, performs its expected duties: a Filter section applying filtering, an EQ section applying equalisation, and a dynamics processing section applying compression/limiting and expansion/gating. Note that, as shown in the block diagram (Figure 20), when the Dynamics section is applied to

\* Whereas Filters, EQ sections and Dynamics sections, when part of the feedback loop, have an unpredictable incidence on the Larsen effect’s timbre. For instance, adding gain to an EQ HMF band might have no effect at all on those frequencies and instead, break up the signal into intermittent low-frequency bursts.
Ch Out, it is placed post the Insert Point. One cannot therefore have the EQ then the Dynamics section, both before the Insert Point.

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Table 1: XL 9000 K Feedback Loop configurations with Channel Line Input and Insert Point

Table 2 shows the signal processing configurations obtained with the Input to the Channel selected as Line and patched to be the feedback loop’s entry point, and the Insert Point set to Pre, selected as the feedback loop’s exit point.

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Table 2: XL 9000 K Feedback Loop configurations with Channel Line Input and Insert Point Pre

A similar set of twin tables are obtained when selecting Mic instead of Line as the Channel’s input and feedback loop entry point, with either the Insert Point set to normal (Post) or Pre, set as the feedback loop exit point. The patch bay points used are Channel Mic Input (row B) and Channel Insert Send (row E). Yet another set of similar tables are obtained when selecting Group instead of Line as the Channel’s input and feedback loop entry point, with either the Insert Point set to normal (Post) or Pre, set as the feedback loop exit point. The patch bay points used are Multitrack Send & Group Monitor Input (row H) and Channel Insert Send (row E). These four tables can be found Appendix G, pages 3-4.
If instead of using the Insert Point as the feedback loop’s exit point, one uses the Fader’s output via for instance the Direct Out function, which supersedes Group Outputs (row G), this would lead to a set of three tables. Each one depicts whether Line, Mic or Group was selected as the Channel’s input and as the feedback loop’s entry point. Table 3 shows the configuration using the Line Input. The two tables using Mic and Group inputs can be found in Appendix G, page 5-6. By including the Fader in the feedback loop, this provides automated pitch control. Note that instead of using the Direct Out function to collect the Fader’s output signal, one could alternatively configure the Routing Matrix to source its signal from the Channel. This adds several elements to the circuitry, including the Group Trim amplifier. Although any of the 48 Multitrack Busses could be used, it is advisable to use the same number as the module, to keep things simple at the patch bay and on the console itself. In fact, to create a feedback loop using the Group Input with the Fader as the exit point, the Direct Out function cannot be used. Instead, one must use the Routing Matrix and again, for clarity, preferably assign it to the same Multitrack Bus number as the module’s. In all three cases, the feedback loop exit point on the patch bay is Group Output (row G), fed into either Channel Mic Input (row B), Channel Line Input (row D) or Multitrack Send & Group Monitor Input (row H).

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Table 3: XL 9000 K Feedback Loop configurations with Channel Line Input and Direct or Matrix Out

Additional configurations using an Aux Send Post-Fader as the exit point, includes the Fader in the feedback loop and also provides automated pitch control. This leads to three tables: when using the Line Input, the Mic Input, or the Group Input. Table 4 shows the configurations obtained using the Line Input. The other two tables are found in Appendix G, page 7. Using a Post-Fader Aux Send adds little circuitry compared to the configurations that use the Fader’s output as the exit point. More
specifically, compared to using the Direct Out function, it is expected that a Post-Fader Aux Send would add more or less the same amount of circuitry (copper tracks and electronic components) than using the Routing Matrix. Whether this produces an audible difference on Larsen effects, needed to be tested.

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<td>Fader</td>
<td>Aux</td>
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<tr>
<td>Line</td>
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<td>Line</td>
<td>Filter</td>
<td>Dyn</td>
<td>EQ</td>
<td>Ins</td>
<td>Fader</td>
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</tbody>
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Table 4: XL 9000 K Feedback Loop configurations with Channel Line Input and Aux Send Post

Having the Aux Send Pre-Fader as the exit point will take the Fader out of the feedback loop, but also presents a very interesting possibility (again, only the Line Input table shown here in Table 5, whereas the two others are found in Appendix G, pages 8-9). Although this is quite similar to the configurations when using the Insert Send Post – if one omits from the discussion the extra copper and electronic components that make up the Aux Send circuitry – there will be a significant difference should one want to use the Insert Point to add external hardware to the feedback loop and the Dynamics section set to Ch Out. Indeed, when the Dynamics Section is set to Ch Out, this places it after the Insert Point. In all of the configurations seen above, it wasn’t possible to have the Dynamics section after the Insert Point, without including the Fader in the feedback loop. What if one wants to have that configuration of Insert Point then Dynamics section, with the Fader acting as volume control? Using an Aux Send set to Pre is the solution. For this study, I have opted not to explore all the timbres that are possible when using external hardware processing via the Insert Point Send and Return loop. However, I felt that this was important to note here. Using the Insert Point to add external hardware to the feedback loop is addressed in 6.8.
Table 5: XL 9000 K Feedback Loop configurations with Channel Line Input and Aux Send Pre

The Insert Return Point can be set via the patch bay as the entry point for the feedback loop, with the Fader set as the exit point. This yields the single table of configurations in Table 6, as it no longer makes a difference if the Channel Input is set to Line, Mic or Group Input. The duplicate configurations are struck through.

Table 6: XL 9000 K Feedback Loop configurations with Channel Ins. Return and Direct or Matrix Out

Rounding off this review of possible configurations in the Channel, are two sets of configurations, using an Aux Send as the exit point combined with the Insert Return as the entry point (Tables 7 & 8). When using a Pre-Fader Aux Send, we are faced with the shortest possible feedback loop. We again come across the scenario where one is able to have the Dynamics section after the Insert Point, yet have the feedback loop’s exit point Pre-Fader, to automate volume rather than pitch.

Table 7: XL 9000 K Feedback Loop configurations with Channel Insert Return and Aux Send Post
Table 8: XL 9000 K Feedback Loop configurations with Channel Insert Return and Aux Send Pre

<table>
<thead>
<tr>
<th>Ins Pre</th>
<th>EQ</th>
<th>Filter</th>
<th>Dyn</th>
<th>Aux Pre</th>
<th>Fader</th>
</tr>
</thead>
</table>

The XL 9000 K also offers advanced flexibility when it comes to the options for positioning the EQ, Filters and Dynamics section in the Monitor path, but not to the extent seen in the Channel path. This is due to two important factors. First of all, the Filter can only be Post-EQ in the Monitor path, as the moment one presses Splt (for Split), this sets the Filters at the head of the Channel path. Second of all, the Insert Point resides in the Channel path; it cannot be placed in the Monitor Path. However, the Monitor path has a total of four possible inputs: Line via Tape Monitor Inputs (row K), Group via Multitrack Send & Group Monitor Input (row H) and two feeds from the Channel path. These are Pre Fader but Post Processing, when pressing Pre LF in the I/O module’s Monitor section or Post Fader when pressing the Pst LF button. Pressing both buttons, at least on the XL 9000 K at SAE Institute Byron Bay, is identical to pressing Pst LF on its own. These two feeds from the Channel path open the door to feedback loops involving both the Channel path and Monitor path.

The only possible exit points for a feedback loop in the Monitor path, are the Routing Matrix, via the Group Outputs (row G) or an Aux Send sourced in the Monitor path, Pre or Post the Fader controlling said path. If we first focus on feedback loops only involving the Monitor path, as we did for the Channel path, the Line and Group entry points and the three exit points, lead to a total of six tables found in Appendix G, pages 10-12. Tables 9 and 10 show two examples of these six sets of configurations.
Table 9: XL 9000 K Feedback Loop configurations with Monitor Line Input and Matrix Out

Table 10: XL 9000 K Feedback Loop configurations with Monitor Group Input and Aux Send Pre

A thorough study of feedback loop configurations would also consider using both the Channel and Monitor path, forming a greater-sized loop. Figure 20 shows the basic signal flow in the XL 9000 K, and by extension, most large-format in-line analogue consoles, when set to "Rec" mode in the Master Section. An example of an extended loop combining both paths, would be to set the Channel to its Mic Input (and having said Mic Input on the patch bay as the entry point), using the Routing Matrix to send the Channel's signal post-fader to the Monitor path set to Group Input, with an Aux Send sourced in the Monitor path as the exit point on the patch bay. Another example would be to have the console in Mix mode, which configures the paths as per Figure 21, and to use a Tape Monitor Input (patch bay row K) as the feedback loop’s entry point. The signal could be taken post-fader of the Monitor path by using the Routing Matrix which would be collected in the same Channel by selecting Group as its Input. The loop could be closed by using the Direct Out of that Channel appearing on Group Outputs (row G) as the exit point. The sheer volume of possibilities once the processing blocks are involved (Filter, EQ, Insert Point and Dynamics section) would have led to a disproportionate amount of time spent on one aspect of data collection, at the expense of other equally-important areas of the research, such as timbre storage and automation. I decided to suggest this for areas
of further study on the XL 9000 K – see 6.8. Likewise, Appendix G only lists feedback loop configurations when using a single path: Channel or Monitor.

Fig. 20: default signal flow in large-format inline analogue consoles, when recording

Fig. 21: default signal flow in large-format inline analogue consoles, when mixing
On the SSL Duality SE, the simplified signal flow – as this is not an in-line console per say – greatly reduces the complexity of available signal processing configurations. Figure 22 shows the console’s channel signal processing block diagram. The available configurations are similar to the ones obtained using just the Channel path on the XL 9000 K, and are listed in Appendix G, pages 13-21.

![SSL Duality block diagram](image)

**Fig. 22: SSL Duality SE signal processing block diagram**

The configurations discussed above for the XL 9000 K and Duality SE, form the basis of 4.3.

### 3.7.4 Addressing sub question – automation time-base precision

The next sub-question addressed whether the console’s automation system time base would be precise-enough to generate rapid, steady rhythmical patterns, akin to using a sequencer or drum machine to produce 16\(^{th}\) or 32\(^{nd}\) notes. The XL 9000 K automation computer is proprietary to the console, and uses timecode as its time base (see Appendix F – SSL XL 9000 K Manuals / Computer / Section 04 Events and Track Lists, pp. 4-5). To test the system’s time-keeping, a series of Mute and un-Mute commands could be programmed on a Channel path, using timecode addresses to position the events in time. Alternatively, a rapid fader movement cycle could be programmed and repeated, to test the responsiveness of the motorised faders.
On the Duality SE, the δ-Ctrl (or Delta-Control) plug-in, can be used in most DAWs to control any of the console’s Channel. The time-base precision should be dependant on the unit used on the DAW: usually, either bars & beats or timecode. The plug-in interconnects with the console via Mackie HUI into ipMIDI protocols, which could affect the time-base resolution and precision (Schmitt 1998; Wherry 2003). As with the XL 9000 K, a series of Mutes / un-Mutes, or cyclic fader movement was programmed to assess this DAW → Plug-In → Console system’s time-keeping precision.

3.7.5 Addressing sub question – automation fader-positioning precision

The next sub-question concerned pitch control. This is another aspect which involves precision, this time in the motorised fader positioning resolution, and its associated VCA. This would impact whether the console’s automation system allows precise-enough control of pitch, or whether the composition would be limited to using noise and textures, much akin to some of Merzbow’s work in the past decade (Doornen 2006). When the fader is included in the feedback loop, as per the configurations discussed above, would the automation system be precise enough to position the fader and more importantly, its associated VCA, to allow the programming of specific notes, to form melodies and chords? This could be tested by perhaps using a tuner connected to the console’s main output, to visually assess the precision in cents of various generated and automated Larsen effects. These time-keeping and pitch control accuracy tests, and other related findings, form the basis of 4.5.

3.7.6 Addressing sub question – timbre settings recall precision

Would the console’s Total Recall capabilities be precise enough to provide repeatability, or would a considerable amount of manual tweaking be necessary before each performance of the piece? A series of tests, involving the storage of a particular setting for a channel, then its recalling and subsequent comparison, would allow to assess that precision. These tests were done on both consoles, and form the basis of 4.4.
3.7.7 Addressing sub question – scoring

Would I have to develop my own symbolism to compose and sketch ideas on paper, or would I be able to rely on existing notation systems, such as the ones used by Pierre Schaeffer as early as 1948 for his Musique Concrète (Hodgkinson 1986), Stockhausen’s and Xenakis’ as seen in the literature review? These aspects were investigated along with the compositional stage as this seemed like the appropriate point in time to weigh the necessity (or not) of such a system. See Chapter Five.

See Appendix H for a full breakdown of the various stages of this research.
Chapter Four: Presentation of investigation data

In accordance with the designed research method, a step-by-step approach to answering the project’s sub-questions was implemented. Sessions at SAE Institute Byron Bay’s SSL XL 9000 K room, then the SSL Duality SE room, allowed hands-on investigation into each step’s particular areas, with video documentation. The findings were presented in the form of videos posted on the project’s YouTube and Facebook pages (all videos can be found under Research Video Files on the supplied USB flash drive), complemented by a written diary which documented as extensively as possible, each of those sessions (See Appendix A). This data was then synthesised into the following topics, relating to the research’s sub-questions: Preliminary investigation, Introductory video, Experimentation with signal flow and timbre exploration, Using the console’s computer system to store and recall timbres, and Using the console’s automation system as a sequencer.

4.1 Preliminary investigation

Session 00a – Associated video file: 00a CE3K Theme.m4v
https://www.youtube.com/watch?v=k0h7ylZ9A9Q
Duration: 0min 33sec

Before embarking on the full research project, I undertook some preliminary tests, to gauge the overall feasibility. This was particularly important as a casual conversation with Bruce McBean back in 2013 regarding my project idea*, hinted that all I might obtain as a result of my purposefully-induced Larsen effects, is ultrasonic signals.

On Sunday 27th of April 2014, a trial session took place in SAE Institute Byron Bay’s SSL XL 9000 K studio, which also allowed me to test the method I conceived for capturing proper stereo audio along with my live comments via a lapel microphone. Likewise, I needed to test whether my video capture system would be suitable for the blogged documentation aspect of the research. Finally, I wanted to experiment with

* Bruce McBean is one of Australia’s leading console maintenance engineer and designer. He was the manager of the Studios 301 Manufacturing Workshop based in Byron Bay’s industrial estate in Australia from 2006 to 2013. He designed the Neve Custom 75 series, and also supervised the manufacturing and shipping of the earlier units before Jan Muths took over management (Customseries75.com 2015; J Muths, pers. comm., 13 July 2015).
basic Larsen-generating setups in order to produce a short demonstration in time for a Work in Progress Seminar at Southern Cross University.

A few tests with the console signal flow immediately demonstrated that the Larsen effects produced were well within the humanly-audible spectrum. It also confirmed that there was great potential in the variety of timbres that could be generated, from just exploring 3 signal path configurations, as well as the unstable nature of Larsen effects should one decide to work close to unity gain.

Having set on producing the five-note theme from Close Encounters of the Third Kind (Spielberg 1977), the only issue encountered was that the computer running the automation was being serviced and was therefore unusable. Hence, I had to perform manually, which led to a few timing issues in the final piece, other than the deliberate rallentando mimicking that of the film. The sounds obtained from just a dozen of console channels were nonetheless quite promising, and have hopefully communicated to the seminar’s participants, the underlying potential had all 112 signal paths been producing signals, handled by the console’s automation.

In the process, it became apparent that, just as most guitarists would resort to the use of a tuning device of some sort to get each string to near-perfect pitch, I too would need visual feedback when tuning the various notes. This was achieved by inserting a tuner plug-in: Trillium Lane’s InTune (Trillium Lane Labs n.d.) on an extra Pro Tools (Avid 2014) Aux Input path, fed from the console stereo master output. With it, I could visualise in real time, the pitch of any signal being output by the console.

Session 00b – Associated video file: 00b Skweeel Trailer.mp4
https://www.youtube.com/watch?v=gDfvkpsfL8Y
Duration: 0min 22sec

To maximise awareness for the project and therefore, generate as much feedback from the social media channels as possible, I felt it was necessary to have a recognisable and exciting title sequence, as per other successful webcasts such as Mental Floss, I F#$!ing Love Science or FPSRussia (YouTube 2011; 2013; 2010). The soundtrack for this short Trailer sequence would have to quickly grab people’s
attention, showcase a variety of timbres, and have an impressive climax, encouraging people to watch the rest of the episodes. I drew my inspiration from THX’s first ever title sequence, *Deep Note* (THX.com 2006), as it had exactly that effect on me when I first saw it opening for Lucas’s *Return of the Jedi* (Marquand 1983). I also drew my inspiration from *Hunt for Red October* (McTiernan 1990), with its stylised sonar “pings”, as my early trials prior to session 00a had revealed that the onset of a Larsen effect, if quickly attended to, results in a similar kind of brief, high-pitched sound.

With these two aspects in mind, I set out to capture a "sonar ping", which morphs into a rich, deep, chaotic multi-layered synth-like pad, itself converging to a final, pleasantly-tuned unison, reminiscent of THX’s *Deep Note* finale. It was also important to capture the visual action that led to those sound changes, to provide authenticity to the whole process – as opposed to a soundtrack accompanying unrelated imagery.

Had the automation system been functional, the faders on the channels responsible for the sawtooth cluster would have been programmed to move from slightly under unity gain, to a position where the sound settled to a pre-determined note, in tune with the other channels. This had to be done manually, reverting to a standard method used when mixing on a non-automatable console: grease pencil markings alongside the faders allow precise-enough positioning if the marks have been carefully drawn in. I chose not to film the hands moving the faders, as I did not want the title sequence to imply in any way that the project involves, or focuses on manual positioning of faders – its goal being to have as much as possible performed automatically by the automation computer.

The rest of the session was spent filming various elements that would most likely be explored in the project, including several expander-gates seemingly opening and closing according to a common trigger signal. I also wanted the final shot to summarise the project visually, as the sounds converge to unison. A signal flow schematic featuring potential feedback loop paths was purposefully crudely drawn on a clipboard, as to suggest much trial and error, and positioned on the console’s Master Section in such a way that it is still apparent that the console in question is a complex, large-format model. Patch cords were also framed in that shot, straddling a
few channels and the Master Section, to indicate that they form a crucial component of the project, as they would be establishing most of the feedback loop connections evoked by the drawn schematic. The frame composition allowed for subsequent logo and titling overlay.

The session was successful as it has achieved all the goals I had set, of producing a suitably exciting, appealing title sequence to introduce each of the project's episodes. It reinforced my confidence in the potential for the project to deliver an impressively complex piece of music, considering the title sequence only used four channels of the hundred-and-twelve signal paths available in SAE Institute Byron Bay's SSL XL 9000 K console frame.

4.2 Introductory video

Session 01 – Associated video file: 01 Skweeel Episode 1.mp4
https://www.youtube.com/watch?v=yvltn3-VCsM
Duration: 5min 20sec

To generate maximum interest for the project in the audio engineering and music community on the social media networks, and hopefully to an even wider public, I felt the first official episode of the research project should present its objectives in as simple a language as possible, in a short time frame. I decided that five minutes would be an acceptable duration if I could capture people's interest early on in the video. The title sequence that would introduce the episode would also hopefully consolidate that effect.

I set out to introduce the project's main goals as my opening shots. This was then followed by a demonstration of the Larsen effect playing its role of a VCO equivalent in a traditional analogue subtractive synthesizer. At this stage (01:18), I refer to the two main types of sounds that I have stumbled upon so far in the initial tests for videos 00a and 00b: Squeal and Flatula, which in my opinion, adequately described the sonic qualities of the two timbres – especially for the latter, according to Chion's "causal listening mode" (Chion 1994, pp. 26-8).

I then demonstrated how the filter section and EQ section combined, can play the role of a traditional synthesizer's VCF section (01:30). This was followed by a
demonstration of one of possibly many applications of the expander/gate section (02:20), here, replicating the effect of an LFO affecting a synthesizer’s VCA. As the console automation was not functional at the time, I had to manually demonstrate how the fader could potentially be used to control the Larsen effects’ pitch (02:53). The automation system was also envisioned to control every other aspect of the musical performance, including the sequencing of notes, and other control signals such as effects sends on/off, filtering on/off, etc., as would a traditional CV/Gate sequencer (03:24).

The video ends with a word of caution should people want to try these techniques at home, in accordance with the ethical considerations outlined in 3.5. The XL 9000 K’s surround compressor was inserted on the Master Output, as part of the production template for all videos and final performance, to limit the signal amplitude before harmful SPL is reached.

The session was again quite successful, as it has achieved all but one of the goals I had set, of producing a broad overview of the project’s scope. Had the automation computer been functional, the full potential of the project and its final piece of music would have been better demonstrated. The video encouraged me to use a less scripted approach as the verbal delivery felt too contrived and artificial.

4.3 Experimentation with signal flow and timbre exploration

Having had the project outline described in Episode 01, it was then time to examine the many different feedback loop signal flow possibilities within a channel, to explore the various timbres that could be obtained. For a full list of possible feedback loop configurations within either a Channel or a Monitor path, refer to Appendix G. The information in this sub-chapter was synthesized from the following sessions and associated videos – see Appendix A for detailed content.

Session 01 – Associated video file: 01 Skweeel Episode 1.mp4
https://www.youtube.com/watch?v=yvlt3n3-VCSM
Duration: 5min 20sec

Session 02 – Associated video file: 02 Skweeel Episode 2.mp4
https://www.youtube.com/watch?v=jGl_yCh8wTI
Duration: 22min 00sec
Timbre exploration was first carried out on the SSL XL 9000 K, by using the I/O module’s Channel path with the Line Input as the feedback loop’s entry point. The Line Input pot provides a range of -20 dB to +20 dB level trim, with a centre indent, at unity gain (Solid State Logic 2003, Console Section 03, p.3-2). As the onset of a Larsen effect is just over unity gain, this centre indentation could be a problem when trying to set the pot ever so slightly beyond that position, for precise control of the Larsen effect in its early development stage, if using the Insert Point as the exit point. I will often refer to the setting where the feedback loop’s overall gain is just slightly positive (the onset of the Larsen effect), as the “tipping point”. Figures 23, 24 and 25 show a sample of the many configurations tested on the XL 9000 K using the Line Input as the entry point:

![Diagram of XL 9000 K Channel Insert Send to Line Input, with the EQ section in the loop](image)
The following spectrograms show the ten different timbres which were identified and named on the XL 9000 K. To view full-sized versions, see Appendix A – Sessions 01, 02 and 03.

Fig. 24: XL 9000 K Channel Insert Send to Line Input, with EQ in the loop and Dynamics post loop

Fig. 25: XL 9000 K Channel Direct Out to Line Input, with Dynamics first then EQ in the loop

Fig. 26: Squeal timbre at 01:13 in Ep. 01 (= 650ms displayed)

Fig. 27: Flatula timbre at 01:21 in Ep. 01 (= 250ms displayed)
Fig. 28: Throat-Like timbre at 07:21 in Ep. 02
(≈ 400ms displayed)

Fig. 29: Low-Frequency Pulse timbre at 08:12
in Ep. 02 (≈ 300ms displayed)

Fig. 30: Pulse Width Modulation timbre at
10:00 in Ep. 02 (≈ 2,800ms displayed)

Fig. 31: Constipated Flatula timbre at 10:43 in
Ep. 02 (≈ 400ms displayed)

Fig. 32: Frog Call timbre at 20:14 in Ep. 02
(≈ 400ms displayed)

Fig. 33: Ring Modulation timbre at 12:23 in
Ep. 02 (≈ 2,000ms displayed)
There are two basic timbres that result from an unprocessed feedback loop involving the Line Input. Whenever the feedback loop gain is slightly over the tipping point, the Squeal timbre shown in Figure 26 is generated. The term Squeal will be used to denote any Larsen effect that manifests itself as a high-pitched, relatively pure timbre (i.e., with few, well spaced-apart and equally distributed harmonics). It can also be heard in Episode 02, at 03:43. When one pushes the gain far beyond the tipping point, the Squeal timbre progressively transforms into the Flatula timbre. This is accompanied by a change in pitch: a Flatula timbre will always be lower than the Squeal timbre from which it originated. Flatula will be used to denote any Larsen effect that manifests itself as a deep, low frequency sound, with extensive harmonics filling the entire spectrum; the flatula connotation is derived from the short-spaced interruptions in the signal, as per Figure 27. It can also be heard in Episode 02, at 03:30.

With the EQ activated inside the feedback loop as per signal flow Figure 23, two new timbres were found in the process. The first one has been found to be “throat-like”, reminiscent of the harmonics heard in Tibetan monk “throat singing” or “overtone singing” (Hightower 2004). It seemed to be mostly recurrent whenever the initial Larsen effect is resting between the Squeal and Flatula position, and negative gain is applied in any of the four EQ bands – although it is best represented when either of the two mid-bands are used. The Throat-Like timbre (see Figure 28) is also akin to the typical sounds one gets from a Texas Instruments Speak & Spell™ which has been subject to some circuit bending (Pill Shovel Productions n.d.). With the EQ’s LF band, combined with pronounced input gain, interesting low-frequency pulses could be produced. These could be either used as a sound, or as a trigger signal to
generate rhythmical patterns, as was shown in Episode 01 (at 02:40) when demonstrating how some timbres could be used as an LFO. This type of timbre has been named Low-Frequency Pulse (see Figure 29).

With the EQ section positioned after the loop, it behaves like a VCF would in a conventional subtractive synthesizer: changing the relative levels of wide frequency bands and/or specific ranges of harmonics, depending on Q settings. This yields a type of timbre which resembles what one would obtain when using Pulse Width Modulation on a Square wave in the VCO of a subtractive synthesiser. The name for that timbre therefore became Pulse Width Modulation, as seen in Fig. 30. In hindsight, I should have predicted that equalising a timbre that is quite rich in harmonics in such a way (sweeping with a relatively high Q factor), would produce this type of timbre.

When adding the Dynamics section into the feedback loop as per signal flow Figure 25, three new timbres were generated when using the Compressor. With a Larsen effect resting between a Squeal and a Flatula, increasing the Ratio on the Compressor, with a Fast Attack setting, produces a variation on said Flatula that I deemed pronounced-enough to deserve its own denomination: Constipated Flatula (see Figure 31). When using a Slow Attack setting on the Compressor, various types of slow pulses were obtained; derivatives of the Low-Frequency Pulse timbre, with different harmonic content: these reminded me of certain frogs in the Northern Rivers of Australia, and as such, the timbre was named Frog Call (see Figure 32). When lowering the Threshold, a new timbre was generated, which sounds like Ring Modulation (see Figure 33). When placing the Dynamics section after the feedback loop and using the Compressor, the dynamic range is affected, as one would expect. Notably, on the “Pulse” types of timbres, the Slow or Fast attack setting has an interesting influence on the high-frequency content of the attack transients. Overall, the Expander/Gate did not seem to have any influence on the sound. I have attributed that to the fact that the audio signal level in a Larsen effect is far beyond what the Threshold control would be able to reach. As such, the Expander/Gate was no longer used in subsequent timbre investigations.
Using the Auxiliary Send as an exit point did not introduce any new timbres. I attribute this to the fact that unlike an EQ, Filter or Dynamics section, which relies on many electronic components, an Auxiliary Send would be kept to a bare minimum as its role is just to apply gain. It would be comprised of an amplifying circuitry not that different from the output buffer of an Insert Send (fixed Line-level gain) or Fader output (variable Line-level gain). The Auxiliary Send was no longer used in subsequent timbre investigations.

When using the Mic input as the entry point, as per Figure 36, the results were puzzling at first. The Mic Input gain pot covers a range of +10 to +70 dB, which can be shifted down with the -20 dB pad switch, providing a range of -10 dB to +50 dB (See Appendix F – SSL XL 9000 K Manuals / Console / Section 03 Input-Output Module, page 3-2). The data alone suggests that without the Pad engaged, the feedback loop is immediately generating a Larsen effect, since at its lowest setting there is already 10 dB of positive gain applied in the loop. This was verified in Episode 03 (06:48).

What the data does not imply, however, is that the Larsen effect generated by this new setup is inaudible; it is not within the 20 Hz to 20 kHz frequency range. This was corroborated by the fact that the channel meter was showing ample signal level – as with regular (audible) Larsen effects – as well as the master output meters of the console, as seen at 07:00 in the video. As I was yet to encounter the ultrasonic Larsen effect predicted by console designer Bruce McBean (2013, pers. Comm. September), I suspected this was the particular setup he envisioned when I explained to him my idea for the research project. Figure 37 shows the spectrogram of this ultrasonic Larsen effect, when using the Mic input without any modification: the lowest frequency in the signal is 56 kHz.
For reasons which are beyond the investigative scope of this research, activating the Polarity Invert (Phase Invert) function lowers the Larsen effect signal’s components back into the audible range. The resulting timbre heard in Episode 03 at 09:04 is interesting, as it is quite rich in harmonics; richer than the Squeal timbre, without breaking up into a Flatula timbre. Inspecting the signal’s fundamental and harmonics frequencies (whole multiples of 250 Hz up to Nyquist), everything indicated that this new timbre should be named Sawtooth (see Figure 34 on page 86). It does appear, though, that the odd harmonics are more pronounced in this new timbre. In a true Sawtooth waveform, all harmonics are at an equal level relative to each other. Furthermore, it seems to be quite stable in pitch no matter where the Mic Gain pot is positioned, which in the video, I initially reported as being uninteresting (09:30).

The Pad (20 dB) used on its own, for reasons also unknown due to my limited understanding of the electronics involved, lowers the Larsen effect signal’s components back into the audible range. Working close to the tipping point produces the standard Flatula and Squeal timbres. However, once well beyond the tipping point, the Larsen effect timbre morphs into a new timbre which cycles between short periods of silence, and a sound which resembles fabric being torn. I have therefore named this new timbre Tearing Fabric (see Figure 35 on page 86). The duration of each event in the cycle varies somewhat, which would make it difficult to use to produce reliable rhythmical patterns. The timbre, though, is so unique and interesting in itself that I decided to feature it in the final composition nonetheless. Combining
the Pad and the Polarity Invert functions shifts the Larsen effect back into the ultrasonic range.

The Monitor path used on its own to create a Feedback loop, as per Figure 38 for instance, did not yield any significant variations of existing timbres, to warrant cataloguing them as new. The “rich low-frequency pulse” heard in Episode 04 at 05:52, upon further investigation in Session 05, can also be obtained in the Channel and is just a variation of the already-established Low Frequency Pulse timbre.

![Monitor Path Diagram](image)

Fig. 38: XL 9000 K Monitor path input connected to its own Group Output, by assigning it to the same module number, in the Routing Matrix

Finally, a combination of Channel and Monitor path, such as the one shown in Figure 39, was tested in an effort to extend the dynamic range of the tipping point to a more manageable fader movement. By cumulating two faders, I was hoping to extend the fader movement leading from silence, to where the Larsen effect is well established. From 10:10 to 12:20 in episode 04, shows that my hypothetical method for improving tuning is no better than when using just one path.
Furthermore, as seen at 12:27, if I wish to make a particular sound appear or disappear, by using the Cut switch, I must use a signal flow configuration where the cut switch is downstream from the feedback loop’s exit point. Otherwise, if the cut switch is part of the feedback loop’s circuitry, the Larsen effect’s pitch is completely lost and requires a fairly long time to stabilise again – and most likely, not at exactly the same pitch achieved prior to applying the cut.

The sheer volume of possibilities when both Channel and Monitor paths form the loop, once the processing blocks are involved (Filter, EQ, Insert Point and Dynamics section) would have led to a disproportionate amount of time spent on one aspect of data collection, at the expense of other equally-important areas. I decided to suggest this for areas of further study on the XL 9000 K; see 6.8.

My method of randomly selecting I/O modules on the XL 9000 K to validate newly-found timbres (see 3.3), has had an interesting outcome. I noticed that each time a particular module would not behave like the others, coincidentally, that module happened to have a fault somewhere in its circuitry. For instance, that module might
have an Aux Send On/Off LED refusing to turn off. This could have applications in console maintenance and is discussed in 6.7.

Once it became clear that I would not be able to complete the project on the XL 9000 K, due to its automation computer malfunction, the research was ported to the Duality SE. Timbre investigation on the new console was not led the same way as on the XL 9000 K. To make up for lost time, I felt the priority would be to assess whether the timbres already found on the XL 9000 K were also available on the Duality SE, instead of investigating everything from scratch. The following paragraphs document these findings, compiled from Session 06 (see Appendix A).

The timbres that were possible to replicate on the Duality SE are the Flatula, Constipated Flatula, Low Frequency Pulse, Ring Modulation and Pulse Width Modulation timbres using the Line input, and the Sawtooth timbre using the Mic input. Timbres which could not be achieved on the Duality SE are: Throat-Like and Tearing Fabric.

The first pertinent difference found, was that the Squeal timbre on the Duality SE is not as rich in harmonics as its XL 9000 K counterpart (see Figures 40 and 41). SSL states, on page 2 - Section 1 of the Duality SE operator’s manual, that the “audio bandwidth exceeds that of 192 kHz recorders” (Solid State Logic n.d.-b). This indicates a bandwidth of at least 80 kHz. The XL 9000 K has a 120 kHz bandwidth (Mozart & Friends Limited 2015). The spectrograms generated from my Larsen effect signals, ranging up to 20 kHz, would indicate that although this difference in bandwidth behaviour of the circuitry might not be audible or measurable when using normal signals in music production (i.e., signals below the audio path’s peak diode threshold), when overdriving the circuitry, there is a notable difference between the two consoles.
Following the posting of Episode 06 to YouTube, Daniel Duskin commented: "I wonder if using the VHD section could add the additional harmonic content you desired from the 9k" (D Duskin, pers. comm., 17 October 2017). Subsequent testing showed no significant difference or improvement in harmonic content, using the Drive pot in the channels’ input stages.

The Frog Call is different on the Duality SE, as shown in Figures 42 and 43: the energy is concentrated on a low-mid spectrum (400 to 800 Hz) on the XL 9000 K whereas it spreads uniformly from DC to approximately 5 kHz on the Duality SE. Additional filtering would provide an acceptable approximation, however.

Another difference is that without the EQ section activated in the loop, the range above the tipping point, which separated the high-pitched Squeal timbre from the lower-frequency squeal, then leading into the Flatula timbre, is much smaller than on the XL 9000 K. In fact, I was not able to generate a high-pitch Squeal at all (as heard at 12:30, 12:59 and 14:18 in Episode 06). On the other hand, an interesting
advantage of the Duality SE in this configuration seems to be that the Low Frequency Pulses derived from going into “extreme Flatula”, are low-enough in tempo to be useable as rhythmical patterns. These can potentially act as a trigger to open/shut expander gates or even act as a makeshift kick drum or other percussive sounds.

Finally, a finding of prime importance for the research is that when activating the EQ in the configuration shown in Figure 44, as demonstrated in Episode 06 at 16:06, not only are higher-pitched Squeals available but pitch control is obtained. On the day, and in that particular configuration and EQ setting, I was able to achieve a range of 4 semitones when moving the fader. Although the Duality SE presents only 48 audio paths, against the XL 9000 K’s potential 112 feedback loop paths, being able to control the pitch on all of these 48 paths more than compensates for the loss of polyphony.

I would like to conclude this chapter on timbre investigation with this final note. When I was summing signals on the XL 9000 K in my preliminary investigation, using Sawtooth timbres on several channels, I noticed that adding a third Sawtooth to two existing ones already paned full left and right, resulted in a thinner overall sound. This was most likely due to phase cancellation. I tried different combinations of polarity reversal switches, to no avail. I have not met this issue on the Duality SE at any stage of the composition, where several identical timbres are at times summed to the main output.
4.4 Using the console’s computer system to store and recall timbres

The information in this sub-chapter was synthesized from the following sessions and associated videos – see Appendix A for detailed content.

Session 05 – Associated video file: 05 Skweeel Episode 5.mp4
https://www.youtube.com/watch?v=m5sJyd7A3ps
Duration: 21min 20sec

Session 07 – Associated video file: 07 Skweeel Episode 7.mp4
https://www.youtube.com/watch?v=N9Mw_F5hCLc
Duration: 12min 42sec

4.4.1 Brief history of console settings storage and recall systems

“SSL introduced Total Recall on their 4000E console in 1981. Total Recall stored, in the SSL computer, the positions of each pot (potentiometer) and switch on the console. The entire console or any part of it could be reset later by manually matching the positions of the pots and switches with those previously stored” (Burgess 2014, p. 101).

The principle by which potentiometer and switch positions are stored by an analogue console recall system is quite simple: when the user tells a recall system to store a “snapshot” of the current console surface, the system sends specific voltages to every single pot and switch. The return voltage is measured and represents the position for each pot and switch. Switches present a very logical yes/no situation: does the voltage that was sent, come back at the same value (the switch was On) or does it not come back (0 Volt is measured – the switch was Off). The voltage is converted by an ADC located on the I/O module’s circuit board, so that the computer can store the switch status: 1 for On and 0 for Off (although some systems might have chosen to use 1 for Off and 0 for On). For potentiometers, the return voltage is measured, and once passed through the ADC, will be stored with a range of values from 0 (pot fully counter-clockwise) to X (pot fully clock-wise); X represents the resolution at which the various positions are stored.

When the user wishes to recall a specific snapshot of stored pot and switch values, a similar process is applied, this time involving a comparison of stored binary values, and real-time data provided by the ADCs. The recall computer will again send specific voltages, which will again be converted to binary values reflecting the current
positions. These current positions are compared to the stored positions to determine which pots and/or switches are in an incorrect position. Some sort of visual feedback is required so the user can see where the pot/switch is supposed to be and where it is currently, as to avoid having to constantly look between screen and console surface. For faders, this was originally done by the means of two or more LEDs, indicating whether the fader needs to be raised or lowered, to match the automation’s VCA value (Solid State Logic 1987). However, it would be difficult, space-wise, to implement such a system for every potentiometer on a complex analogue console: the SSL SL 4000 G has thirty potentiometers per I/O module - the XL 9000 K has thirty-two. A video display became necessary, showing both current and stored pot positions and switch statuses. Over the years, Recall computer graphics have gone from very crude, 8-bit video renderings of the console’s various sections (SSL 4000 E and G), as seen in Figure 45, to today’s near-photographic renditions, as seen in Figure 46.

AMS were able to use fibre optics to display a ring of lights on top of the potentiometers of their Logic 1 digital console in 1988. This would display the computer automation value at all times. One should bear in mind, though, that the Logic 1 only sports 8 potentiometers per module, and that this would most likely not be economically viable for 30 or 32 potentiometers. They nicknamed this display system the Logicator (Studio Sound 1988). Other companies have since then used a similar system, with a ring of LEDs more often surrounding the pot, which has become the de-facto standard for displaying potentiometer values on digital consoles and control surfaces.
To physically reposition a potentiometer so that it matches the stored position, the user would be assisted by the video display: two marks would identify the current and stored positions. On the earlier Total Recall displays, this would be by the means of a white highlight for any misaligned pot position (current position) and a dark grey mark (stored position), as seen in Figure 45 (Solid State Logic 1987). On the XL 9000 K Total Recall system, the pot is shown as it is on the console (current position) and a yellow ring around it not only identifies which pots are not in the correct position, but also shows the stored position as a gap in that ring, as seen in Figure 46 (Solid State Logic 2002b).

Once the user has re-aligned the current, physical position, with the stored position, visual feedback is given. On the earlier Total Recall displays, the white highlight disappears (Solid State Logic 1987). On the XL 9000 K Total Recall system, the yellow ring disappears (Solid State Logic 2002). In an attempt to minimise the amount of time spent looking at the screen so one could focus on the console surface, a voice prompt system, called Vocal Recall, was developed by Amek for their Big by Langley console, to let you know if you needed to raise or lower your pot value – also telling you when matching was achieved (Amek 1993).
Analogue console Recall systems are prone to some issues. First of all, there is that of the precision at which the position is stored in the computer (discussed below). There is also a problem that could arise if between the moment the positions were stored and the moment they are recalled, modules have been switched around in the console’s frame.

I recalled a mix at Enterprise studios in Burbank, California, that sounded nothing like the original mix, despite it having been done in the same room. It turned out that a maintenance person had been repairing some modules and had changed their position in the console since the original mix had been done (Burgess 2014).

The issue described by Burgess was more often caused by a common maintenance engineer procedure: several spare modules are usually kept so that they can be used to fill in the gaps, whilst the faulty modules are being repaired. In both cases, having a different module in a specific position results in slightly different potentiometer values being displayed, as the DCAs of the new module will behave differently. A prime example, shown at 19:33 in Episode 05, is that of potentiometers which have a physical indentation, such as pan pots. Whenever such a pot is stored nesting in its indentation, and the module’s Total Recall information is either copied to another module, or the module has been swapped, recalling the information in the new module will more often than not show that the pot is incorrectly set, even though it is properly nesting in its indentation. Attempts to rest the pot on the edges of the indentation, to satisfy the computer, are often futile. The inaccuracies of the principle, however big or small, are summarized by Burgess when it comes to the conventional usage of recalling a mix: “with most recalls, a degree of fine tuning by ear was necessary” (2014). This would prove to be even more so, for No-Input Mixer music applications.

4.4.2 The SSL XL 9000 K Total Recall system

For a complete overview of the SSL XL 9000 K’s Total Recall system, please consult Appendix F – SSL XL 9000 K Manuals / Computer / Section 07 Total Recall. The console’s manual claims that its Total Recall system allows “the console to be reset to within 0.25dB tolerance” (Solid State Logic 2002, Computer manual, Section 07 Total Recall, p. 7-1). Nowhere does the Computer Manual mention how many bits are used to achieve this precision. After several sessions devoted entirely to reading the extensive literature provided by SSL when purchasing the console, I was able to piece together the information, from the Computer Service Manual.
In terms of measured values, switches are switched between 0V and +5V and so should switch between approximately ‘0’ and approximately ‘1023’ whilst potentiometers should exhibit a linear relationship between position and value but again are bounded by ‘0’ and ‘1023’ (Solid State Logic XL 9000 K Computer Service Manual 2002, p. 35).

The lower and upper values of 0 and 1023 are indeed those of a 10-bit system ($2^{10} = 1024$ values). Switches on the XL 9000 K either pass the +5V voltage if they are On, which the ADC sees as a value close to maximum (1023), or they do not pass the voltage if they are Off, which the ADC sees as a value close to minimum (0). As there is no need to store more than one bit of information for a bipolar switch position, this, in turn, is stored in the computer’s Recall file as either a 1 for On or 0 for Off. The full counter-clockwise to full clock-wise range of a potentiometer, will register as 0 to 1023 at the output of the ADC, and will be stored as such.

If we know the mechanical angle that the average potentiometer on the console travels between its two extreme positions, we can calculate the storage precision, in degrees, by dividing the pot’s mechanical travel by 1024. I had always assumed that a typical Mic Input Gain pot or Channel Pan pot, had a total mechanical travel of $\frac{3}{4}$ of a turn, i.e. 270°. Some top websites explaining how potentiometers work will actually state this (Electronics-tutorials.ws 2016; Elliott 2002). However, a scan of a few pots taken from the console manual, and measured using Photoshop, reveals that the mechanical travel is closer to 300° between the two extreme markings, than 270° (see Figure 47). This was then confirmed by reviewing the mechanical specifications of potentiometers designed specifically for audio applications (Altronics 2016; Element14 2016; RS Components 2016). This was further confirmed by Jon Pinkerton, Senior Technical Officer for the School of Arts and Social Sciences at Southern Cross University in Lismore (J Pinkerton 2016, pers. comm., 11 October).
Fig. 47: Measurement of average mechanical angle on XL 9000 K potentiometers

Using 300° as the reference for the average mechanical travel of audio potentiometers on the XL 9000 K, we obtain a storage precision of roughly 0.3°, as $\frac{300}{1024} = 0.293$. Had the mechanical range been 270°, this would not have made a substantial difference, as the precision would then be 0.264°. It would seem that a 10-bit resolution is enough, in regards to a human’s hand ability to turn a potentiometer by increments of 0.3 degrees. Would this be precise enough, though, to accurately control the timbre and pitch of Larsen effects? A session in the studio was therefore organised to measure the XL 9000 K’s Total Recall system precision. This led to the production of the video for Episode 05.

The testing was conducted by selecting two I/O modules at random, dialling in a Larsen effect timbre on one’s Channel, and without listening to the signals, replicating the settings on the other. The difference is then measured between achieving this visually, or by copying the settings from one Channel to the other by using the Total Recall system. The results of test #1 for the visual copy method, heard at 05:20 in Episode 05 is seen in Figure 48. This led to a 6-semitone difference between the two signals.
Figure 49 shows the result of the timbre difference using the Total Recall system to port settings from one module to the other. This led to a 4-semitone difference between the two signals.

The results of Test #2 using different Channels are shown in Figures 50 and 51. The manual copy method yielded a 7-semitone inaccuracy, whereas the Total Recall copy method only had a 2-semitone difference between the two signals.
A replication leading to identical timbre and a pitch difference to within just a few cents, was not achievable. This indicated that the precision of the XL 9000 K’s Total Recall system might not be sufficient for efficient storage and retrieval of timbres, and that pitch control might not be achievable. Bill Wade, manager at SAE Institute Byron Bay, who shares a similar background in audio engineering as mine (2016, pers. Comm. 10 October), suggested that to better ascertain the precision of the Recall system, I needed to use a single module, instead of two. Indeed, if I were to dial in a timbre on one module, store those settings and record the resulting sound, then reset the module, recall the settings and record the resulting sound, I would eliminate all other factors, such as potentiometers not having the exact same values. I would therefore truly measure the accuracy of the Recall system, whereas what I had done so far could also have highlighted discrepancies between the electronic components of two different modules; this method was implemented and discussed below in 4.4.3.
4.4.3 The SSL Duality SE Total Recall system

Known as the Multi-User Total Recall, this is an optional feature on the Duality SE. The operator’s manual, found in Appendix F – SSL Duality SE Manuals / SSL Duality SE Manual, does not indicate the precision at which settings are stored, or recalled. Just as I had to go through the XL 9000 K’s Computer Service Manual, to piece together that the storage resolution is 10-bit on that console, I suspected I would have to do the same to figure out the storage resolution on the Duality SE. I was unfortunately unable to locate any document with that information for that console. All things tended to indicate, however, that the resolution on the Duality SE is at least that of the XL 9000 K, if not better, as per my findings in the experimentation of the subject, documented in part via Episode 07.

Implementing Bill Wade’s suggestion of using just one Channel on the Duality SE, led to the following tests: a Larsen effect timbre was randomly produced on a channel, as seen at 01:49 in Episode 07. The audio output was recorded to a DAW track. The settings for that channel were stored into the Total Recall system. The parameters on the channel were then reset – I chose to return the settings to their default position, although any other type of position shuffling would have produced the same outcome. The channel’s settings were then recalled, which led to a second audio signal, also recorded to the DAW. The exact pitch of both signals could then be analysed to determine the accuracy of the storage/recall process for that particular sound. The result of the first test can be heard at 05:56. These two spectrograms can be seen in Figure 52. The two signals are only 4 to 5 cents apart.

Another test was performed on a different channel, with a more complex timbre (a Frog Call derivative), the result of which can be heard at 10:30 in the video. In this test, both pitch and the amount of pulses per minute, could be compared for accuracy. The two spectrograms for test #2 can be seen in Figure 53. With spectrogram analysis (as a tuner would not lock onto any particular pitch), the “source” signal’s pitch was estimated to be 480 Hz, whereas the “target” signal produced a slightly lower pitch of 460 Hz. By calculating the time separating a large group of pulses (only 4 are shown on the spectrograms), the number of pulses per minute (ppm) extracted from the “source” signal was 712 ppm, whilst the “target”
signal’s was 704 ppm. This is again remarkable, when one considers how little difference there is between a 712 and a 704 ppm rhythm: less than 1 % (0.98876).

This is in stark contrast with the accuracy measured on the XL 9000 K, which was at best, around 2 semitones (200 cents). Sadly, the computer on that console having then become permanently inoperative, I could not go back to assess whether or not
this much greater difference in pitch was due to my flawed method of using two different channels, or the XL 9000 K Total Recall system’s actual inaccuracy.

4.4.4 Issues with the Duality SE’s Total Recall system for No-Input Mixer music applications

When exploring timbres, whether on the SSL XL 9000 K or the Duality SE, I quickly noticed that in order to fully retrieve a particular timbre, I could not count entirely on the consoles’ Total Recall systems. On the XL9000 K, it is quite clear if the Group input was selected as the entry point for the feedback loop in an I/O module’s channel, as the Total Recall screen would indicate that the SUB GP switch must be in its depressed state by having it highlighted in yellow – see Figure 54. There is also clear indication as to whether the Mic or Line input has been used, by means of an associated red LED (Mic) or green LED (Line); both LEDs are off when the Group signal feeds the channel, which is yet another method to visually ascertain when that input is selected. This in turn informs which entry point to use on the patch bay when establishing the feedback loop with the patch cord. However, one would not be able to tell from the other recalled settings, whether the Insert Send, Direct Out, or any of the Aux Sends were used as the exit point for the feedback loop.

![Fig. 54: SSL XL 9000 K Total Recall screen (top section only)](source: Solid State Logic (2002a))
Whereas on the Duality SE, the channel strip itself does not indicate which of the Mic, Line or Bus (equivalent to the XL 9000 K’s Group) input has been used, nor does the Total Recall screen provide that information when displaying that channel’s settings. When using the console for “normal” applications (i.e., not for No-Input Mixer music), this is not a real issue, as the operator would most likely just want to reset all the non-latching switches to whichever state they were in on the day the data was stored. As found in chapter four of the Duality SE’s operator’s manual (p. 6), “a double press on a Fader status button, or on Set Sel or Set All, will reset all the non-latching functions on that channel strip, channel strips selected on the central routing panel, or all channel strips respectively” (Solid State Logic n.d.-b). On page 5 of the same chapter, Figure 55 shows the Total Recall screen’s accurate reproduction of the actual physical settings; there is no LED associated with the red Variable Gain Input Amplifier (equivalent for the sake of this discussion, to the Mic Input gain) pot or the blue Gain Trim (equivalent to Line input) pot in the top left corner.

Fig. 55: SSL Duality SE Total Recall screen for individual channels
Nor is there a Bus LED to indicate that signal as being the source for the channel. This information is normally visible when not using Total Recall, at the bottom of the TFT screens in the meter bridge, as seen in Figure 56. Below the meters, Input (in red) or Line (in blue) indicates which of the Mic input or Line inputs have been used. Another method for visualising which input was used, is to select that channel for display in the master section’s Central Routing Panel (Figure 57). In both cases, this is not very convenient when recalling the console according to previous settings, for No-Input Mixer music applications. And just as with the XL 9000 K, this does not indicate either which output was used for the exit point of the feedback loop: Direct Out, Aux Send, Bus Out, etc.

Fig. 56: SSL Duality SE Channel TFT Screen
It therefore became apparent that I would need to perhaps keep a separate tally of input and output settings. During the signal flow and timbre investigation of the research, I have not taken note of which input was used, as this was implied by all of the accompanying diagrams and documentation for each timbre, in the diary and 4.3. After my very first composition session (on the Duality SE, as by that time, I had switched to that console for the remainder of the project), it was clear I needed to somehow keep a trace of input and output settings. This is discussed in detail in 5.2.1.

4.5 Using the console’s automation system as a sequencer

In the summer of 1985, I landed my first internship in a professional music production studio (Sofreson, in Paris’ 18th Arrondissement). Analogue console automation had already been available in the industry for more than a decade; sadly, the D&R 4000 console at Sofreson back then had no such feature (D&R Electronica n.d.). As a budding audio engineer trying to take my mixes on analogue consoles to the next level of polishing – once EQs and dynamics processors had ironed out the bulk of the “audio wrinkles” – I had a longing for a system that could move the faders, turn the pots and activate the switches for me. That “system”, when available, was simply to have the artists themselves lend a hand in the mixing process (Inglis 2011). With the tape machine’s time counter positioned so that everyone could see it, each artist
having been handed their own set of written instructions (actions to perform at such
and such time on the counter), mixing became an organised choreography.

Most often, however, I would be mixing on my own and thus found it difficult to
concentrate on the creative aspect itself, while also focusing on reaching for that next
pot in time to turn it, and whether or not I had turned it to the correct value. One
method I developed to increase my efficiency, was to use little bits of paper on which
I would draw arrows if need be, pointing at the controls than need to be adjusted.
This made reaching for a particular control quicker, as I would not have to first
identify on which channel the control to be adjusted resides. Further information on
this technique, applied to this research’s exemplar piece performance, can be found
in 5.3.

**4.5.1 Brief history of analogue console automation**

The first console automation system was presented in October 1971 by Wayne
Jones, president of Olive Electro Dynamics. Their Olive Series 2000 console sported
the revolutionary Automated Remix Programmer which used two tracks of the
multitrack tape machine to store fader movement and mute information (with
additional features optional). As the engineer would make the first mix pass, control
voltages from each fader would be sent to the automation programmer where they
would be converted to digital, that information multiplexed and channel coded into a
single analogue signal, recorded on one track of the multi-track tape. This signal
would then be played back, decoded back to a digital data stream, de-multiplexed
and converted back to send each VCA its own control voltage. Any updates
performed by the engineer in the next pass, would be added to the data being played
back and multiplexed into a new data stream, to be recorded onto a second track on
the multi-track tape (Jones 1971). The two tracks used by the system could not be
adjacent, as otherwise, a magnetic feedback loop would occur between the head gap
on the Sync head playing back the pulses, and the head gap recording the new
ones*. They also could not be edge tracks as, unlike timecode which can free-wheel
over a few drop outs or edge damage, the automation data was quite fragile despite
its error correction system (ibid.). Other console manufacturers such as Neve,
Harrison, API, along with now-defunct MCI, quickly followed with their own systems,

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*I have personally experienced this type of feedback on high-coercivity +9 Tapes (such as the 3M 996 and BASF
911) even though I had been careful leaving one track acting as an extra guard band in between the two.
all based on data being recorded to audio tracks. Neve’s NECAM system is possibly the most well-known of the tape-based systems of that era along with George Massenberg’s GML (Martinson 1999). Another issue reported by engineers at the time, using those systems, was how the information would develop lag, pass after pass. This was due to the time it took to decode, error-correct, process and re-encode the data. Therefore, common practice was to encode fader level changes first, and mutes or any other change where accurate timing was critical, last (Burgess 2014, p. 100; Price 1994).

As computer RAM and storage increased in capacity, the data could now be stored in the computer itself, rather than tape tracks. The computer would synchronise itself to your tape machines by means of a timecode signal recorded onto one of their tracks; convention quickly became to use the last track of your n-track analogue recorder. One would have to take precautions when striping the tape with the timecode signal (one of which was being careful what signal would end up right next to it on track n-1) but gone were the issues of the automation data being degraded with every new mix pass. This principle, of driving the automation system via a timecode signal on the audio recorder, is still in use today in many facilities sporting a large-format analogue console with either VCA or Motorised faders. Motorised faders have an advantage over VCA-based automation systems, in that the user does not need to find the “null point” before placing their fingers on a fader to write new information. Also, visual feedback is improved as the fader position always indicates the true level of gain being applied to the signal. On VCA-based systems, either a video monitor screen could be used to display the gain in the form of columns, or the meters on the console could be told to switch to that view (Solid State Logic 1987).

Alongside improvements in traditional automation systems (i.e. proprietary computer / timecode-driven) by companies such as SSL (Ultimation), Neve (Flying Faders), Harrison (GML automation), the development of MIDI-equipped consoles has opened another avenue for console automation (Duncan 2004; GML 2003; Martinson 1999). Ever since manufacturers have added MIDI ports to their audio mixers, it has been

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* Although a “relative” mode (such as SSL’s “Trim” mode) was implemented on later VCA automations, which circumvents the issue, by simply tracking the offset applied to the fader rather than its absolute position (Dennis 2001).
technically possible to control at least some of the console’s parameters with an external MIDI sequencer or DAW. Mackie’s commercialisation of the HUI protocol in 1998 has helped standardise the messages involved in controlling fader movement and standard switches, such as channel mutes (Wherry 2003). The concept was ported to smaller-format analogue consoles by manufacturers such as SSL (with the AWS 900) or API (with the 1608), which dawned the current “hybrid mixing system” generation of analogue console automation (Allen 2017; API n.d.; Robjohns 2005).

4.5.2 The SSL XL 9000 K automation system
This automation system runs on a proprietary computer, driven by timecode. The elements that can be controlled are the Large Fader (motorised) and its Cut switch, the Small Fader (VCA only) and its Cut switch, as well as the following switches which can be toggled: EQ On/Off, Insert In or not, and all Aux Sends On/Off (Cue St and FX 1-6). Information for this sub-chapter was compiled from the following sessions (see Appendix A for full details):

Session 03b – Associated video file: 03b Skweeel NIME Proposal
https://www.youtube.com/watch?v=NspETYo2UnY
Duration: 6min 11sec

Session 08 – Associated video file: 08 Skweeel Episode 8.mp4
https://www.youtube.com/watch?v=2E2hSLqKBro
Duration: 40min 35sec

Just as I had trouble finding the potentiometer storage resolution in the available documentation, the fader movement’s resolution (or more specifically, the VCA’s resolution lying beneath the motorised fader) is not clearly indicated. Page 32 of the Computer Service Manual mentions that a ‘Val’ calibration parameter “sets all VCAs to the dB value indicated. The dB value can be preset between +20dB and -100dB” (See Appendix F – SSL XL 9000 K Manuals / SSL XL 9000 K - Computer Service Manual). This gives us a range for the VCA of 120 dB. Likewise, page 34 lists the Large Fader as ‘object number’ 184, and the Small fader as number 181; page 35 indicates that all ‘objects’ are stored with a value between 0 and 1023. In other words, storage resolution and therefore automation resolution, for the Small and Large Fader VCAs, is 10-bit. The Level Trim function of the automation system, shown in Figure 58, offers offsets of 0.1 dB increments to whichever automation has
been applied to the faders – See Appendix F – SSL XL 9000 K Manuals / Computer / Section 05 Mix System, p.60).

If one divides the 120 dB range of the VCA by 0.1 dB steps, this means that 1,200 steps are stored across 1024 values. I can only assume, for lack of any information in that regard, that the 176-value difference is compensated for by a reduced resolution in the very low levels of the VCA (i.e., close to the -100 dB lower limit), to preserve full 0.1 dB resolution around Unity Gain.

My first (and last) opportunity to test the automation system was to produce a video demonstration for the 2016 NIME held in Brisbane – see Episode 03b. I was quickly reminded of the major obstacle I was going to potentially have to face for the next two years: the fact that such a console automation was never designed for intensive off-line programming. These automations are primarily designed for an engineer to move a few faders along with the music, perhaps cutting a channel or two along the way. Then, the automation interface allows the engineer to scrutinise the fader movements and cut positions, to eventually, refine their position; but emphasis is on re-doing the movements as another “take”; not so much on the editing of said moves. And definitely not, programming moves and switches from scratch, on a very small time scale, such as events spaced apart by just a few frames. This is corroborated by SSL’s statement that the Trim Mix function “is for power users only - read on if you dare!” That section ends on a single word: “Phew!” (Appendix F – SSL XL 9000 K Manuals / Computer / Section 05 Mix System, p.65).

Although first commercialised in 2013 (and therefore designed just a year or two prior to that), the automation’s visual feedback is far from providing the clarity of parameter automation visualisation that can be found on the most basic DAWs of the same time.
period, such as Pro Tools, Cubase or Logic. Having worked with Neve’s Flying Faders automation and SSL’s SL 4000 G VCA-based automation and later on, its Ultimation automation, it is fair to say that very little improvement has been made into the displaying of fader positions vs. time, with this last iteration of SSL’s automation system, found on the XL 9000 K. Comparatively, having also worked on a Lafont Producer console, programming mutes and other switch manipulations on its Optifile automation which ran on an Atari Mega ST 2/4 in the early ‘90s, was quick, easy to use, and caused little eye fatigue.

It took about thirty minutes to program the six-second sequence which can be heard at 04:02 in Episode 03b. Even a mere zooming-in operation requires clicking in very specific zones in the graphical representation, otherwise, you end up either zooming-in on a section that is in no way what you are interested in, or you view things at a zoom value which is not what you had hoped for. Furthermore, programming a switch to activate or deactivate at a specific time, requires mouse movements either side of an imaginary line which is easily missed. Last but not least, this is all done on a computer screen which was already not the best you could hope for back in 2013, in terms of size and resolution. One aspect of the research I had not envisioned was the potential health hazard presented by having to operate such a system: I did not know at that stage how I was going to preserve my eyesight from having to work in such a way for hours on end, each session, until the end of my candidature in 2018.

The rest of the session involved pre-tuning three channels to different notes, in order to perform a very simple melody. This was done relatively quickly. As demonstrated earlier in the video (02:24, Episode 03b), and stated several times in previous videos, I was yet to find a way to achieve precise-enough control of pitch to be able to use a single channel, to perform various notes.

In terms of time-resolution, when I first thought about this project many years ago, and how the automation computer of a system such as Martinsound’s Flying Faders or SSL’s G-series automation could be used as a sequencer, I had already thought of the issue of programming very rapid rhythmical events. In a timecode-based

* As Head of the Audio Dept. at ESRA Paris (1990-92), I designed and implemented a multi-track studio based on a 2” 24-track Otari MX80 and Lafont Producer console equipped with AD Système’s Optifile automation. I then spent most of my time in that studio, teaching students how to produce music with it.
automation system, the minimum interval which separates two events would most likely be a timecode frame. If one decides to work with an EBU frame rate of 25 frames per second, then each event has to align to a “time grid” that has 25 subdivisions per second. This is insufficient for the audio industry, as 1,000 milliseconds divided by 25 implies events can only occur every 40 milliseconds and nowhere in between.

Back in the late ‘90s, I produced (as well as engineered and mixed) several EPs and singles, for The SemiToneS and Watcha amongst others*, on SAE Paris’ Neve VR console; the Neve VR series was fitted with Martinsound’s Flying Faders back then (Martinson 1999). I clearly remember the off-line resolution for repositioning Mute On/Off commands to be a quarter of a timecode frame. This proved invaluable for muting/unmuting heavily-distorted guitar chords being palm-muted, to prevent compression artefacts and various string noises in between chords. It also allowed precision un-muting of tom tom channels, where expander gates failed to properly do the job for the entire track. This quarter-frame precision was achieved by adding a 0, 1, 2 or 3 behind the HH:MM:SS:FF address of the events in the cue lists.

In 1998, I produced (as well as engineered and mixed) Watcha’s first eponymous album (Watcha 1998). It was mixed on Studio Marcadet’s SSL SL 4000 G console in Studio B (Angus 1988). I used a 25 fps timecode to drive the automation from the 2” 24-track analogue tape. I was far less familiar with SSL’s automation system (VCA-based at the time) than I was with Neve’s Flying Faders. Luckily however, I had tracked the instruments in such a way that I did not have to program mutes that had to be that critically-precise. I do not recall, therefore, whether or not the VCA automation on these consoles offered the same quarter-frame mute/unmute precision. Nor do I know if the Ultimation system which came out in 1991, sported such a feature (Duncan 2004). I have been unable to find this information in the available literature.

I had all of these aspects in mind when I thought of the project, and knew I would most likely have to pick a tempo and frame rate that would work well together –

quarter-frame mute accuracy being available or not. I therefore set out to calculate the various tempo possibilities, which work with an SMPTE frame rate (both 30 fps and 29.97 fps), an EBU frame rate of 25 or a Film frame rate of 24. This has led to the creation of four spread sheets, in Figures 59 through 62. The corresponding Excel spread sheets can be found in Appendix K. For example, if a composer decided to have 32\textsuperscript{nd} notes as their smallest possible resolution running with an EBU timecode format, then the quarter note would last 8 times longer (320 ms), meaning the tempo would be 187.50 BPM (see Figure 60).

<table>
<thead>
<tr>
<th>Note value</th>
<th>If note value = 1 TC frame (ms)</th>
<th>1/4 note duration (ms)</th>
<th>Tempo (BPM)</th>
</tr>
</thead>
<tbody>
<tr>
<td>64th triplet</td>
<td>41.67</td>
<td>1000.00</td>
<td>60.00</td>
</tr>
<tr>
<td>64th</td>
<td>41.67</td>
<td>666.67</td>
<td>90.00</td>
</tr>
<tr>
<td>64th dotted</td>
<td>41.67</td>
<td>444.44</td>
<td>135.00</td>
</tr>
<tr>
<td>32nd triplet</td>
<td>41.67</td>
<td>500.00</td>
<td>120.00</td>
</tr>
<tr>
<td>32nd</td>
<td>41.67</td>
<td>333.33</td>
<td>180.00</td>
</tr>
<tr>
<td>32nd dotted</td>
<td>41.67</td>
<td>222.22</td>
<td>270.00</td>
</tr>
<tr>
<td>16th triplet</td>
<td>41.67</td>
<td>250.00</td>
<td>240.00</td>
</tr>
<tr>
<td>16th</td>
<td>41.67</td>
<td>166.67</td>
<td>360.00</td>
</tr>
<tr>
<td>16th dotted</td>
<td>41.67</td>
<td>111.11</td>
<td>540.01</td>
</tr>
<tr>
<td>8th triplet</td>
<td>41.67</td>
<td>125.00</td>
<td>480.00</td>
</tr>
<tr>
<td>8th</td>
<td>41.67</td>
<td>83.33</td>
<td>720.00</td>
</tr>
<tr>
<td>8th dotted</td>
<td>41.67</td>
<td>55.56</td>
<td>1080.01</td>
</tr>
<tr>
<td>Quarter triplet</td>
<td>41.67</td>
<td>62.50</td>
<td>960.00</td>
</tr>
<tr>
<td>Quarter</td>
<td>41.67</td>
<td>41.67</td>
<td>1440.00</td>
</tr>
<tr>
<td>Quarter dotted</td>
<td>41.67</td>
<td>27.78</td>
<td>2160.02</td>
</tr>
</tbody>
</table>

Fig. 59: Timecode to BPM mapping table for a Film frame rate of 24 fps
### Fig. 60: Timecode to BPM mapping table for an EBU frame rate of 25 fps

<table>
<thead>
<tr>
<th>Note value</th>
<th>If note value = 1 TC frame (ms)</th>
<th>1/4 note duration (ms)</th>
<th>Tempo (BPM)</th>
</tr>
</thead>
<tbody>
<tr>
<td>64th triplet</td>
<td>40</td>
<td>960.00</td>
<td>62.50</td>
</tr>
<tr>
<td>64th</td>
<td>40</td>
<td>640.00</td>
<td>93.75</td>
</tr>
<tr>
<td>64th dotted</td>
<td>40</td>
<td>426.67</td>
<td>140.63</td>
</tr>
<tr>
<td>32nd triplet</td>
<td>40</td>
<td>480.00</td>
<td>125.00</td>
</tr>
<tr>
<td>32nd</td>
<td>40</td>
<td>320.00</td>
<td>187.50</td>
</tr>
<tr>
<td>32nd dotted</td>
<td>40</td>
<td>213.33</td>
<td>281.25</td>
</tr>
<tr>
<td>16th triplet</td>
<td>40</td>
<td>240.00</td>
<td>250.00</td>
</tr>
<tr>
<td>16th</td>
<td>40</td>
<td>160.00</td>
<td>375.00</td>
</tr>
<tr>
<td>16th dotted</td>
<td>40</td>
<td>106.67</td>
<td>562.51</td>
</tr>
<tr>
<td>8th triplet</td>
<td>40</td>
<td>120.00</td>
<td>500.00</td>
</tr>
<tr>
<td>8th</td>
<td>40</td>
<td>80.00</td>
<td>750.00</td>
</tr>
<tr>
<td>8th dotted</td>
<td>40</td>
<td>53.33</td>
<td>1125.01</td>
</tr>
<tr>
<td>Quarter triplet</td>
<td>40</td>
<td>60.00</td>
<td>1000.00</td>
</tr>
<tr>
<td>Quarter</td>
<td>40</td>
<td>40.00</td>
<td>1500.00</td>
</tr>
<tr>
<td>Quarter dotted</td>
<td>40</td>
<td>26.67</td>
<td>2250.02</td>
</tr>
</tbody>
</table>

### Fig. 61: Timecode to BPM mapping table for an SMPTE frame rate of 29.97 fps

<table>
<thead>
<tr>
<th>Note value</th>
<th>If note value = 1 TC frame (ms)</th>
<th>1/4 note duration (ms)</th>
<th>Tempo (BPM)</th>
</tr>
</thead>
<tbody>
<tr>
<td>64th triplet</td>
<td>33.37</td>
<td>800.80</td>
<td>74.93</td>
</tr>
<tr>
<td>64th</td>
<td>33.37</td>
<td>533.87</td>
<td>112.39</td>
</tr>
<tr>
<td>64th dotted</td>
<td>33.37</td>
<td>355.91</td>
<td>168.58</td>
</tr>
<tr>
<td>32nd triplet</td>
<td>33.37</td>
<td>400.40</td>
<td>149.85</td>
</tr>
<tr>
<td>32nd</td>
<td>33.37</td>
<td>266.93</td>
<td>224.78</td>
</tr>
<tr>
<td>32nd dotted</td>
<td>33.37</td>
<td>177.96</td>
<td>337.16</td>
</tr>
<tr>
<td>16th triplet</td>
<td>33.37</td>
<td>200.20</td>
<td>299.70</td>
</tr>
<tr>
<td>16th</td>
<td>33.37</td>
<td>133.47</td>
<td>449.55</td>
</tr>
<tr>
<td>16th dotted</td>
<td>33.37</td>
<td>88.98</td>
<td>674.33</td>
</tr>
<tr>
<td>8th triplet</td>
<td>33.37</td>
<td>100.10</td>
<td>599.40</td>
</tr>
<tr>
<td>8th</td>
<td>33.37</td>
<td>66.73</td>
<td>899.10</td>
</tr>
<tr>
<td>8th dotted</td>
<td>33.37</td>
<td>44.49</td>
<td>1348.66</td>
</tr>
<tr>
<td>Quarter triplet</td>
<td>33.37</td>
<td>50.05</td>
<td>1198.80</td>
</tr>
<tr>
<td>Quarter</td>
<td><strong>33.37</strong></td>
<td><strong>33.37</strong></td>
<td><strong>1798.20</strong></td>
</tr>
<tr>
<td>Quarter dotted</td>
<td>33.37</td>
<td>22.24</td>
<td>2697.33</td>
</tr>
</tbody>
</table>
I have highlighted in green, the tempi which would be useable considering most DAWs or hardware sequencers won’t offer tempi over 300 BPM. For DAWs such as Pro Tools, which offer higher values, the table extends to BPMs well over 1,000.

### 4.5.3 The SSL Duality SE automation system

This is a newer-generation automation system, with motorised faders, driven by the \( \delta \)-Ctrl plug-in (or Delta-Control), compatible with most DAWs (see Figure 63). The elements that can be controlled are the Large Fader (motorised) and its Cut switch, as well as the following switches which can be toggled: EQ On/Off, Insert In or not, and all Aux Sends On/Off (Cue St and FX 1-4). Information for this sub-chapter was compiled from the following sessions (see Appendix A for full details):

Session 08 – Associated video file: 08 Skweeel Episode 8.mp4
[https://www.youtube.com/watch?v=2E2hSLqKBro](https://www.youtube.com/watch?v=2E2hSLqKBro)
Duration: 40min 35sec

Session 09 – Associated video file: 09 Skweeel Episode 9.mp4
[https://www.youtube.com/watch?v=Tybk_LPYZPA](https://www.youtube.com/watch?v=Tybk_LPYZPA)
Duration: 13min 03sec
Again, the operator’s manual is vague, regarding the Channels’ fader resolution: “a high resolution, digitally controlled gain element in each channel allows the motorised faders to be switched to control any DAW that supports the HUI or Mackie Control protocols, while retaining control of the analogue signal path” (Solid State Logic n.d.-b). One can find the resolution for the three VCAs which control the stereo Mix Busses A, B and C: three 12-bit Multiplying Digital/Analog Converters (MDACs) yielding 36-bit resolution (Rudolph 2007). Data for the resolution of Channel faders, however, could not be found in the available literature. Whilst programming the arpeggio section in the project’s exemplar piece (see I Pressed Play video at 22:40), I noticed I could adjust the Fader’s automation curve by values of $1/100^{th}$ of a dB, which turned out to be necessary in my application of fader automation, for No-Input
Mixer music. Assuming the VCAs cover the same -100 to +20 dB range as the ones on the XL 9000 K, and with a resolution of 1/100th of a dB, this requires storing 12,000 different values. This would require at least 14-bit resolution ($2^{14} = 16,384$ possible values). Interestingly, I noticed that whenever the physical fader had reached its top-most physical position, by using the $\partial$-Ctrl plug-in, the VCA’s level could still be increased by several tenths of a dB.

The time-resolution for programming events is determined by the DAW’s own precision – theoretically, as little as one sample. However, I assumed that the actual resolution for, say, Muting / un-Muting a Channel, would most likely be limited by factors such as the slew rate implemented in the Mute function by SSL to avoid clicks in the audio signal. Additionally, since the $\partial$-Ctrl plug-in communicates its automation changes to the console via the HUI protocol, itself running through the ipMIDI protocol to relay information to the console’s computer, I was expecting some latency from the entire communications process.

When using MIDI over the original 5-pin DIN connectors, the UART on the transmitting side has to serialise the data, and its 31.25 kbauds transfer rate (MIDI Manufacturers Association 2014), results in one MIDI Note On message consisting of three bytes taking about 1ms before another note can be sent. This leads to the realisation that in traditional 5-pin DIN MIDI, there are no simultaneous note-on instances; just rapid successions of notes spread roughly 1ms apart (not considering the Running Status data reduction system). I am not familiar with ipMIDI’s inherent latency (and felt this was too far off-topic for the research) but deemed that, just as with MIDI over USB, where the transmission rate is so much greater than conventional 5-Pin DIN MIDI, even busy downbeats do not see their individual elements perceptibly smeared, the transfer rate of MIDI data over a modern Ethernet connection would be so high as to not generate perceptible lag. This needed to be confirmed.

An initial test to see if 48 instances of the $\partial$-Ctrl plug-in in Pro Tools would slow the system down, indicated that was not the case: the system was barely affected, as seen at 07:33 in episode 08. I then programmed a fader movement with a very rapid, cyclic change (using Pro Tools’ pencil to draw in a square Fader automation in grid
mode, with a ¼ note resolution). With the camera set to the side, I would be able to see if all faders reached the top of their course at the same time or not. If there was lag in the transmission and/or interpretation, I was expecting to see a wave develop, or perhaps, a random behaviour (if the lag isn’t spread sequentially). Both the video at 08:05 (Episode 08) and my own eye set at the same level, to insure that the video camera’s shutter wouldn’t affect the synchronicity, confirmed that everything was moving in perfect sync.

At 09:01 in Episode 08, I demonstrate the resolution that is achievable when using a tempo-based grid, which – on Pro Tools – is 1/64th. I also demonstrate another major advantage of working to tempo, as shown at 09:15, which is to be able to alter tempo after having already programmed events to follow the tempo grid. If the tracks containing the automation information are properly set to Tick-based instead of Sample-based, then the events will remain in position relative to their bar, beat and sub-division. This led me to make a mental note that I would include a tempo change in my composition, to demonstrate this possibility. Working to timecode is also possible, as well as any other time grid offered by your DAW. In Pro Tools, sub-frame timecode resolution is available, beyond quarter frame.

Using a quick No-Input Mixer re-interpretation programming of Orchestral Manoeuvres in the Dark’s Enola Gay (1980) with its very recognisable introduction rhythmical pattern, Episode 08 also shows, at 30:23, how programming melodies with Larsen effects works: one needs to automate a fader included in the feedback loop, to alter pitch. This leads to the realisation that unless the Mute switch is also automated, the result is that the notes are legato and change with a glissando (or portamento), as heard at 32:05. As long as the note values are long and the tempo is relatively slow, it is possible to Mute and un-Mute in between notes, also giving the automation time to position the fader for the new note’s pitch. If a quick tempo, and/or short time values are required, then the solution is to use two (at least) different channels, and to alternate between the two. This is shown from 33:02 onwards.

Early on during the filming of Episode 08, timing issues became apparent, in patterns that were not deemed complex in regard to the time-resolution expected from the
system. This led to further investigation to try and find if this was inevitable, or could be solved. I came up with the following three hypotheses:

Hypothesis #1: *the high voltages resulting from the channels being overdriven by Larsen effects, are interfering with the various control voltages (interpreted from the data sent by the d-Ctrl automation plug-in).* To test hypothesis #1, I used the same Pro Tools session from Episode 08 to control the various Mute/un-Mute commands in my *Enola Gay* re-interpretation, only this time, using tones from signal generator plug-ins to replace the Larsen effects. This is shown in Episode 09 at 04:00. I used very conservative levels to rule out any possible interference between audio and control signals. Yet, as seen in Figure 64, the 8th note audio pattern is not being unmuted and muted in sync with Pro Tools’ click track, at a conservative tempo of 143.1 BPM. This rules out the hypothesis that the Larsen effects’ audio signals are interfering with the automation control signals.

![Console Audio](image)

![Pro Tools Click Track](image)

**Fig. 64: Screenshot of click track vs. audio output from console**

Hypothesis #2: *Pro Tools has communications issues when using the HUI protocol.* The way to test this would be to use any other DAW with exactly the same programming, such as Apple Logic Pro X (Apple 2017) or Ableton Live 9 (Ableton 2017). Apple’s Logic was quickly dismissed, as the Delta-Control AU plug-in does not
work with Logic as per SSL’s website, under the heading “Logic Studio Not Supported”:

There is an inherent limitation in the architecture of Apple’s Logic Studio software. Logic prevents any controller from controlling more than one plug-in at a time. Viewed purely within an ‘inside-the-box’ paradigm this makes some sense in that the control interface is the mouse or trackpad but it presents significant limitations to the way hardware controllers can be used with the software. It is with much regret that the single Fader δ-Control plug-in does not contain an Audio Units version and thus does not work with Logic. The multi-channel δ-Control plug-ins available for AWS, Matrix and Sigma are not compatible with Duality Delta consoles (Solid State Logic 2018).

This was investigated in Episode 10, devoted entirely to the matter. The same simple 16th note pattern at 120 BPM was programmed in Avid Pro Tools 10 and Harrison MixBus 4 (Harrison 2017). Both showed time-keeping issues. Statistics were compiled from several recordings of each software sequencing the mutes on the console. The outcome is that timing issues are apparent in both systems, ruling out the hypothesis that Pro Tools is at fault. For more detail and screenshots, see Appendix A – Session 10.

Harrison MixBus 4 Maximum offset from grid = 47 ms
Average offset from grid = 20 ms

Avid Pro Tools 10 Maximum offset from grid = 26 ms
Average offset from grid = 12 ms

Hypothesis #3: the HUI protocol’s timing being somehow linked to timecode frames, one has to optimise tempo so that automation events coincide with Timecode frame durations. If the Mackie HUI protocol indeed relies on timecode frames to organise its data packets, then choosing a tempo which conveniently places bars, beats and sub-beats on the edge of timecode frames, should show an improvement in overall timing, compared to any other tempi. This was investigated in Episode 09 at 05:35 and the results ruled out this reason for timing issues. As I was using an EBU frame rate, which is 25 fps, the corresponding tempi which align to that frame rate – as seen in Figure 60 on page 116– are 187.5 BPM or 93.75 BPM. The audio coming out of the console has timing issues and events missing at the faster tempo, and still has timing issues at the slower one. This therefore rules out hypothesis #3, that the
tempo should ideally align events to timecode frame positions, to enhance HUI transmission time-keeping.

This investigation into time-keeping issues led to the acknowledgment that this would not be something I could solve, and that I would have two options for composing the piece of music. I could either renege on benchmark points 4 and 5 (generating rapid sequences and precise rhythmic patterns), and compose a piece that would feature slower, more spaced-out and less rhythmical events, or I could use expander/gates in the console’s Channels, triggered by signals generated by Pro Tools.

Having conducted extensive timbre-generating research on the XL 9000 K and Duality SE, and investigated automation techniques and the limitations of both timecode and DAW-based systems, the focus of the research then turned to developing the exemplar piece.
Chapter Five: Composing and performing exemplar piece

Associated video file: Skweeel - I Pressed Play.mp4
https://www.youtube.com/watch?v=SbSbqhlyG-M
Duration: 30min 04sec
The musical piece itself starts at 16:04, ending at 26:38
Associated Pro Tools file in Appendix J

The audio from the *I Pressed Play* performance is presented in two forms in the Pro Tools session’s Bounces folder: *Compressed* and *Full Dynamic Range*. The *Compressed* version was produced as a post-process using multi-band compression, to be suitable for monitoring on an average system or ear buds. To experience the piece the way the audience members did on the studio’s main monitors, use the *Full Dynamic Range* version. Warning: this version will challenge even the most advanced audio systems, with low frequencies extending near DC.

In the Pro Tools *I Pressed Play* session, the teal-coloured track was used to record the desk output since Session 4. The Pro Tools session has been saved with all the playlists unfurled in chronological order. The *Compressed* version is in the soloed active playlist seen in Figure 65.

Fig. 65: Pro Tools session desk output track with its playlists unfurled
As explained in the research methodology, I could have demonstrated the feasibility of each of my project’s benchmarks, with a series of short studies. Instead I opted to present the research output in one single larger-scale work. I believe it is important to reiterate here, that the research focuses on the validity of the tool used to compose the piece of music – not the quality of the composition in itself. In the following sub-chapters, aspects of the composition which are not of prime importance, or do not introduce a new concept or knowledge, have been left out. For a full itemisation of the compositional procedure, please refer to Appendix A – Composition.

Considering a suitable title for the composition, I have always had an inclination towards song titles that trigger reflection, via double entendre for instance: AC/DC’s Big Balls or Stiff Upper Lip come to mind (1976, 2000). This is how I realised that I could name the piece I Pressed Play. This refers to a common joke in the dance music industry regarding EDM DJs: David Guetta and Skrillex being the primary targets due to their international fame (Dawson 2017), several memes can be found when searching the web using “I pressed play” as the search string. A T-shirt design is also available with that tagline (Spreadshirt.co.uk n.d.). The joke highlights the fact that some EDM DJs who compose a piece or entire set at home, then have nothing to do for the entire two hours of their live show, once they have hit Play on their sequencer/playback device. Indeed, I too would be in that situation of having pre-programmed as much of the performance as possible in the automation computer, to ideally just have to press Play on the console’s transport section, for the whole piece to be performed as intended. Like Conlon Nancarrow, I would be removing the performer from the performance (Slought Foundation.org 2004). However, I already envisioned some aspects that would require either manual tweaking, or actions that cannot be performed by the automation computer, so I would not just be pressing Play.

The composition had to meet the various benchmarks outlined in the research methodology:

1. Include several different types of timbres with a variety of nuances
2. Demonstrate a high accuracy over pitch control and other parameters such as modulation via various triggers
3. Incorporate sections where a large number of sounds are generated and controlled simultaneously – far more than ten, which would be impossible to achieve by hand by a single artist
4. Incorporate sections where sounds are generated in rapid sequence, which would be impossible to achieve by hand
5. Incorporate sections where the rhythmic precision of particular patterns would be impossible to achieve by hand
6. Incorporate an element of surprise (if tonal control is achievable) by composing a piece which starts off as noise, but progressively tends towards melodic and harmonious content.

5.1 Composing

Having established that pitch control was achievable on the SSL Duality SE, all six benchmark items could be demonstrated, including points 2 and 6 which otherwise would have had to be left out. Point 6 therefore instructed how the piece would be structured: it would start with noise timbres and gradually develop into tonal components. To keep an element of surprise, I decided early on that the tonal components would emerge from the noise components in such a gradual way that one would not notice any particular point where the transition occurs. This would be in the form of several sustained notes slowly evolving to resolve in a chord. To reinforce the element of surprise, I also decided early on that I would introduce a sudden change into a predominantly-tonal arrangement, once the chord has resolved.

Before I set out to produce the piece with the console in front of me, I had a few compositional ideas derived from the benchmarks. Inspiration for the introduction based on noise timbres with emerging chord, came from Pink Floyd’s intro to Cluster One, from the Division Bell album (1994). I also had in mind the insect-like noises from David Lee Myers’ Ourobouros album, especially in Entomo 1 and Calyx (2001). To link the final composition with the research project’s posted video episodes, the first sound to come out of the speakers would be the “sonar ping” which introduces the episodes via the videos’ trailer – see 00b Skweeel Trailer. I also wanted to incorporate elements of Glitch (Cascone 2000; Church 2017) which would demonstrate benchmark point 4. Inspiration for this came from both Alec Empire’s
Squeeze the Trigger (1997) from 0:10 onwards on track 1 of the eponymous album, and Animals as Leaders’ The Price of Everything and the Value of Nothing at 3:05, on their debut album, Animals as Leaders (2009). The term for these effects, where a short section of sound is repeated, is known as “ratcheting”. It was originally introduced to denote the use of rapid repeats of a specific note inside a step sequencer, as heard in early Berlin School electronic music such as Klaus Schulze in Tangerine Dream (Meyer, C 2016). The line between ratcheting and glitch can become blurry, as when the ratcheting rate is pushed up to just repeating one sample, the result is no longer perceived as a rhythmical pattern. I knew I would be limited by how quickly the Mute On/Off circuitry could react, and following the timing issues discovered in Episodes 08 through 10, that I would have to program these using triggered expander/gates instead. I would therefore be limited by the shortest attack and release times of the expander/gates, in terms of the shortest possible interval for a ratchet effect. Last but not least, I knew the piece would have some rhythmical elements, which would easily address benchmark point 5. Benchmark points 1 and 3 would inevitably be achieved, once the composition is fully developed, as the number of timbres and the amount of simultaneous sounds would exceed ten – or at the very least, the complexity of the arrangement would not be achievable by a single pair of human hands.

Important note: expander/gates triggered by sine waves generated by Signal Generator plug-ins in Pro Tools, were used to circumvent the timing issues identified in Episodes 08 through 10. I do not consider this method to be in infringement of the No-Input Mixer music paradigm, as the sine waves generated from Pro tools are never heard: they are just control signals for some of the console’s expander/gates, just as the $\partial$-Ctrl plug-in controls certain switches and the fader. All other events whose timing is not critical, were handled by un-Muting / Muting the channels via their respective $\partial$-Ctrl plug-in.

Having had no training in composition, other than having produced and engineered singles and albums for a variety of genres over the past three decades, my approach to the construction of the rest of the piece was driven by the sequential nature of the various night sessions I was able to book at the studio. This has resulted in a linear collage of various sections, for the most part nearly finished before I would move on
to the next. However, although the bulk of my compositional method was sequential, working section per section, I then came back to add elements to some sections or modified the duration of some sections or layered some elements (such as a Shaker timbre) over multiple sections, during the latter production sessions. Nine sessions totalling seventy-seven hours were required to produce the piece. During the one and only rehearsal of the piece prior to its public performances, some new elements were added. There was only one rehearsal because I was quite confident with the various manual operations required (having had to perform them throughout each session, as I added those parts), and I had to make up for lost time to meet my proposed performance date at SAE Institute Byron Bay.

Each composition session followed the same procedure:

1. reloading previous computer session files (in no particular order) – See Appendix J
   a. in the external computer running the DAW using the relevant Pro Tools session files
   b. in the Duality SE’s on-board computer using the Duality Remote software on the external computer, to reload the files in the console’s computer in case they had been erased there in the meantime
2. recalling the channel settings
3. re-establishing all the patch bay connections with the Settings - FX and Patchbay.rtf file – see Appendix I
4. fine-tuning some of the channels’ settings with the Settings - Tuning.rtf file – see Appendix I
5. recalling outboard time-domain processing settings (delay and reverberation)

The role of the Pro Tools session files is to automate the vast majority of the various parameters of the piece. The Duality Remote files allow the console to retrieve all of the Total Recall settings for the piece (Channels and Master Section) as well as the Channel names in the electronic scribble strip. The Settings - FX and Patchbay and Settings – Tuning files, store additional information that cannot be saved in the two previous sets of files.
Establishing the patch bay connections after recalling the channel settings has two advantages. First of all, as discovered in Episode 07 (visible at 04:06), as the feedback loops are not yet established, there is no interference with the Total Recall voltages when resetting the pots and switches. Secondly, although the console has not shown any signs of fatigue during the entire research process, I felt it was safer practice to expose the circuitry to as little as possible of the Larsen effects’ high voltages. As it takes between an hour to an hour and fifteen minutes to complete the recall of all console parameters, it is a significant amount of time sparing the circuitry for the channels that have been reset early on in the process.

The first composition session was spent programming the insect and animal-like sounds inspired by my Pink Floyd and David Lee Myers influences. Channels 4, 5 and 6 on the Duality SE were set to produce the Frog Call timbre; a low-pitched one on Channel 5 and a high-pitched one on Channel 6. For details, refer to the *I Pressed Play* Pro Tools session in Appendix J. The fader is included in the feedback loop, thereby changing not only the volume of these Frog Calls, but also the rate at which the pulses are emitted. The automation lines shown in Figure 66 show Duality SE fader automation via the $\partial$-Ctrl plug-in.

![Fig. 66: Console fader automation for introduction Frog Calls and Sonar Ping timbres](image)

To add more variety to those three combined sounds, I decided to have manual control over the third Frog Call. This was done on Channel 4 by manipulating the EQ section’s HF Gain pot. To remind myself which channels require manual modifications throughout the piece, and of which parameters, I implemented the following rules:
- Pro Tools tracks that control Channels which require manual operation, are coloured brown.
- Instructions as to which parameters are to be moved are in the corresponding track’s Comments box.

Therefore, one can see for example in Figure 67 (and subsequent figures depicting the Pro Tools session) that the track controlling Channel 4, labelled 4 Frog 1, is brown as opposed to channels which are fully handled by automation, in the standard dark green colour for Pro Tools’ audio tracks. The Comment box for that track reads: “Play with HF GAIN”. I have used capitalisation to help distinguish from a distance, whether the GAIN or FREQ pot of a particular EQ band needs manipulation, and which is more important when two or more pots are to be moved.

A “Sonar Ping” was programmed using the Squeal timbre on Channel 6, with the fader automation shown on Figure 66 rapidly crossing the tipping point. In subsequent sessions, two additional “noise” sounds were programmed to alternate with the manually-controlled Frog Call timbre during the introduction, which was also lengthened a few times. In Session 02, I stumbled upon a timbre reminiscent of the Tearing Fabric timbre found on the XL 9000 K (see Episode 03, 10:28). When investigating the timbres on the Duality SE, I was not able to generate that timbre (see Episode 06, 29:49) so I was quite excited at the prospect of featuring that very peculiar sound in the introduction to the piece. However, having lost all of the data for that session in the backup process, I was never able to recreate that timbre afterwards. Discussion on why the data was lost and how to avoid that can be found in 6.4. In Session 03, a replacement timbre was programmed and initially named
“SFX”. In hindsight, the perfect name for this timbre and its rapidly-oscillating two notes, should have been “UFO”. This was later confirmed by Ian Slade from Southern Cross University (2018, Pers. Comm., 25 June), as he was helping me with the video editing of the performance. He thought it sounded like the flying cars’ sound effect found in the cartoon series The Jetsons (Hanna & Barbera 1962). Indeed, it is quite close to the sound effect heard at 0:15 in the TV show’s trailer (Cartoonsintros 2011). The pitch of the sound effect is altered by manipulating Channel 13’s LMF Frequency pot, with some manipulation required on that EQ band’s Gain pot as well, as per comments in the corresponding Pro Tools track, seen in Figure 67. Later in the composition sessions (Session 05), I stumbled upon a Gecko timbre, which I had not found in my timbre investigation on the Duality SE – I was too focused on making sure the timbres found on the XL 9000 K could be achieved, to fully investigate existing feedback loop configurations using the Compressor in the Dynamics section. The rate and pitch of the Gecko’s “kisses” is controlled by manually changing the LMF Gain on the corresponding track, assisted by some centre Frequency pot manipulations.

To introduce the tonal aspect of the composition, sustained notes were programmed to appear very gradually. These are obtained with Squeal timbres, spread over Channels 25 through 32, of which Channels 25, 27, 28 and 29 are sub-grouped to Channels 23 & 24 for level control. Figure 68 shows an overview of these channels at the beginning of the piece. Channel 26 inexplicably has refused to produce any Larsen effect since day one. This meant that the composition would only potentially feature a maximum of forty-seven sounds.
The third, fourth and fifth channels in Figure 68 (two fully-automated dark green channels and one manually-altered brown channel) produce an E4*. Instead of tuning all three notes to the exact same pitch, a slight error was purposefully applied to produce a rich chorusing effect. Their level is gradually raised via the audio subgroup made up of Channels 23 & 24 seen in light green, from Bar 42 (17:27) to plateau at Bar 80 (18:42)†. Their pitch was also programmed to slowly converge during that time, to their target E4 notes. From Bar 81, a pair of higher notes and a pair of lower notes are progressively introduced in the chord (added to the three E4s), generated by Channels 29 to 32: A5, F♯4, E4 and E3 respectively. The chord is at its nominal level with all of its constituents at Bar 103 (19:29).

At Bar 115 (19:54), Channels 1 to 3 introduce a soft-level low-frequency pad consisting in a B2, E1 and E2. At this stage, I just wanted to control their level and since their faders were included in the feedback loops to control their pitch further down the line, this is where the Duality SE’s three separate stereo Mix busses – A, B and C – were used. As seen in Figure 69, Mix B which had been contributing to controlling some of the intro levels for the Frogs, Sonar Ping, SFX and Gecko, is now free to dip down to -∞ at Bar 103 (19:29) to then slowly bring up Channels 1 to 3 which have been assigned to that Mix bus. These three Mix busses would prove

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† I am using the Scientific Pitch Notation system throughout this research (Young 1939), where C0 = 16.352 Hz and A4 = 440 Hz.

‡ Indicated Bar values correspond to the Pro Tools session. Indicated times refer to the position in the Skweeel – I Pressed Play video.
essential for taming very loud Larsen effects, whilst allowing the inherently-softer ones to compete in the mix.

Fig. 69: Console VCA automation for Mix A, B and C stereo busses

At Bar 103 (19:29), benchmark point 5 is addressed by having the expander gates on the stereo subgroup that is Channel 25 and 26, made to shut and open via the disappearance of the Signal Generator plug-in on the two corresponding tracks, being rapidly muted and unmuted. Figure 70 shows that this not only happens for Channels 25 and 26, alternating sides, but then also for Channels 32, then 30, then 31.

Fig. 70: Pro Tools Mute automation to trigger console expander/gates via Signal Generator plug-ins*

* The ramp seen on track 25 in the fifth cluster of mutes, and the ramp seen at the beginning on track 30, are Pro Tools display bugs, which randomly appear depending on horizontal zoom values.
To highlight the transition into the predominantly-tonal section of the piece – the element of surprise from benchmark point 6 – I thought I could simulate an automation system breakdown, by purposefully introducing randomness into the very regular patterns that had been occurring since Bar 103 (19:29). This was done by changing the grid size when drawing rectangular Mute automation with the pencil tool, or deleting sections at random, increasing the arhythmicity. This is first apparent at Bar 128 (20:18) and becomes more chaotic until, to enhance the effect, I also drew in random VCA level changes on the tracks controlling Mix A and B stereo busses, as shown in Figure 71. Near-complete silence is achieved shortly before Bar 134 (20:30) by Mix Bus C reaching $-\infty$ and other channels (bussed to Mix A or B) being muted via individual console Mute automation.

![Fig. 71: Pro Tools Mute automation and console Mix Bus level automation, simulating system breakdown](image)

The ensuing sudden silence is broken at Bar 134 (20.31) by Channels 9 to 11 programmed to generate a Hans Zimmer “Horn Blast”, also known as the “Inception Braaam” (Abramovitch 2015; Mills 2014). These are Sawtooth timbres with the feedback loop including the fader, set so that when the fader reaches a certain point, the signal breaks up. To only keep a minute section of the break-up, the console
Mute was automated as exact time was not critical; there was no need for resorting to triggering expander/gates. When absolute quiet is required, the triggered expander/gate solution is not ideal as the maximum gain reduction is 40 dB and therefore not sufficient to fully block out the louder signals.

To further enhance this element of surprise, I purposefully kept the timbres in the “noise” introduction from generating too much low-frequency content. A first allusion to the capabilities of the composition to generate sub-frequencies occurs between Bar 80 (18:42) and 90 (19:12): a Ring Modulation effect is achieved by turning the LMF Gain down in the EQ section on Channel 28, which is one of the three high-pitched Es in the seven-note sustained chord. As Channels 25, 27 and 28 are subgrouped to Channels 23 and 24, manipulating the EQ section on 28 has had this interesting effect, stumbled upon as I was tuning that Channel at the beginning of a new compositional session. The resulting low side-band frequency goes down to DC and is then brought back up to unison. Likewise, the low-frequency “sawtooth pad” of Channels 1 to 3 is quite soft in that section. Both of these are kept subtle so the first obvious apparent occurrence of sub-frequencies at Bar 135 (20:33), produces ample contrast. This is achieved by the “LF Drone” on Channel 12, supporting the three “Zimmer Braaams” on Channels 9 to 11. Channels 1 to 3 play 8th notes in between the four “Zimmer Braaams”, from Bar 135 (20:33) onwards. Their volume is automated to provide some inflection of the generated notes.

The percussive elements of the piece are introduced starting with a kick drum at Bar 146 (20:55). This is achieved with Track 8 controlling mutes on Channel 8, by keying its expander/gate. The sound is a heavily-equalised Squeal timbre. In the latter sessions, Channel 47 was added to support it as that sound was not coming out loud enough in the mix. Likewise, at Bar 153 (21:09), a 16th note Frog timbre derivative has been programmed via a keyed expander/gate on Channel 44 to act as a high hat (panned full left) and another Frog timbre derivative “snare drum” playing on beats 2 and 4 appears from Bar (21:13) onwards. To bring more variety to the piece, and to further evidence benchmark point 4, I wanted to add a “shaker” percussive element. The shaker, appearing at Bar 150 (21:03), is programmed with a variety of rhythmical patterns to bring diversity overall. This is the only sound that is not Larsen-effect-based in the entire composition. I tried to reach a timbre that was closer to white or
pink noise, than sawtooth or square, by running a Larsen effect through multiple gain stages – hoping to distort the signal so much that the predominance of odd and even harmonics would be washed out... with no success. I therefore chose to use the console’s built-in oscillator section, set to Pink Noise, coming into a channel, which is controlled by a keyed noise gate. I believe that this still fits within the paradigm of No-Input Mixer music, as I am not using any external signals to the console. To mimic a shaker’s accents, the EQ section was used to boost the high frequencies, and said EQ section was toggled On and Off where needed.

At Bar 159 (21:21), as the rhythm thus far was already very influenced by Frankie Goes to Hollywood’s hit single Relax (1984), I thought I would further the homage by programming chords which mimic the rhythmical chordal pattern of the original song. This was achieved by reprogramming the fader positions on Channels 25, 27 and 28, whilst compensating for relative volumes with the 23 & 24 subgroup. Indeed, as one pushes the fader up to produce a lower-pitched note, the overall volume also increases, and vice-versa for higher-pitched notes. Another example of this volume compensation was implemented for the melodic arpeggios, described further below.

To break out of this rhythmical section, I thought I could introduce another concept, of which I had not thought until quite late in the research. When establishing my benchmarks, I had the XL 9000 K in mind, and therefore would not have been able to program tempo change. This opportunity revealed itself when switching to the Duality SE and its DAW-based automation. I saw this as another way to support benchmark point 5 and to a certain extent, point 4. By having all the relevant tracks set to “tick-based” in Pro Tools, I programmed a ritardando as a convenient way to shift the focus of the piece to the major benchmark point for the research: point 2 – pitch control.

Pitch control has already been evidenced since the introduction of the sustained chord early on in the piece but I thought the composition needed to make it clear that not only chords were achievable, but complex melody as well. Channels 15 and 16 were set to produce a Squeal timbre, with the fader included in the feedback loop. These were then fed into Channels 17 and 18, which handle the volume compensation via simple console fader automation. Triggered expander/gates on
Channels 17 and 18 would control when the notes generated by Channels 15 and 16 appear. Figure 72 shows the “Lone Notes” section which starts at Bar 192 (22:31), where the audience is given time to acquire this new set of sounds, along with visualising how the associated faders move. The notes are spread over two channels to allow them to play staccato, as explained in 4.5.3. Programming a glissando is as easy as having the two notes generated by the same Channel. This is heard for the first time leading into Bar 200 (22:49), played by the Channel 16/18 tandem. You can see in Figure 72 how the sixth mute “crenel” on track 18 lasts twice as long as the other ones for the single notes. Whilst Channel 18 is un-Muted (via its keyed expander/gate), Channel 16, generating the sound, is seen changing its fader position mid-way.

The most complex section of the piece, which required many hours to program, commences at Bar 220 (23:27). This section, shown in Figure 73, was designed to showcase the overall concept of the research project: to demonstrate that No-Input Mixer music can be taken far beyond its current state, by using a large-format automated analogue console. This section addresses all of the key benchmarks for the composition: point 1 of having more than five different timbres, point 2 of accurate pitch control, point 3 of a large number of simultaneous sounds, and points 4 & 5 of rapid sequence and rhythmic precision of patterns. To a certain degree, even point 6 is addressed here, as this is the first time the audience sees the faders move that rapidly, producing this many notes.
The next point of interest in the composition, is the room I made for a solo section from Bar 268 to 300. I wanted to bring back some of the noise components to show how they too could be integrated inside an otherwise more conventional piece. It also allowed some additional randomness into the otherwise fully-replicable performance. Channels 33 to 36 generate the various timbres, whilst Channels 37 to 40 provide additional level control via their dynamics section. The first sound at Bar 268 (25:03) was an opportunity to use the Flatula timbre in the piece, alternating with Squeal timbres via manipulations of EQ section parameters (see Figure 74). This is followed by a Frog Call timbre at Bar 276 (25:19). The next sound, at Bar 284 (25:35) is one I did not find during the timbre exploration stage on the Duality SE, which I have named Flipper. It is derived from the Frog Call timbre, using the Compressor in the Dynamics section; variations are again produced with the EQ section included in the feedback loop. The fourth solo sound is also a Frog Call timbre derivative, with the EQ offsetting the pulses in the upper spectrum; variation is this time provided by modifying the Compressor’s Release Time pot from its fully clockwise position, to anti-clockwise, whilst at the same time, activating Fast Attack (depressing the pot) before the sweeping movement, to be deactivated once the sweep is finished.
For the piece’s finale, I drew inspiration from a section of John Williams’ Star Wars *Main Title/Rebel Blockade Runner* which uses ritardando and unison very effectively at 1:41 (1977a). Figure 75 shows the unison of Channels 1 – 3 in that section. During the one and only rehearsal of the piece, I still felt the finale lacked the complexity it needed to make a final statement. I therefore programmed the arpeggio channels, this time playing a rapid descending scalic 16th note pattern, inspired by Williams’ Star Wars *The Throne Room/End Titles* at 2:17 (1977b). These are then used to mimic a triangle, rapidly alternating between two notes during the final sustained chord, such as in Rossini’s *William Tell Overture* finale, where the triangle is playing 16th notes continuously over 7 bars (Malinowski 2014) slightly before, and during the final coda.
Last but not least, to close the piece with an event which combines sound and visuals, underlining the performance aspect of the No-Input Mixer music genre, I thought it might be interesting to have all of the sounds appear one last time, spaced apart by a very short amount of time, creating a visual wiper effect from left to right and back to left. This propagating wave would be initiated by a flick of the finger, and “caught” on its way back. Even though I had the idea in one of the earlier sessions, I had to wait until the latter sessions to implement it, as I needed all the channels to produce sound. I settled on a spacing of 1/64th to make the event quick enough not to garner too much importance, and remain what I intended it to be: a funny gimmick. Figure 76 shows the programming of Channel Mutes and their cumulative offset.

Fig. 76: Wave sound effect Mute automation
5.2 Scoring

When I presented a work in progress of the project at the Melbourne Postgraduate Music Research Conference 2018, I was asked whether or not there was a score for the piece, that other people could use to perform it on their own. The essence of my reply was that as I was composing and producing the piece, the various methods I put in place and their documentation were in fact the very score necessary to perform it. The literature review on scoring (see 2.14) provided guidance with some of those aspects.

Firstly and most importantly, the Pro Tools session itself, forms the vast majority of the score: it is a near-equivalent to the piano roll fed into a player piano. “Indeed, when musicians use the program to create clips and tracks, they are also producing the specific graphical interface they will use in the performance” (Butler 2014, p. 82). To impart some manual aspects to the performance, the instructions written in Pro Tools’ Comment boxes tell the performer which specific settings require manipulation, and how they are to be manipulated. A certain degree of freedom is allowed, by the deliberate use of the word “play”, as for instance in “Play with LMF FREQ and Gain” on Channel 13’s SFX timbre instructions (Figure 67 on page 130). To identify those specific Channels, the corresponding tracks in Pro Tools are brown, whereas fully-automated Channels have dark green tracks. Pro Tools Memory Locations are used to highlight the various sections, and at the same time, indicate when a particular Channel needs manipulation – see Figure 77. For instance, before reaching Bar 80 (18:42), one needs to stop manipulating the SFX on Channel 13, to prepare for manipulating Channel 28 on the other side of the console’s Master section, in time to start the Ring Modulation effect sweep.

* Butler refers to Ableton Live here, but this is true of most DAWs, such as Pro Tools.
These instructions are however not enough, on their own, to be able to perform the piece. Before the console can produce any sound, let alone the sounds expected from the piece, some setting up is required – just as John Cage’s prepared pianos or Conlon Nancarrow’s prepared player pianos: they could not just ‘walk up to the piano and open the lid’, for it to produce the required sounds. The following sub-chapters detail these processes and the two additional documents that, along with the Pro Tools session and Duality Remote files, form the score for the composition. All the information necessary for either the computer systems, or the human performers, to setup and perform the musical piece, are contained in these four sets of files.

5.2.1 Setup procedure
At the end of the very first composition and production session, as one would for normal music production work, documentation of session data which cannot be stored by the consoles’ on-board computer had to be done*. One option would have been to use the corresponding Pro Tools’ comments box for each track. One quickly realises, however, that only a dozen words can be visible at any one time, which might not be enough. Although a viable option, if one is willing to have to click in the Comment box and scroll to view the full content, there are paths that do not correspond to specific Pro Tools audio paths, that still need to be documented, such as which outboard reverberation unit is patched to which console Auxiliary Send. I therefore opted for a simple rich text file as seen in Figure 78.

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* At the end of each session, all console settings were stored using the Total Recall system on the console’s on-board computer. These files were backed up each time to my research computer. I have made the last one, Duality Remote backup - I PRESSED PLAY.zip available in Appendix J.
For the purpose of this research, as I first thought it would perhaps not be necessary otherwise, I chose to keep track of the elements I added or changed in each session with colour coding. In hindsight, the colour coding could assist in back-tracking if one was not happy with the latest results but this would also imply saving a different file at the end of each session, unlike in my case where I have overwritten whatever would have been noted for a particular channel in previous sessions.

A list of channels 1 through 48 indicate first and foremost, which input is being used (see issues with displaying input selection when the Total Recall screens are being displayed on the meter bridge TFTs): Line, Mic or Bus. Then I have indicated the exit and entry points for that channel to be connected via the patch bay. Note that on the SSL Duality SE, when one selects “Mic” in the Central Routing Panel, the corresponding access point on the patch bay* is labelled “Channel Input” (not “Variable Gain Input Amplifier”), whilst the Input selected when choosing “Line” in the Central Routing Panel is labelled “Monitor Input” on the patch bay – as seen in Figure 79.

* This is the standard patch bay supplied by SSL out of the factory floor.
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<thead>
<tr>
<th>Session 2</th>
<th>Session 3</th>
<th>Session 5</th>
<th>Session 6</th>
<th>Session 7</th>
<th>Session 8</th>
<th>Session 9</th>
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<tr>
<td>01: LINE Chan Out 1 -&gt; Mon In 1</td>
<td>Daw Out 1 -&gt; Key In 1</td>
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<td>02: LINE Chan Out 2 -&gt; Mon In 2</td>
<td>Daw Out 2 -&gt; Key In 2</td>
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<td>Daw Out 3 -&gt; Key In 3</td>
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<td>04: MIC Chan Out 4 -&gt; Chan In 4</td>
<td>Tweak HF Gain</td>
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<td>05: MIC Chan Out 5 -&gt; Chan In 5</td>
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<td>07: LINE Chan Out 7 -&gt; Mon In 7</td>
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<td>08: LINE Ins Send 8 -&gt; Mon In 8</td>
<td>Daw Out 8 -&gt; Key In 8</td>
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<td>Tweak LMF, Freq and HF Gain</td>
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<td>13: MIC Ins Send 13 -&gt; Chan In 13</td>
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<td>14: MIC Ins Send 14 -&gt; Chan In 14</td>
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<td>17: BUS</td>
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<td>18: BUS</td>
<td>Daw Out 18 -&gt; Key In 18</td>
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<tr>
<td>19: LINE Chan Out 19 -&gt; Mon In 19</td>
<td>Daw Out 19 -&gt; Key In 19</td>
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<td>20: LINE Chan Out 20 -&gt; Mon In 20</td>
<td>Daw Out 20 -&gt; Key In 20</td>
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<tr>
<td>21: LINE Chan Out 21 -&gt; Mon In 21</td>
<td>Daw Out 21 -&gt; Key In 21</td>
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<tr>
<td>22: LINE Chan Out 22 -&gt; Mon In 22</td>
<td>Daw Out 22 -&gt; Key In 22</td>
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<td>23: BUS</td>
<td>Daw Out 23 -&gt; Key In 23</td>
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<td>24: BUS</td>
<td>Daw Out 24 -&gt; Key In 24</td>
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<tr>
<td>25: LINE Chan Out 25 -&gt; Mon In 25</td>
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<td>27: LINE Chan Out 27 -&gt; Mon In 27</td>
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<td>28: LINE Chan Out 28 -&gt; Mon In 28</td>
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<tr>
<td>29: LINE Ins Send 29 -&gt; Mon In 29</td>
<td>Daw Out 29 -&gt; Key In 29</td>
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<td>30: LINE Ins Send 30 -&gt; Mon In 30</td>
<td>Daw Out 30 -&gt; Key In 30</td>
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<tr>
<td>31: LINE Ins Send 31 -&gt; Mon In 31</td>
<td>Daw Out 31 -&gt; Key In 31</td>
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<tr>
<td>32: LINE Ins Send 32 -&gt; Mon In 32</td>
<td>Daw Out 32 -&gt; Key In 32</td>
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<tr>
<td>33: MIC Ins Send 33 -&gt; Chan In 33</td>
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<tr>
<td>34: MIC Ins Send 34 -&gt; Chan In 34</td>
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<tr>
<td>35: MIC Ins Send 35 -&gt; Chan In 35</td>
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<tr>
<td>36: MIC Ins Send 36 -&gt; Chan In 36</td>
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<tr>
<td>37: LINE Chan Out 33 -&gt; Mon In 37</td>
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<tr>
<td>38: LINE Chan Out 34 -&gt; Mon In 38</td>
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<td>39: LINE Chan Out 35 -&gt; Mon In 39</td>
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<tr>
<td>40: LINE Chan Out 36 -&gt; Mon In 40</td>
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<td>41:</td>
<td>TURN OSC and PINK OM level pot fully down</td>
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<tr>
<td>42: LINE OSC Out ----&gt; Mon In 42</td>
<td>Daw Out 42 -&gt; Key In 42</td>
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<tr>
<td>43: LINE Ins Send 43 -&gt; Mon In 43</td>
<td>Daw Out 43 -&gt; Key In 43</td>
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<tr>
<td>44: LINE Ins Send 44 -&gt; Mon In 44</td>
<td>Daw Out 44 -&gt; Key In 44</td>
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<td>45: BUS</td>
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<td>46: BUS</td>
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<tr>
<td>47: LINE Chan Out 48 -&gt; Mon In 47</td>
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<tr>
<td>48: LINE Chan Out 48 -&gt; Daw In 43 (Non Lin Rev)</td>
<td>Non Lin Rev Send Level Control</td>
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</tbody>
</table>

DAW Out 47 & 48 -> Ext B 1 In

Mix Out L & R -> Daw In 47 & 48

Cue A L&R -> Lexicon PCM 96 IN
Lexicon PCM 96 OUT -> Echo Rtn 4 L&R

Effects:
Lexicon PCM 96: 200 Phil's Box Canyon

Cue B L&R -> TC M2000 IN
TC M2000 OUT -> Echo Rtn 2 L&R

Echo Rtn 3 R BROKEN ??

FX Send 1 -> DAW Input 45 (H-Delay)
DAW Out 45 & 46 -> Echo Rtn 1 L&R (H-Delay from Pro Tools)

FX Send 2 -> Mon In 48
DAW Out 5 & 6 -> Echo Rtn 3 L&R (Non Lin Rev from Pro Tools)
Also note that for channels 37 to 40, the signal feeding their Monitor inputs comes from the Direct Out of channels 33 to 36. As there were already instructions for establishing the feedback loop for channels 33 to 36 in these channels' first column, I opted to have these four separate loops documented in the first column of channels 37 to 40, instead of a second column for 33 to 36 – as I felt this second column was to be used for keying information rather than feedback loop entry and exit points. This means that the primary information for the first column (the one associated with the channel number) is the input being used, not the output. The second column is used to indicate extra connections – which ended up being just keying the dynamics sections in this particular composition. I started documenting on this same text file, in a third column, some of the manual manipulations required. In Session 1, for instance, I had rigged Channel 4 for a Frog Call, alongside the other two Frog Calls in Channels 5 and 6, which already had enough variation provided by automated channel fader movement included in the feedback loop. In Session 3, as I added another sound which needed manual manipulation for extra expressivity, I realised...
that a better place to store that information would be in the relevant Pro Tools tracks’ Comment box. This would allow the performer to see both these instructions and the visual cues as to when to execute them with the markers in the timeline, within the same document. I have therefore stopped annotating those manipulations in the text file. I have kept the two items mentioned above in the file provided in Appendix I, to support the demonstration of this thought process.

In hindsight, one extra column would be quite useful and help speed up the recall process: having the “channel names” for each of the channels, as per the Duality Remote channel names tab. Although this would be a redundant piece of information as it is already being saved when exporting the backup Duality Remote files from the console, it would improve the navigation around the console when recalling settings with the Settings – FX and Patchbay file, as the timbres would be more easily associated with their corresponding channels. For example, having just performed the piece a few months ago, it is easy to remember now, that Channels 5 and 6 were two Frog Calls. Three years from now, if I wanted to perform this again, would I remember this when reconnecting all of the patch cords? I would remember once the Pro Tools session is launched and/or the Duality Remote session information is transferred back to the console, as I kept the audio path names consistent in both platforms. I believe it would help, however, to have these names also stored in the Settings text file; a good place would be in the first column, to assist in memorizing the association of timbre name and channel number. Figure 80 shows an example of what this improved file would look like.

| 01: LINE Pls Md 1 | Channel Out 1 → Mon In 1 | DAW Out 1 → Key In 1 |
| 02: LINE Pls Low | Channel Out 2 → Mon In 2 | DAW Out 2 → Key In 2 |
| 03: LINE Pls Md 2 | Channel Out 3 → Mon In 3 | DAW Out 3 → Key In 3 |
| 04: MIC Frog 1 | Channel Out 4 → Channel In 4 |
| 05: MIC Frog 2 | Channel Out 5 → Channel In 5 |
| 06: MIC Frog 3 | Channel Out 6 → Channel In 6 |
| 07: LINE Ping | Channel Out 7 → Mon In 7 |
| 08: LINE Kick | Ins Send 8 → Mon In 8 | DAW Out 8 → Key In 8 |
| 09: LINE Zimmer 1 | Channel Out 9 → Mon In 9 |
| 10: LINE Zimmer 2 | Channel Out 10 → Mon In 10 |
| 11: LINE Zimmer 3 | Channel Out 11 → Mon In 11 |
| 12: LINE LF Drone | Ins Send 12 → Mon In 12 |
| 13: MIC SFX | Ins Send 13 → Channel In 13 | Tweak LMF Freq and HF Gain |
| 14: MIC Gong | Ins Send 14 → Channel In 14 |
| 15: LINE App 1 | Channel Out 15 → Mon In 15 |
| 16: MIC App 2 | Channel Out 16 → Mon In 16 |

Fig. 80: Example of possible improvements to Settings – FX and Patchbay text file
Other connections were documented, such as which inputs and outputs were used to record and monitor the main mix (a combination of Mix A, B and C busses), which Auxiliary Sends would feed which effects processing unit, and which Channels or Echo Return paths would be used to return those processed signals to the main mix. Last but not least, the oscillator was used to provide a “shaker-like” timbre as this could not be achieved despite numerous attempts with Larsen effects. Instructions were therefore added regarding its settings to the Pink Noise position and output level.

Aside from the patch bay connections, other data that is definitely not normally stored along with either the Pro Tools session or the Duality SE’s Total Recall information (whether in the console’s computer, or via the export of backup files with the Duality Remote software), are the settings of your outboard gear. I thought that the Settings file documenting the patch bay connections would be the best place to not only indicate where the reverberation units I used were connected, but also, to which user memory they should be set. If I had used any outboard gear with physical settings, typical of earlier time base processors which would have input and output potentiometers and even at times EQ parameters in the analogue domain, I would have noted the physical position of each of those settings. An alternative would have been to take pictures, and paste them in the Settings documentation. I decided early on in the research that I would not explore the myriads of possible timbres made possible by passing the console signals through various chains of analogue processing (mainly outboard EQ and dynamics processing), as I felt the focus was on the console’s ability to push the limits of No-Input Mixer music, not so much on chaining gear together, the possibilities of which have already been explored by David Tudor and David Lee Myers for several decades. Time base processing, however, was deemed acceptable as I felt that when producing music with conventional synthesizers, one would also add reverberation and various delay-based effects. As the use of such processing is not part of the focus for this research, discussion on how to port the performance to another venue can be found in Appendix A – Composition Session 01.

* The Snapshot 2 plug-in from Non Lethal Applications allows you to store pictures of your outboard gear, as part of the Pro Tools session information, within the .ptx file itself. It is also available as a VST or Audio Unit plug-in (Non Lethal Applications 2017).
5.2.2 Fine tuning procedure

As early as the second composition session, it also became evident that I would need to develop a process to fine tune specific Larsen effects, that could not be recalled precisely-enough by the Duality SE’s Total Recall system. The method I developed is as follows: for each channel that produces a Larsen effect which requires a very specific pitch, I would make a note of a particular location in the timeline where I knew which specific pitch was to be produced there. For instance, as seen in Figure 81, if I park the transport at the *Ring Mod Start on 28 Memory Location* (Bar 80 – 18:42), I am at a point in the piece where the elements which make up the seven-note chord on channels 25 to 32* are all at a stable pitch, or at least, Channels 25, 27 and 28 which are all supposed to “play” an E.

![Fig. 81: Example of Memory Locations to assist tuning](image)

A little further in the timeline at 126|2|612 (not shown in Figure 81), another Memory Location called *Tune Nappes* allows Channels 29 to 32 to be tuned to their respective pitches. A document therefore had to be drafted to keep track of what

* A reminder that Channel 26 has always refused to generate a Larsen effect, from the very start of the composition process.
pitches need to be achieved on which channels, at which time position. This document can be seen in Figure 82.

To use the *Settings – Tuning* document, one would locate the DAW’s transport to one of the Memory Locations in **bold**. In that second column, the channel numbers to be fine-tuned are also indicated. In the third column, details are given as to which of the channel’s controls need to be tweaked and which note must be obtained. I have also chosen to indicate in some instances, to which octave the note should be set. This is because although the Total Recall system should have already brought the pitch within a few semitones of the ideal note, I found that in some instances, there could be minute errors on several of the channel’s pots, leading to a cumulative error greater than an octave for the resulting note.

| TUNING                      | Ring Mod Start                  | 25  | Adjust LMF Freq for E  |
|                            |                                | 27, 28 | Adjust HPF for E       |
|                            | Tune Pads                      | 29  | Adjust EQ LF Freq for A5 |
|                            |                                | 30  | Adjust EQ LF Freq for F#4  |
|                            |                                | 31  | Adjust EQ LF Freq for E4  |
|                            |                                | 32  | Adjust EQ LF Freq for E3  |
|                            | Tune 2                         | 1   | Adjust EQ LF Freq and LMF Q for B2  |
|                            |                                | 2   | Adjust EQ LF Freq for E1  |
|                            |                                | 3   | Adjust EQ LF Freq for E2  |
|                            | LF Drone                       | 12  | Adjust Line Input Gain and EQ LF Freq for E1 and min harmonics  |
|                            |                                |      | Check tuning in Breakdown section |
|                            | Kick Tune                      | 8   | Adjust Line Input Gain for Low E and EQ LF Freq for Minimum buzz then increase until good attack transient during Beat and Frankie section |
|                            |                                | 43  | Nothing to tune so far  |
|                            |                                | 44  | Adjust Line Gain and Compressor Threshold for max consistency |
|                            | Tune Arpeggios                 | 15  | Adjust EQ HF Gain for F#4  |
|                            |                                | 16  | Adjust EQ HF Freq for unison with 15 @ F#4  |
|                            |                                |      | Fine Tune during Arpeggio Mundo section |
|                            | Tune Dive 1                    | 19  | Adjust LF Freq for G  |
|                            |                                | 20  | Adjust LF Freq for G for unison with 19 @ G  |
|                            | Check with Session 8 Mix @ Dub Siren |      | Adjust 45 Line Gain if need be, comparing with Mix from Session 8  |
|                            | Dub Siren                      |      |                             |
|                            | Tune Dive 2                    | 21  | Adjust LF Freq for A  |
|                            |                                | 22  | Adjust LF Freq for A for unison with 21 @ A  |
|                            | Check with Session 8 Mix @ Dub Siren |      | Adjust 46 Line Gain if need be  |

Fig. 82: Settings – Tuning.rtf
Additionally to the indications regarding pitch, in some instances, some notes are made indicating a particular timbral target. For example, the low frequency drone generated by Channel 12, to shake the room between bars 135 (20:31) and 146 (20:55), needs to generate an E1 with minimum harmonics.

At times, parking the transport at a Memory Location inside the piece, would have the console generate far too many simultaneous sounds to be able to focus on the particular channel(s) you want to fine tune. I found that it was easier to place Memory Locations far down the timeline – far beyond the point where even if one forgot to press Stop at the end of the piece, you would be likely to startle your audience with sudden burst of sounds – with just those sounds coming through. All the other “interfering” sounds are muted there. Memory Location “Kick Tune”, “Tune Arpeggios” and “Tune Dive 1 & 2” are some examples of these tuning-only locations, the earliest of which is at bar 2155. At times, as an extra precaution, there are indications to double-check the fine-tuning at another specific location, against another set of channels. An example of this is for Channels 1 to 3, which play an 8th note chord in the rhythmical sections of the piece, and shifting pads in the “Breakdown” and “Lone Notes” sections. Once they have been fine-tuned at the indicated Memory Location (Tune 2), it is also advised to check that the tuning also holds during the “Breakdown” section at bar 175 (21:57), as the 3 notes change to alternate between 2 chords from then on, until bar 220 (23:27). For the four channels generating the two stereo “Dive Bomb” sounds occurring from bar 220 (23:27) onwards, I found that it was safer to actually double-check their tuning with the recording of the desk output made during Session 8. This is indicated as such for the “Tune Dive 1 & 2” sets of instructions. Last but not least, the fine tuning of channels 15 and 16 being so critical to the piece and the overall success of demonstrating pitch control, I found that once I had fine-tuned at the “Tune Arpeggios” dedicated Memory Location at bar 3585, some additional fine-tuning could be done by playing the “Arpeggio Mundo” section.

To assist with all tuning aspects, I had at first thought that a good system would be to have a Tuner plug-in in a Pro Tools Aux Input path, connected to one of the console’s main stereo output channel. This proved to be unusable on most timbres that needed fine-tuning, however, probably due to the harmonic complexity of the
signals. The Tuner plug-ins (I tried several) were not able to latch onto any particular pitch. I therefore resorted to working by ear, using a Piano App on my iPhone (Junpei Wada n.d.) as a reference for each pitch.

5.3 Rehearsal and Performance

When I decided to start recording the desk output from Session 4 onwards, I was effectively rehearsing the performance of both new sections, and consolidating my execution of previous sections, as I was composing the piece. Although the four sets of files described above, constitute its score, and should enable its performance to go exactly as planned, when it came to performing the piece, I had the following considerations.

Although the Duality Remote files allow each Channel on the console to be named via six alphanumeric characters on an electronic scribble strip above the fader, too much time can be wasted when identifying which setting needs to be moved. This is especially true if one is not directly in front of the Channel, as parallax can have as a consequence that you manipulate settings on one of the adjacent Channels instead. In my early days working as house engineer for Sofreson, having had to mix by myself on a non-automated analogue console yet wanting to have as much production value as possible by adjusting as many settings as possible in real time, I developed a system to assist in quickly locating pots and switches*. It consists in placing a piece of paper with an arrow, and perhaps a few words instructing what to do, pointing to the setting(s) that need to be tweaked. This makes first identifying the Channel with the scribble strip, then running up the Channel to find the setting to tweak, unnecessary, as the paper is already pointing to it. I have used this system to assist with my performance of I Pressed Play.

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*I have not been able to find in the existing literature, any reference to other engineers and producers using this technique. I refuse to believe I am the first person to have thought of this, as I started engineering in 1985.
Figure 83 shows a wide shot, demonstrating how easy it is to see which settings should be reached for manipulation. Figure 84 is a close-up showing that for the Solo 1 sound, the EQ’s LMF Freq pot should be manipulated first, then the LMF Gain pot, then the LF Gain pot. The pieces of paper need to be positioned close-enough to avoid any ambiguity as to which setting the arrow pertains to, whilst at the same time, be far enough so that fingers do not dislodge the paper during the performance. I find that “sticky notes” are ideal as the adhesive band can be strategically positioned to stick onto nearby pot caps.

To assist in monitoring the cues given by the Pro Tools session’s Memory Locations, I found it was crucial to have the DAW computer’s screen in a centralised location. During my composition sessions, I had the session running on my laptop, resting on the far left of the console. This made reading the cues nearly impossible from channels 25 to 48, unless I had memorised the cues by ear. For the public performance of the piece, I made sure I had the DAW screen in the middle of the console.
Another realisation, which helped in the performance, was that my favourite mode for Pro tools window scrolling, which is Page (found under Options), was not ideal. In Page mode, the Edit Window only refreshes once the playback cursor has reached the right edge of the window, revealing what might be an event that is now too late to implement. To assist in foreseeing upcoming events and cues, I found that the Continuous mode is much more adapted to this type of work (see Figure 85). Indeed, by having “real time” always centred on the screen, depending on the horizontal zoom value, several bars of look-ahead can be constantly displayed.
5.4 Production of musical performance video

I opted to present the performance as part of a short documentary synthesising the entire research project, rather than on its own. I felt this would again assist in generating interest for the project, as well as provide an opportunity for online viewers to participate in the exact same survey that the live audience did. The video was produced in 1080P resolution but the highest available resolution for download from YouTube is 720P, thus the file made available on the USB flash drive is not full HD. Moreover, the 1080P file produced from Final Cut Pro is far too large to fit on the USB flash drive. The 1080P version can be watched online at: https://www.youtube.com/watch?v=SbSbqhlyG-M

The audio track used for the short documentary video was the I Pressed Play – Compressed version as I did not want viewers to damage their ear buds, speakers and most importantly, eardrums.
Chapter Six: Conclusion

With this project, I believe I have provided evidence which suggests that the music production industry could reconsider one of its main tools: the mixing console; to an extent where, provided enough momentum was generated with the concept, recording/mixing studios around the globe would also look at their downtime as potential for composing Larsen effect-based music. The feedback collected from peers and the general public was strongly positive – whether from audience members who attended the live performances, or people who have followed the progress via the online videos. The exemplar piece composed to demonstrate the potential of large format automated analogue consoles in No-Input Mixer music applications, has surpassed my own expectations and, I believe, has met or exceeded all the criteria set out in the sub-questions and benchmarks. The following sections provide additional information on the research’s various stages, as well as feedback analysis, possible applications, directions and implications, and areas for further study.

6.1 Timbre exploration

The first step of this research was to investigate the available timbres on the SSL XL 9000 K. Ten timbres were initially found at that stage, each being open to multiple variations in terms of pitch, harmonic content and in some cases, the number of emitted pulses per minute. When I ported the project to the SSL Duality SE – the XL 9000 K’s computer having ceased to operate – most of these timbres were confirmed as being also available on the new console. Most importantly, pitch control could now be achieved, using the Duality SE.

New timbres found very late in the compositional stages on the SSL Duality SE, show that I have not exhausted the sonic possibilities of this music production tool. Timbres such as the SFX found in Composition Session 03, the Gecko in Composition Session 05 or the dolphin-like timbre (named Flipper in the automation files) found in Composition Session 08, were not stumbled upon in the timbre investigation stages. Despite a systemic process, it is inspiring to know that there is potential for other artists to be able to utilise the tool in their own direction via other found timbres or variations. As stated in 4.3, many configurations were not explored.
One of the reasons for this was because they were deemed at the time to not produce any significant differences, such as configurations involving Aux Sends and Routing Matrix outputs as the exit point, instead of a Channel's Direct Out. Other configurations were not investigated in depth because the resulting timbres would be predictable, as happens when processing (EQ, Filters and Dynamics section) is outside the feedback loop. Finally, some of the investigation was left out due to time constraints: where the volume of exploration required would have taken too much time compared to other equally-important investigations into the automation systems’ capabilities. This is why timbres derived from combining a Channel and a Monitor path on the XL 9000 K were not investigated. Likewise, longer feedback loops involving several daisy-chained Channels (on the XL 9000 K or the Duality SE) were not tested.

Having had to trade the XL 9000 K for the Duality SE partway through the research, was a blessing in disguise. Some timbres could not be reproduced on the new console, such as the Throat-Like and Tearing Fabric sounds although I did stumble upon a variation of the latter in my second composition session, but was never able to re-obtain it in subsequent sessions. Some of the timbres sounded duller, which was compensated by applying corrective EQ on certain sub-groups. All those differences, however, including the reduced polyphony (having 48 signal paths instead of 112), were more than compensated by the ability to achieve pitch control.

6.2 Console automation system’s ability to store and recall timbres

The computer systems for both consoles proved to be a viable method for storing and recalling timbres. A methodical approach would have been to dial up each specific timbre on a different I/O module or Channel – via a copy/paste process between various sets of session files, to collate all timbres into a single Recall file. Additionally, the Channel Names function on the XL 9000 K (found in its Master Control Panel) or the Duality Remote application for the Duality SE, would be used to store corresponding timbre names.

In both cases – using the XL 9000 K Total Recall or Duality SE Multi-User Recall – manual tuning is necessary once the computer has deemed that all the settings are back to their previously-stored position. One cannot therefore rely entirely on the
Total Recall files for No-Input Mixer music. This is not a new problem on analogue consoles. Burgess’ words from 4.4.1 now take on a whole new meaning: “with most recalls, a degree of fine tuning by ear was necessary” (2014). Here, to be able to retrieve an exact pitch, one has to write down which pitch is meant to be achieved, and which controls on the Channel would best assist in obtaining said pitch. In this case, the “fine tuning by ear” is literally that: adjusting the Channels’ controls until each Larsen effect is in tune. Although some type of tuner is worth having as a visual aid, due to the harmonic complexity of some of the sounds in this particular piece, I had to resort to tuning by ear, using a Piano app (Junpei Wada n.d.) as my pitch reference.

Once in tune, the precision is such on the Duality SE that in some instances, I chose to purposefully detune an otherwise perfect unison. A prime example is the seven-note chord achieved in the first section of the piece: its three Es can be tuned to such perfect unison that there is no beating produced in between them. I found it more interesting to introduce a slight chorusing and beating effect, by detuning two of the channels during the fine-tuning procedure. This is then compounded by the fact that it is quite difficult to return from the Ring Modulation sweep effect on Channel 28, between Bar 80 (18:42) and 90 (19:12), and achieve perfect unison again with the other two “E Channels”.

6.3 Console automation system’s effectiveness as a sequencer

Rudimentary automation was achieved on the XL 9000 K (see 4.5.2) before the automation computer irremediably broke down. However, this single evaluation of its automation system has provided significant data regarding the use of a timecode-based system. This has allowed for an interesting comparison with the more modern approach on the Duality SE.

Programming on the automation system for the Duality SE, with its 8-Ctrl plug-in, has been an overall pleasurable experience. Being DAW-based meant there was no learning curve: I could immediately work at a great level of detail. In my composition, I was too ambitious in the range of notes covered by the arpeggio section, which led to two issues: many of the arpeggio notes are audibly out of tune, despite the excessive time needed to tune each of the two channels. I should have restricted
myself to a range that I knew was manageable from the Duality SE timbre exploration sessions. Had I still wanted to program the same arpeggio patterns, this would have been done with perhaps four or five channels instead of two. Some of the glissandos which cover too wide a range might not be possible when using more channels, as each is handling a smaller range (say, four semitones each). Using four or five channels for better arpeggio note control would also mean using four or five other channels for volume compensation, which on a 48-channel frame, could limit the creativity as you would be left with perhaps 30 channels for the other sounds.

The timing issues encountered in the final investigation stages were a major disappointment. I had considered that there could be some imprecision due to the protocols used between the DAW’s ⌘-Ctrl plug-in and the console, but I could never have imagined the time-keeping would be this poor. I could fortunately implement the vast majority of the rhythmical patterns I had in mind by using triggered expander/gates, but this is not ideal. First of all, the SSL Expander/Gate only offers 40 dB of maximum gain reduction: very loud signals in the mix still make it through if the rest of the arrangement goes quiet. One therefore needs additional level control in that case, by either sub-grouping to another pair of Channels (to preserve stereo image) where the Cut switch is automated and/or by automating the associated stereo Mix Bus. Second of all, having to set aside the expander/gate to take on the role that Cut switch automation should have been able to fulfil, means that one can no longer use it for other effects, such as what I intended to do early on in the conceptualisation for the piece: generate ratchet effects on some sounds, via other sounds from other Channels. Likewise, the timing issues encountered made it impossible to consider the Cut switch as a viable source of ratchet effects.

An unpredictable finding of Episode 08, 09 and 10, is that Pro Tools 10 has an additional quirk compared to Harrison MixBus 4. This is explained at 04:13 in Episode 10, although, typical of when one tries to demonstrate a random fault, I could not evidence the problem on camera. When opening a session in Pro Tools with ⌘-Ctrl plug-ins, I noticed that sometimes, particular channels would not respond, until I opened the plug-in and either clicked on the un-responsive command (say, its Cut button), or re-assigned the Channel number. At times, I have had none of the
plug-ins respond. Whereas with MixBus 4, perhaps it was just coincidental but I did not run into that issue throughout Session 10.

When composing the piece, I was often in the situation where complex automation programming on a Pro Tools track from one section, had to be copied over to another section; I made an important discovery to facilitate the process. In No-Input Mixer music applications, provided one doesn’t have to “cheat” the timing issues by having Signal Generator plug-ins triggering expander/gates on the corresponding Duality SE Channels, the ø-Ctrl plug-ins would be the only plug-ins in your Pro Tools session. Common sense would be to create Aux Input paths as these control paths are never going to contain audio to record or playback. I found that instead, Audio Tracks had a considerable advantage. When copying a section of automation data from an Aux Input, only that automation lane which is displayed is copied over to the new section. Whereas by creating Audio Tracks, and displaying Waveform once the various automation lanes have been programmed, allows a section to be selected and pasted elsewhere – containing all of the automation information. A further refinement to the method, which I did not think of at the time, would be to create a Clip Group once the selection is made in the Audio Track: this would allow the visualisation of the block of data that is about to be moved or copied, rather than having to memorise the start and end times.

When I first conceptualised the project, with the SSL XL 9000 K in mind, I had already thought of how programming mock-up sequences at home on a DAW, using a timecode-based grid, would then assist in programming the same events in the console’s automation computer. This technique revealed itself as being essential, had I been able to complete the project on the XL 9000 K, once some experimental sequence programming was performed for Session 03b. The amount of time taken to program an 8th note pattern over 5 bars on just three channels, demonstrated that any method that would assist in the programming of automation data would be a significant improvement over programming from scratch. To mimic the Larsen effect sounds at home, one could use the Signal Generator plug-in or even pre-recorded signals. Exact fader positions would be useless to pre-program, as once ported to the console’s computer, the odds that the positions would provoke the exact Larsen effect pitch envisioned would be infinitesimal. However, fader position pre-programming
could still be useful to determine the beginning and end of sections, for instance. Preprogramming of the Cut switch (or any other type of switch – Insert In/Out, EQ On/Off or Aux Sends On/Off) is where the process would really be beneficial, as this information transfers from one platform to the other with highly-predictable results. The comfort of being able to manipulate a user-friendly interface for hours on end instead of working directly on the console automation, is well worth noting. The technique of pre-programming material is just as effective if one uses a console with DAW-based automation, although, since you would be using the same DAW in the studio, your user interface would be identical. It is perhaps more productive, then, to work on sequences in the studio as you directly hear the results of your automation on Larsen effects.

6.4 Composing and performing the piece

My goal with the musical piece for this research being to demonstrate the degree of control the console automation can have on Larsen effects, I knew from the onset of the project that I would be delegating as much of the performance as possible to the computer. I would be following in the footsteps of Conlon Nancarrow and his career-long endeavour of "getting rid of the performers" (Slought Foundation.org 2004). In my scenario, the score is the set of instructions in the automation computer’s memory. One could argue that the videos shot for the pedagogical content, as well as the public performance, constitute a form of score for the performance, as other potential performers would be able to replicate what they see. After having performed the piece, produced the video documentary that features the performance, and reviewed the documents that are necessary for the performance, I realised that not all gestural aspects can be properly deduced. I was not aware that instructions that make perfect sense to me, would most likely be quite obscure for other performers. The Settings – FX and Patchbay document, and even more so, the Settings – Tuning document, would need some extra written explanation, akin to what can be found on Stockhausen’s scores (1974). Likewise, the Pro Tools Comments boxes are not big enough to fit all performance information, such as the clockwise to counter clockwise motion, combined with the Fast/Slow Attack switching, required on the Compressor’s Release Time on Channel 36, for the fourth solo timbre (Bar 292 – 25:51 in the I Pressed Play video). These instructions should therefore be part of the written explanations mentioned above. This could be in the form of a Performance
Instructions sheet and would co-exist with the Settings – FX and Patchbay.rtf and Settings – Tuning.rtf documents. I felt I could draw valuable techniques and inspiration from studying the scores from the composers in the literature review chapter.

When composing the piece, I ran into an issue on the Duality SE where the backup for the Total Recall file overwrote and corrupted previous data, meaning I lost all the settings that determined the timbres found in the previous session. This happened because I disconnected the Ethernet cable from my computer, linking the Duality Remote software (see Appendix F – SSL Duality SE Manuals / SSL Duality SE Manual, Section 6) running in my computer to the console’s on-board computer. As I noticed I had forgotten to backup the session data stored in the console’s computer, I reconnected the Ethernet cable, and it seemed like the Duality Remote software was again properly-linked to the console. Subsequent back-up and restore operations led to corrupt ZIP files, which overwrote the proper files for session 02. I would therefore strongly advise not to interrupt the communication between the console and the Duality Remote software until you are finished for the day. In case you do, I recommend shutting down both the console and the Duality Remote software, then reboot both (console first), and wait for proper communication to be established, before you attempt any remote file management.

In order to minimise possible damage to the console’s circuitry, as there is always a risk when exposing electronic components to voltages which are at the upper tolerance level for extended periods of time, I found that the Duality SE has the perfect tool to interrupt all feedback loops, if one wants to go on a lunch break for instance. In the Master Section, selecting all the channels using the Channel Select functions, then pressing Input Flip, instantly breaks all the feedback loops whose entry points are Mic, Line or Group Inputs. Once back from your break, simply press Input Flip again to continue working. Note that this will not break feedback loops using the Insert Return as the entry point. This technique can be applied to any console that has a global Input Flip function in its Master section, or as in the case of the Duality SE, an Input Flip function which can be applied to selected channels.
6.5 Peer feedback

Overall, the project was met with enthusiasm from the viewers of the online videos. Likewise, the industry professionals invited to the public performances showed their appreciation, some commenting that it had exceeded their expectations. The paper I submitted for the 2016 NIME was rejected mostly because it was still too early in its development stage. However, most reviewers have indicated that they found the project interesting, and were looking forward to hearing the final result, which indicates that to this particular audience – the experimental music composer – the project outcome would be significant. “The concept of re-purposing an analogue mixing desk as [a] NIME* is introduced in a novel, creative and performative way and I look forward to hearing the piece as it develops” (anonymous NIME conference secondary reviewer 2016, pers. comm. 29 March). “Perhaps this would make an interesting contribution as a paper or as a demo in the future with the addition of a description of the piece or issues that arose and were resolved in its production” (ibid.). Comments on the Youtube page following the post of the Skweeeel – I Pressed Play video also indicate that the response was very positive: “Fantastic!”, “Has SSL seen this yet? They really should, so awesome”, “amazing!!” (for details, see Appendix A).

The peer feedback system I put in place via these blogging channels (see 3.3), in an effort to spread awareness about the concept, generate constructive discussion, and possibly direct the final composition, was far from being as efficient as I had hoped. Apart from two comments which allowed me to refine my investigation into timbres and Total Recall precision, the rest of the feedback had no other value than to provide encouragement – which was at times necessary. I felt the results other than the ones mentioned above, were not significative for the outcome of the project, and are therefore not discussed here. Appendix A has details of peer feedback in each Session’s analysis section. Appendix L has the full transcripts of the surveys conducted on the day of the public performance – survey which was then extended by providing an online link in the published video of the performance.

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* New Interface for Musical Expression
6.6 Context in which the tool can be used

Several people have commented, towards the end of the project, that it could be interesting to “tour” the performance. Seeing that the Duality series of consoles (Duality, Duality SE and now Duality Delta) have been quite popular, and “firmly established as the industry standard for large-scale professional audio production worldwide” (Mix 2007; SSL n.d.), organising a performance of *I Pressed Play* in any of the studios equipped with a Duality SE would be fairly easy. I am not sure if the original Duality console would behave in a similar way (perhaps the \(\partial\)-Ctrl plug-in would not work with that series), but I am quite convinced that the newer Duality Delta would respond exactly as the Duality SE did. All I would need for the performance would be the four sets of files in Appendix I and J, making sure the studio has at least 76 patch cords, and has a Pro Tools system with at least 32 outputs (19 keying signals + miscellaneous). Some if not all of the outboard effects would most likely be different, which would require some extra preparation time. See notes regarding the porting of time-domain outboard processing in *Appendix A – Composition Session 01*.

Also suggested, a sample collection could be assembled, of the various sounds obtained. Some of the sounds are quite unique, and would most likely be impossible to reproduce on a conventional synthesiser. The tearing Fabric timbre comes to mind, as being so elusive I myself had trouble replicating it.

In terms of the applications of automated No-Input-Mixer music, the scope is quite vast. Hopefully, with this research, I have demonstrated that there is potential to use these techniques inside various genres of music outside of the No-Input-Mixer music niche – and not necessarily in electronic music. Indeed, although I have shown it is possible to compose and perform an entirely No-Input-Mixer-based production on a large format automated console, I believe the true potential lies in the application of its concepts (since pitch-control is achievable) to other genres. There is also the possibility of collaborative performances: on such a large console, there is room-enough for more than two people to manipulate parameters. Seen as there have been 96-channel frames sold (Mix 2007), one could envision multiple performers improvising over a solid, multi-channel Larsen-based fully-automated backing track.
6.7 Directions and implications

Throughout the timbre investigation process on both consoles, it was imperative that a sample of randomly-selected modules or channels be used. This was to establish whether a particular timbre was the result of either “normal” or “abnormal” behaviour, of the signal path and the electronic components involved. An example of the “normal”, i.e. standard expected behaviour, is that of a smooth transition between silence and the “high-pitched squeal” timbre, when crossing the tipping point. The typical sample comprised 5 different modules, spread across the 56 available on the XL 9000 K, either side of the central master section.

It quickly became apparent that whenever an I/O module’s path revealed any “abnormal” behaviour, such as an unstable and at times crackling high-pitch squeal at the tipping point, one of the features in that module path would later reveal itself as faulty. An example would be that the Group Output would not function in a module’s Channel path. Another example was that a module’s Auxiliary Send switch would not turn off. I have not documented the other instances where an abnormal Larsen effect behaviour was encountered but have, however, made a note that, invariably, at least one of the features was faulty in the module path.

The reason for these abnormal Larsen effect behaviours in relation to faulty features, at an electronic component level, is beyond the scope of this research. But logic would indicate that, since every single electronic component within the signal path contributes to the Larsen effect’s timbre, the slightest change within this path would have a consequence. One such change would be a faulty component: switch, resistor, capacitor, inductor, operational amplifier, etc. I have not been able to confirm at this stage that there is a 100% correlation between an abnormal Larsen effect and a fault within a module’s path, but this already shows a potential for the audio industry, beyond the boundaries of this study.

Analogue consoles require constant maintenance due to ageing components. This can become very time-consuming on a large-format console due to the extremely complex circuitry and numerous components, if the faults have not been duly catalogued as the various session engineers have come across them. Even then, a maintenance engineer would still need to check each module for additional faults that
no-one had yet noticed, from not having had to use that particular function. It would appear that an analogue console owner or maintenance engineer, could apply this technique for regular maintenance checks, by first identifying the normal Larsen effect behaviour of their console around the tipping point. This would be done by surveying a large number of modules and making a mental note of the most consistently-obtained timbre. Any deviation from that normal timbre, would therefore quickly flag those modules which require further investigation for faults, from the modules which are most likely in proper working order. On any large-frame console, this would most likely considerably reduce maintenance time, during which no income can be made.

I have emailed Stephen Crane, who is in charge of maintenance at Studios 301 Sydney, which happens to also have an SSL XL 9000 K, followed by a thirty-minute conversation. In this discussion, he has confirmed that “pushing everything to the limit inside a signal path will reveal defaults in transistors ready to fail” (2016, pers. Comm. 7 March). However, Stephen did not endorse my proposed method for quickly testing the health of a particular path, via the use of Larsen effects. According to him, a thorough testing of each functionality in each path (Channel and Monitor) is inevitable, but he has not ruled out the potential for the method entirely. It remains to be tested over a long-enough period of time, that correlations between the Larsen effect’s stability, and the presence of a fault in a signal path, can be established.

6.8 Areas for further study

As noted in 3.3, there are feedback loop configurations that are yet to be investigated on the XL 9000 K, such as combining a Monitor and Channel path, with processing in various positions in the two paths. Also not investigated, are feedback loop paths combining multiple Channels (on either the XL 9000 K or Duality SE) or Monitor paths on the XL 9000 K. Adding the Insert Point to these paths, with various outboard gear now part of the feedback loop, provides countless possibilities which would enable each artist to develop their own sounds. One of the areas I briefly touched when sub-grouping my three E4s, could be investigated much further: the summing of several feedback loops to a Channel via Aux Sends, Routing Matrix outputs, etc. or other types of summation points in a console, such as stereo Mix Busses, Monitor paths (XL 9000 K only). And of course, a much greater area for further study, is the
application of all these techniques to other automated analogue consoles, such as the ones from Neve, Harrison, Amek, Euphonix and API. I always felt that another perfect console candidate for this type of research would have been a Euphonix CS2000 – although its automation system is timecode-based (Funky Junk 2018).

To avoid the time-keeping issues encountered in the research due to the HUI / ipMIDI protocol combination, it would be interesting to have an opportunity to automate an analogue console via the EuCon protocol. Originally developed by Euphonix, EuCon carries automation information (and more) via Ethernet, but without first passing through a MIDI-format bottleneck (Milne et al. 2006). The information is passed on via Euphonix’s proprietary protocol, which, since AVID (Pro Tools) has purchased the company back in 2010, might have condemned any chance of that protocol being made available under license to SSL or API for their recent automatable analogue consoles (Sound on Sound 2010).

The available literature has no documentation of a large-format console having been used this way before; this research project will hopefully inspire other people in the audio industry – from studio owners to producers and artists - to investigate on their own and contribute to the vast potential in this field of knowledge.

I envision online collaborations and live webcast performances with other artists around the world, triggering Larsen effects between the consoles via Internet audio links. I also visualise a dedicated website centralizing all available information (methodology, performances, contacts with peers, …) for the No-Input Mixer music community

This research is most likely not my last incursion into the world of automated No-Input Mixer music production. I already have a new project in mind, which would make these techniques available to home-studio owners.
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