An Analysis of Roland's Super Saw Oscillator and its Relation to Pads within Trance Music

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By

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ABSTRACT

In 1996 Roland released the JP-8000, their first virtual analogue modelling synthesizer. The JP-8000 became very famous for one thing, the Super Saw. This unique sound helped define trance music and was most commonly used when synthesizing pads. Detailed research has been conducted providing an in-depth analysis of the Super Saw waveform along with basic synthesis practices that take place when creating trance pads. Parameters taken into account include waveform type, multiple oscillators, detune and phase. Relevant background theory behind the Super Saw has also been examined. These findings have been implemented in the recreation of trance pad timbres with an attempt to find out exactly what particular aspect constitutes to the Super Saw's iconic sound.

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1. INTRODUCTION

1.1 Background

Trance is a style of electronic music predominantly produced using synthesizers. It is often referred to as anthemic, ‘hands in the air’ music, employing long breakdowns and huge melodic up rises to create an uplifting sense (Snowman, 2004). Pads play a fundamental role in achieving this desired effect. The importance of timbre through subsequent synthesis processes should not be underestimated.

1.2 What is a pad?

Pads are simply sustained legato chords, frequently used to fill out areas of the frequency range. Pads within trance music are most commonly used in breakdown sections to enhance the atmosphere, creating a “floating universe” (Fassbender, 2008, p.70).

2. ANALYSIS

2.1 Example Selection

An analysis of pads in the following tracks (Table.1) has been conducted for both research basis and reference purposes that can be referred to at a later stage. These tracks were chosen from Dance Foundations independent survey of the top 1000 trance tracks ever created (Dance Foundation, 2011), all of which feature a prominent use of pads.

Track Name	**Artist**	**Original Release**
Airwave (Original Mix)	Rank 1	1999
Silence (Tiesto Remix)	Delerium feat. Sarah McLachlan	2000
Find Yourself (Cosmic Gate Remix)	John O'callaghan feat. Sarah Howells	2009

Table.1 – Tracks selected for analysis.

2.2 Analysis Findings

After listening to the pads within these tracks the following similarities have been found:

- Harmonically rich frequency content determines that they are most likely composed of multiple sawtooth waves
- Can be closely related to a string type of sound
- Detuning of waveforms to create a texturally thicker timbre
- High amount of with a long decay time to enhance harmonic content and the size of the sound
- Primarily fills out the mids and highs within the frequency spectrum
- Predominately feature in the breakdown sections

3. LITERATURE REVIEW

3.1 Basic Sawtooth Waveform

The sawtooth waveform is a complex analogue waveform that features a full series of harmonics. This gives the wave a naturally thick texture, with the ability to create extremely rich timbres that are characteristic of trance music (Hewitt, 2008). A sawtooth wave is comprised of a series of sine waves, each with a difference in both frequency and amplitude. The waves perceived note pitch is determined by the fundamental frequency, with each overtone relative in amplitude by its order in the series. A mathematical algorithm can be introduced to calculate the waves harmonic content. If f is the fundamental frequency and has an amplitude of x, then the resulting second harmonic in the series will be double the frequency, $2f$, and half the amplitude, $x/2$. The following harmonics continue this pattern, with the third harmonic, $3f$, having an amplitude of $x/3$ (Manning, 1993). Figure.1 demonstrates the frequency content of a sawtooth wave with the fundamental frequency of 100Hz. Although this diagram seems to suggest that overtones cease to exist beyond the tenth harmonic, this is in fact not the case. In theory, it is possible for the overtones to extend themselves to an infinite bandwidth, bound only by the oscillators analogue or digital components.

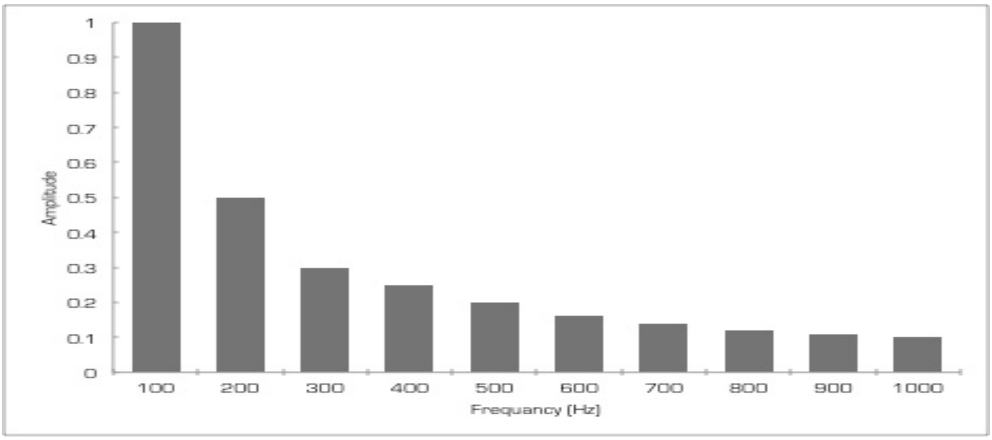


Figure.1 - Sawtooth wave frequency content (after Beau, 2006).

3.2 Multiple Oscillators and Detune

Using just a single waveform on its own to create a pad would sound very thin in comparison to the pads heard in the examples. To create the extremely thick sounding, big-trance style synth chords, multiple oscillators are used with a greater range of detuning (Music Radar, 2010).

When two or more oscillators are detuned, the structure of their harmonic content is altered. This has the effect of producing a much richer timbre. The reason this occurs is due to how the inner ear carries out its analysis of sound. When the fundamental frequencies of the two waveforms are identical, a sole pitch is heard. Detuning increases or decreases the fundamental frequency of the waveform by a relatively small amount, without the ear noticing a change in pitch. Depending on the amount of detune applied between the waves affects the way the ear perceives them. A difference of approximately 20Hz or less results in perceived fluctuations in amplitude due to interference between the two frequencies, resulting in a beating effect, demonstrated by Figure.2 (Rasch and Plomp, 1999).

Figure.2 - Resulting beats waveform of two tones with slight frequency difference (after Rasch and Plomp, 1999).

When the frequency difference between waves is greater than around 20Hz, the ear no longer perceives a beating effect, but an impression of roughness up until a point where a separation into two smooth single tones occurs. According to Pitkow (2000), the separation varies logarithmically with the centre frequency, which indicates the existence of a sensory feature called the Critical Bandwidth (CB). This term is used to describe an array of frequencies over which one sound will interfere with another (Figure.3).

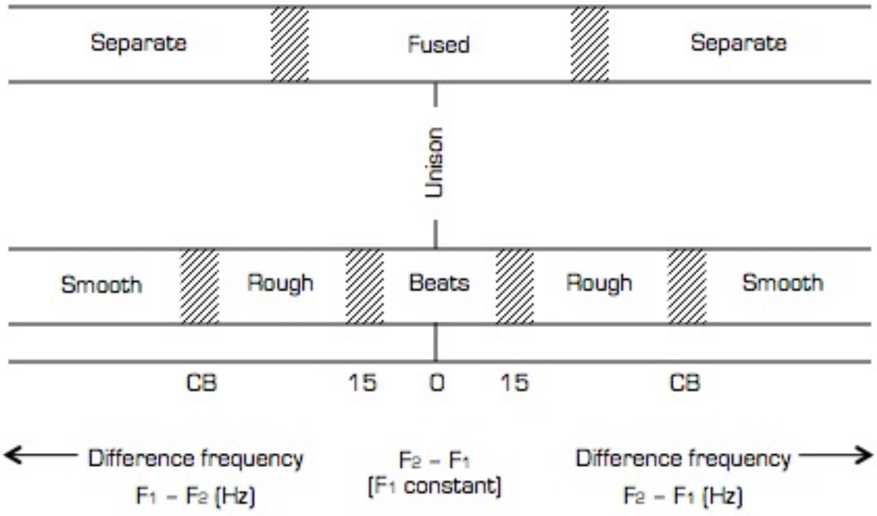


Figure.3 – Illustration of changes in perception of two tones of varying frequency with Critical Bandwidth areas (after Howard and Angus, 2006).

3.3 The Super Saw Oscillator

The Super Saw is a unique type of oscillator found in the Roland JP-8000 and JP-8080 range of virtual analogue synthesizers. Designed to emulate a combination of multiple sawtooth waves through the use of a single oscillator, Roland suggests it is “especially suitable for creating thick string-type sounds” (Roland Corporation, 1998, p.66). This could be a reason for it being regularly used for synthesising trance pads. According to Rank 1 (2012), the Roland JP-8000 Synthesizer was used to create the pad in the track ‘Airwave’. Along with this, many other trance producers still make use of this synthesizer within their music.

Notable users include: Ferry Corstern, Tiesto, Super8 & Tab, 7Skies and The Thrillseekers (Vintage Synth Explorer, 2012).

The Super Saw oscillator integrates seven combined sawtooth waves. The central wave is the master wave, which cannot be controlled by the oscillator parameters, thus remaining static in frequency and amplitude. The remaining six waves act as slaves that can be modified with control parameters.

3.3.1 Detune Control

The detune parameter works by spreading apart the pitch of the six waves around the central oscillator. The amount of detune applied determines the resulting offset of the six frequencies. A spectral analysis of this effect has been produced by Szabo (2010) shown in Figure.4.

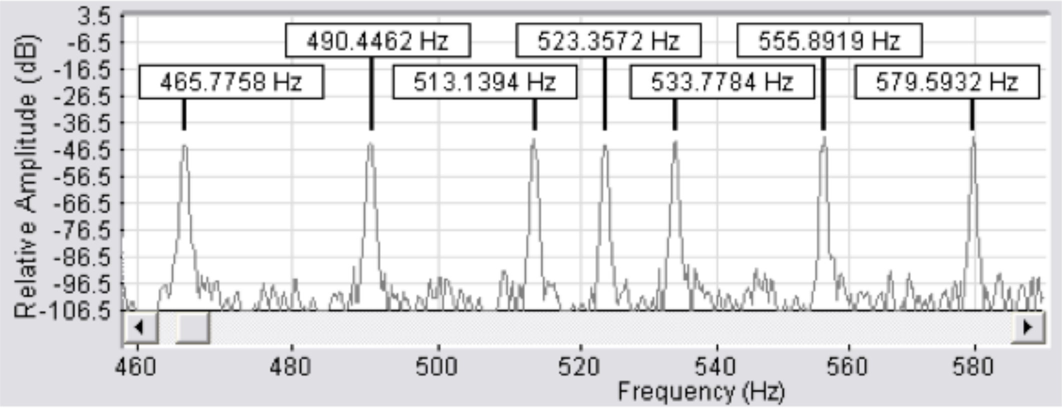


Figure.4 – Maximum detune at 523Hz [C5] (courtesy of Szabo, 2010).

The spectrum shows the behaviour of the Super Saw when its detune is set to maximum. The centre oscillator refers to the note pitch, in this case a frequency of 523Hz (C5). The diagram also displays the outcomes of the other six waveforms, three at either side of the central oscillator. From this analysis, it would be possible to determine the relationship between the detuned waves and the centre oscillator. Szabo (2010) also suggests that the detune amount is nonlinear, instead following a parabolic curve (Figure.5).

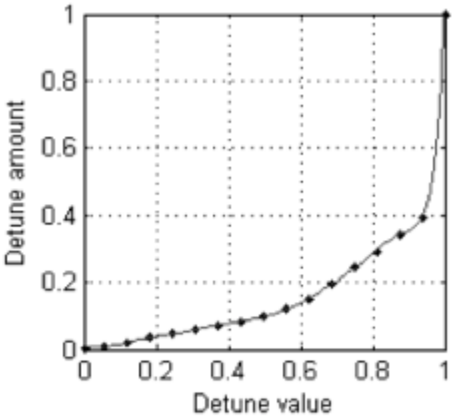


Figure.5 – Detune values plotted for every eighth interval (courtesy of Szabo, 2010).

Using a curve instead of a line allows the oscillators to be detuned by extremes at either end of the scale. When the detune control is set high, the waveforms exhibit a large frequency difference, so that it is possible that the ear would begin to move out from a perception of roughness, hearing separate tones instead. Alternatively, setting the detune control low allows the oscillators to be detuned by much smaller amounts, making it possible to create string-like pad sounds. This choice of the detune amount following a curve is most probably due to the emulation of analogue synthesis and popularity of its nonlinear aspects (Stilson, 2006).

3.3.2 Mix Control

The Mix parameter controls the amplitude of the six detune-able oscillators, without affecting the amplitude of the central oscillator. By increasing the volume of the detuned saw waves, the sound will be made thicker by masking the central frequency, causing the ear to perceive the roughness in sound described earlier. This mix control effect can be determined from Figure.6 taken from the operation manual of the JP-8080.

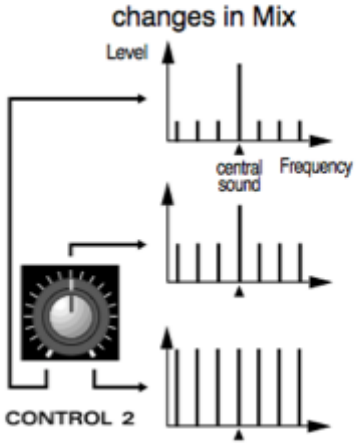


Figure.6 – Illustration demonstration the mix control for the Super Saw oscillator (courtesy of Roland Corporation, 1998).

3.3.3 Oscillator Shape

Whenever an analogue waveform is emulated in digital form an effect known as aliasing occurs. Aliasing is when a signal with a frequency that is greater than half the sampling rate is generated into a digital system. To avoid aliasing the Nyquist theory dictates that the sampling rate should always be at least double the highest frequency point being captured to achieve lossless sampling (Pohlmann, 2005). For example, a waveform with the highest frequency of 30Hz would demand a sampling rate of 60Hz to produce an accurate representation. However analogue waveforms rich spectra often consist of discontinuities, thereby having theoretically infinite bandwidth (Valimaki et al. 2010). It is possible when recreating analogue waveforms that inconsistencies in harmonics that are above Nyquist limit are aliased, thus introducing unwanted artefacts and noise.

In the case of the Super Saw, Szabo (2010) identifies that these additions of digital distortion are present and add depth to the sound, causing it to sound full and airy. In his analysis of the waveform he notes that the noise introduced below the fundamental harmonic has however been cut. This suggests that a high pass filter has been used to reduce that low frequency aliased content that has been generated by the waveform. Although, for this technique to function correctly the filter must follow the pitch of the oscillators, otherwise notes would be attenuated at certain point (Figure.7 and 8).

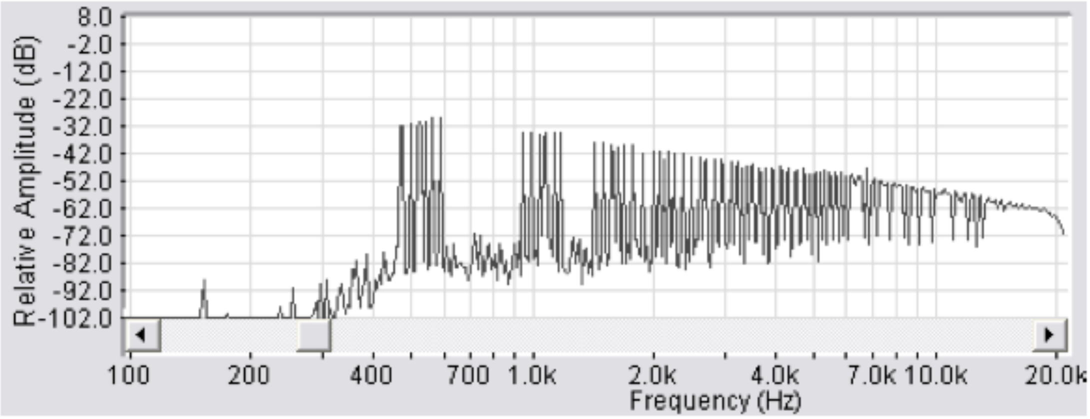


Figure.7 – FFT Spectrum of the Super Saw Waveform with noise cut below the fundamental, suggesting use of a high pass filter (courtesy of Szabo, 2010).

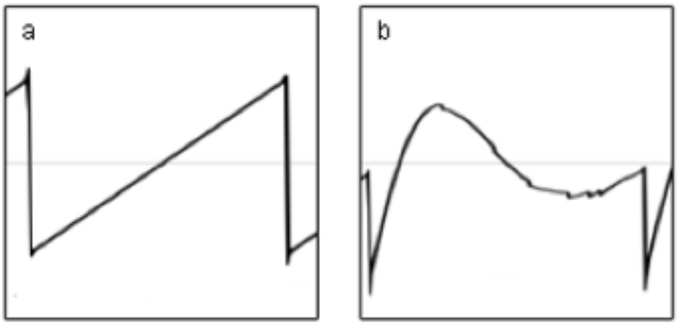


Figure.8 – Waveform comparison of a standard sawtooth wave [a] and the Super Saw [b] (courtesy of Szabo, 2010).

According to Valimaki et al. (2010), an emulation of a Super Saw waveform can be created by using the Bandlimited Impulse Train (BLIT) method, originally proposed by Stilson and Smith (1996). This introduces a filter band that limits the frequency contents of the wave to less than half the sample rate. This results in the higher harmonics to be excluded. A single band limited sawtooth can be generated with the following function:

$$Saw(n) = \sum_{k=0}^{n} BLIT\,(k) \leftrightarrow C_2$$

This can then be summarised to produce the seven waves similar to Roland's Super Saw, ($N = 7$). A reduction in aliasing is achieved by the harmonics masking the audio inefficiencies, making them inaudible to the human ear. Figure.9 shows the waveform of a sawtooth wave created using this method, along with a magnitude spectrum demonstrating the masked aliases.

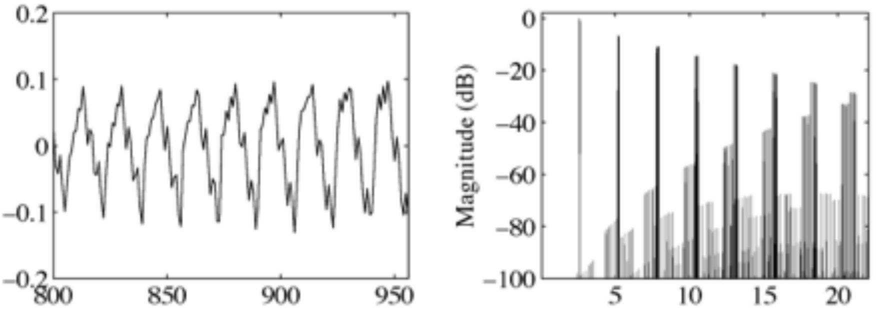


Figure.9 – Waveform and magnitude spectrum of Super Saw wave form with a fundamental frequency of 2631Hz with detune values 7, 14, 21 cents flat and 4, 8, 12 cents sharp (after Valimaki et al. 2010).

Compared to the waveform captured by Szabo, this emulation does not exhibit the same curved shape at the top of the wave. This could mean that this Super Saw emulation produces a different sound outcome to the Super Saw oscillator created by Roland. However, Szabo's findings do not provide any information about how Roland's waveform was generated before being sent through the high pass filter. Therefore, it is possible that this Super Saw emulation, using bandlimiting impulse responses, could suggest how Roland originally created the oscillator. Putting this emulation through the high pass filter to roll off any low-end aliases could produce a similar waveform to the original Super Saw.

3.3.4 Oscillator Phase

Phase refers to a specific point in a waveform or to a relative position when comparing multiple waveforms (Dodge and Jerse, 1997). When a multiple of waveforms that vary in phase are combined, a new wave is formed that behaves differently depending on their merged amplitudes. Szabo (2010) identified that the Super Saw incorporates a random phase shift between the seven oscillators (Figure.10).

Figure.10 – Recording of the Super Saw repeatedly playing the same note displaying the varying phase of the oscillators (courtesy of Szabo, 2010).

Oscillators that do not model static phase can be referred to as free-running oscillators (Mirzaei, 2010). This mimic the behaviour of voltage controlled oscillators found in analogue synthesizers as they are continually running. Whenever a note is triggered it will start at a different point depending on where the oscillator is in its cycle. This effect could have been implemented by using a low frequency oscillator to modulate the wave phase. Alternatively, the wave start time could be randomly delayed by a minute amount resulting in an irregular phase.

4. METHOD

4.1 Plan

Based on research and analysis it has been determined that the Super Saw is commonly employed for synthesizing trance pads. Previous findings suggest that the Super Saw exhibits interesting characteristics that give the waveform a unique sound. It is therefore possible that the Super Saws popularity for creating trance pads is due to these exclusive traits it possesses. The objective of this experiment will attempt to answer the following research question:

- What is the most important element(s) that gives the Super Saw its particular sound?

A simple emulation of the Super Saw oscillator will be produced based upon previous study discussed earlier. This will allow the freedom to manipulate the parameters in an attempt to deconstruct the Super Saw's sonic characteristics. By removing or changing certain aspects of the oscillator it would be possible to find out what are the real components that give it its specific sound.

After looking at relevant papers, for instance a paper from the Art of Record Production by Zeiner-Henriksen (2006), the author found that whenever similar judgements are made a test method of critical listening and subjective observations are made. These are then reinforced with a visual representation, with a comparison made between the two.

A subjective listening test will then be conducted to compare outcomes of the emulations to the original waveform from the author's perspective. Objective measurements will be conducted using FFT spectrum analysis and sonograms. The results of these two methods will then be examined against each other to find out if there is any correlation between the results.

To test each of these characteristics and their influence on the sound, the patch will be modified with the following variations (Table.2).

Parameter	**Patch Changes**
Detune	Detune amount doubled, halved and linear
Mix	Three, five and nine oscillators
Shape	No Filter, 6dB/Octave and 12dB/Octave

Table.2 – Parameters with their relevant patch changes for testing.

Testing will not however take into account the random phase offset. This has been deemed appropriate due to the fact that the phase only varies at the beginning of each note. This is therefore not entirely relevant when playing sustained legato chords, as the notes change less frequently compared to a melody.

4.2 Listening Test Method

According to Bech and Zacharov (2007), a listening test can be applied as a means to measure the following:

- Consider which sample is perceptually superior and to what extent
- Establish which audio system is preferred

As a result of this it has been determined that a listening test will be conducted to provide a perceptual judgement. Although the test will be designed to focus on the attribute in question, it is possible that other attributes not intended for measurement may cause bias, or influence the outcome. Findle (1997) has written that a level difference of more than 0.1dB could affect testing results, with louder perceived superior. The order in presentation can introduce contraction bias, with subjects referring back to the previous stimuli for reference (Bech and

Zacharov, 2007). They suggest that to avoid this bias the stimuli can be reversed. This could help eliminate a mentality that the first stimuli heard is always superior to the second. Careful control and consideration of irregularities will need to be taken into account to produce accurate results.

On the basis that the experiment is to compare two stimuli against each other to evaluate perceived differences, the listening test is effectively being used to measure their timbre in relation to each other. Russ (2009) describes timbre as a description of the sounds contents. This is determined by harmonic content and their relationship with the fundamental. Pratt and Doak (1976) defines timbre as an attribute of the auditory sensation by which the listener can make a judgement that two sounds are dissimilar without referring to pitch, duration or loudness. Moylan (2007) identifies the three primary characteristics that change how we perceive timbre to be dynamic envelope, spectrum, and spectral envelope. With these points in mind it is necessary to research descriptive techniques that can provide a standard for the listening test.

An analysis conducted by Zacharakis et al. (2012) produced a conclusion that timbre can be categorised as a description of:

- Luminance: Brilliance and Sharpness
- Texture: Roughness and Dirtiness
- Mass: Thickness and Fullness

Taking into account these three specifications, along with common adjectives found in the research section, the following relations to the test parameters have been made:

- Detune: Measure of texture (concluding of roughness and dirtiness)

- Mix: Measure of mass (concluding of thickness and fullness)
- Shape: Measure of luminance (concluding of brilliance and sharpness)

4.3 Listening Test Standards

Listening tests must be conducted in a scientific manner. Standards developed by both experts in the fields of industry and academia provide the benefit of an agreed approach (Bech and Zacharov, 2007).

A common standard used for conducted listening tests is the ITU-R BS.1116-1 developed by the International Telecommunication Union. It is recommended for use in the assessment of systems that introduce small impairments that could go undetected without rigorous control of experimental conditions (International Telecommunication Union, 1997). The test method allows the user to switch between three stimuli, A, B and C. Stimuli A is the known reference with B and C randomly assigned as the hidden reference. The listener is asked to compare B and C to A through a grading system. This test method uses the ITU-R five-grade impairment scale, see Appendix 13.2.

ITU-R BS.1116 is however intended for the assessment of small impairments, the requirements of which are stringent (International Telecommunication Union, 2003). With this test involving larger differences it has been determined that such close control of test parameters is not required. A test method that measures how different the stimuli are to one another would be more adequate. As a result, these tests will therefore be conducted in accordance with the ITU-R BS.1284-1 specification.

The BS.1284-1 follows similar principles to the BS.1116-1 but is intended for a more general assessment of sound quality. The author also feels that the BS.1284-1 provides advantages in its grading system for comparison tests. Here the listener is simply asked to compare reference A to reference B. It specifies that for paired comparisons excerpts should be no longer than 20 seconds. This test method uses the ITU-R seven-grade comparison scale, see Appendix 13.3.

4.4 Objective measurements

The spectral description of a sound has the most straightforward correlation with its timbre was concluded by Helmholtz (1895). The Fast Fourier Transform (FFT) technique can be used to obtain a spectral representation of a sound. FFT can be used to gain information and estimate harmonic amplitudes with relative precision (Dodge and Jerse, 1997).

A visual representation of the stimuli will be produced using FFT analysis software. Images produced will provide information relating to timbre by showing their frequency and amplitude content. Although this does not provide a definitive measure, they will be useful for showing differences and variations between stimuli allowing assumptions of timbre to be made. These will be compared to listening tests results to see if there is any correlation between the outcomes.

4.5 Testing Sources and Standards

A solo pad extract will be recorded for each of the sources. This has been decided for two reasons. Pads in trance are utilised predominantly within the breakdown, as noted in the analysis section. The addition of any other elements of a track, for instance drums, may distract the listener from focusing on the pad. Each extract will comprise of a four-chord progression. To help eliminate contraction bias, a different progression will be recorded for each of the parameters. Along with this the order in which the sources are played will also be mixed.

- Roland JP-8080 used to record the original Super Saw sources

- Samples will be recorded at 44100Hz, 24Bit Stereo into Logic Pro using an Apogee Duet 2 for conversion
- Sonalkis FreeG plugin used for level matching
- Sonic Visualiser and SignalScope used for objective measurements
- Emulation will be built using Native Instruments Reaktor

5. EXPERIMENT OVERVIEW

5.1 Aims and Objectives

The focus of this experiment was to determine exactly what particular aspect(s) of the Super Saw oscillator constitutes to its particular sound. The components tested were as follows:

- Detune Amount
- Number of Oscillators
- Shape of the Waveform

Additional testing was also conducted measuring the accuracy of the emulation built.

5.2 Overview of Experiment Structure

In order to identify the most important element(s) that constitutes to the Super Saw's sound, the following basic experimental structure was devised. From here a viable conclusion was then made in answer to the research question.

- Design and implement a successful emulation of the Super Saw oscillator based on relevant theory and research presented previously
- Emulations modified with certain parameters changed to create a relevant distinguishable difference in sound
- Comparison of emulations against the original through a series of critical listening tests and objective measurements
- Results cross-examined to ascertain the leading element
- Subjective listening test conducted with additional listeners to clarify results

6. REAKTOR PATCHES

6.1 Design Choices

Emulation was constructed using Native Instruments Reaktor due to its ease of use, previous experience and flexibility. Reaktor is a visual modular software package that provides the functions needed to build and reverse engineer the Super Saw (Native Instruments, 2012).

6.2 Construction of Master Patch

6.2.1 Basic Implementation

A Reaktor patch was built consisting of synthesizer functions equivalent to those found on the JP-8080 synthesizer. A basic design schematic with the implementation of the Super Saw oscillator was first planned out (Figure.11) that was then constructed in Reaktor (Figure.12). Although this experiment was only focused on the sound of the oscillator it was necessary to include other basic elements found on the hardware unit, allowing for the control and ability to set them the exact same as the original.

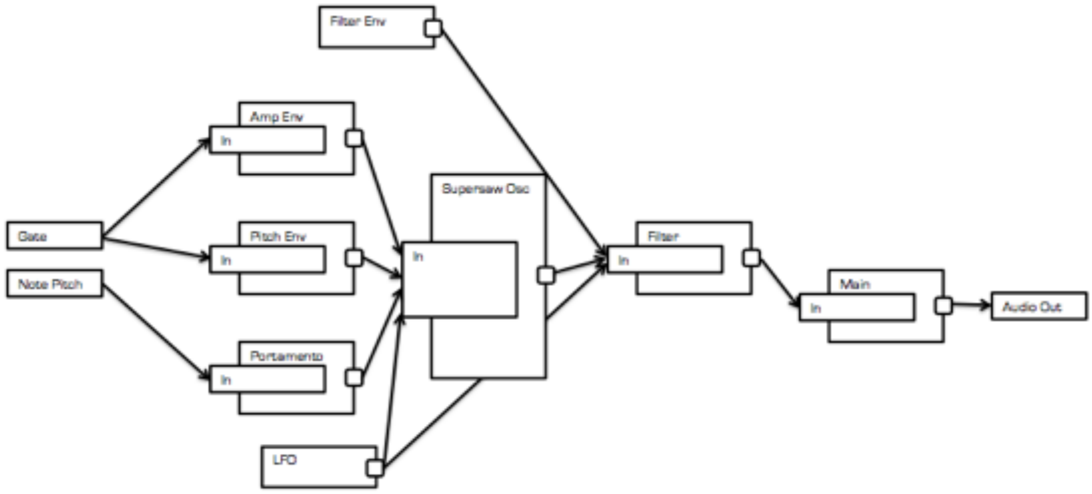


Figure.11 – Basic design schematic of the emulation built.

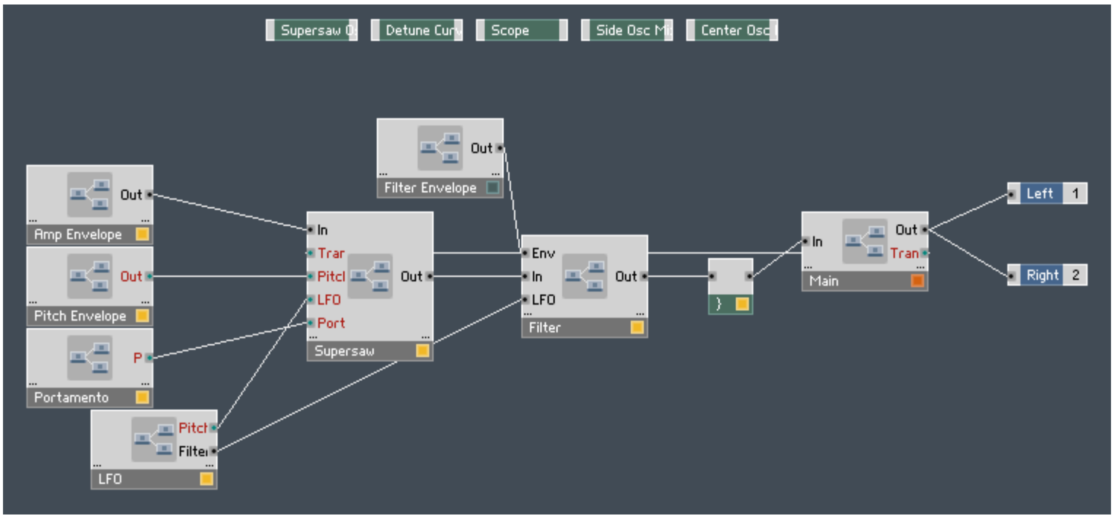


Figure.12 – Screenshot showing signal paths for individual modules within Reaktor.

The emulation of the Super Saw oscillator was implemented based on analysis techniques and findings presented by Szabo (2010) with the module correctly programmed to mimic detune amounts, mix amounts, number of oscillators, and waveform shape. Similar to the hardware unit, controls were added allowing alterations to be made to the spread of the oscillators (detune control) and their amplitude in relation to the centre oscillator (mix control). Other than these no other parameters on the oscillator could be adjusted. Again, a basic design schematic of the Super Saw oscillator, demonstrating the signal routing for within the module, was planned out (Figure.13). This patch was designed as an exact copy of the original, with no modified parameters, and is therefore referred to as the *master patch*.

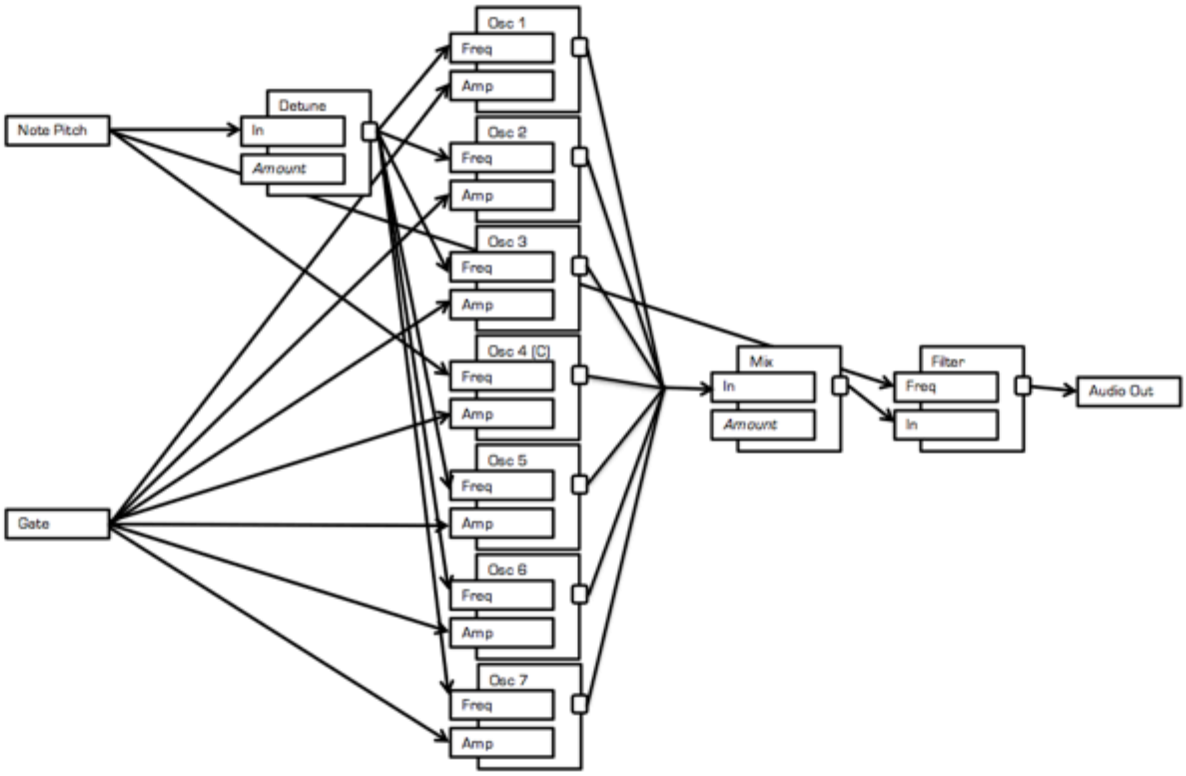


Figure.13 – Design schematic of the Super Saw oscillator within the emulation with user controllable elements shown in italics, oscillator 4 being the centre (C) oscillator.

6.2.2 Implementation of Detune Control

An event table module was used to create the detune curve with detune amount values entered as data points (Table.3). Event tables allow for flexible handling of data and provide the ability to create envelopes with curve shapes drawn by hand or with a number of breakpoints (Reaktor Application Reference Manual, 2010).

Data Point	Value
1	0.00967268
2	0.0220363
3	0.0339636
4	0.0467636
5	0.0591273
6	0.0714909
7	0.0838545
8	0.0967273
9	0.121527
10	0.147127
11	0.193455
12	0.243418
13	0.293382
14	0.343345
15	0.3928
16	1
17	2

Table.3 – Detune amounts used to create detune curve.

With these values entered, Reaktor then plots the curve in a stepped graph (Figure.14). In order to smooth out the curve, linear interpolation was used between the discrete values (Figure.15). This allows the table to produce a corresponding value between two data points to be sent to the output. The top was clipped to stop the graph resetting to zero.

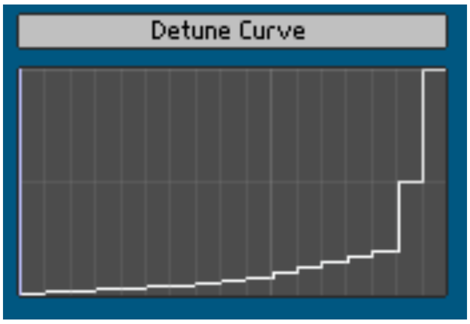


Figure.14 – Stepped graph plotted in Reaktor using detune amount for the 17 data points.

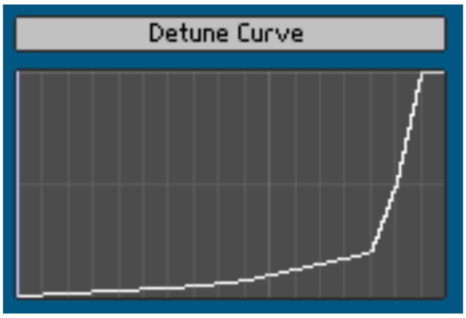


Figure.15 – Result of linear interpolation to create smoother detune curve.

To achieve the desired oscillator spread it was necessary to calculate the relationship between the six slave oscillators in respect to the centre. An FFT analysis was conducted of the hardware unit set to maximum mix and detune (Figure.16). The spectrum showed the relative frequency values of the slave oscillators when the centre oscillator is at 523.2549 Hz (C5). To find the relationship it was possible to divide each of the slave oscillators frequency by the centre oscillators frequency (Table.4).

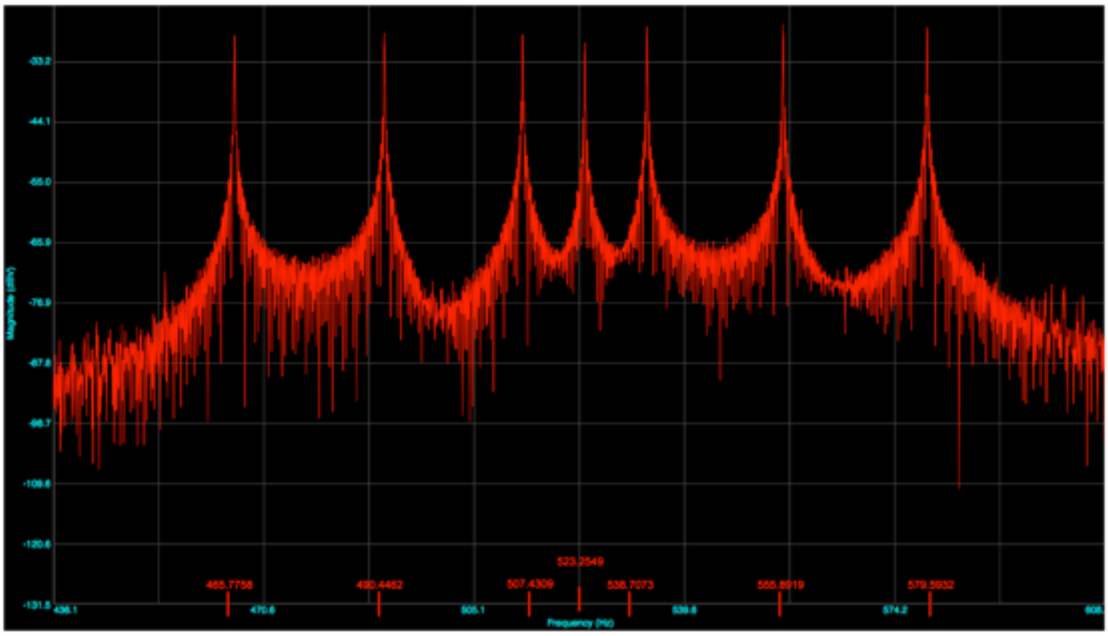


Figure.16 – FFT spectrum with maximum detune at 523 Hz (C5).

Osc Nr.	**Frequency (Hz)**	**Division by Centre Osc.**	**Relation**	**1 ± offset**
1	465.7758	465.7758 / 523.2549 =	0.88997686 =	1 - 0.11002313
2	490.4462	490.4462 / 523.2549 =	0.93711560 =	1 - 0.06288439
3	507.4309	507.4309 / 523.2549 =	0.96975852 =	1 – 0.03024148
4 (C)	523.2549	523.2549 / 523.2549 =	1 =	1 ± 0
5	538.7073	538.7073 / 523.2549 =	1.02953130 =	1 + 0.02953130
6	555.8919	555.8919 / 523.2549 =	1.06216538 =	1 + 0.06216538
7	579.5932	579.5932 / 523.2549 =	1.10745242 =	1 + 0.10745242

Table.4 – Results of relative oscillator frequencies.

From these calculations (after rounding) equations for each of the oscillator frequencies were determined below:

- Equation for oscillators three and five: $f(1 \pm \frac{1}{3}x)$
- Equation for oscillators two and six: $f(1 \pm \frac{2}{3}x)$
- Equation for oscillators one and seven: $f(1 \pm x)$

f = Fundamental frequency / x = Detune amount

The patch control functions by selecting the detune value along the *x*-axis and then outputs the corresponding detune amount from the event table along the *y*-axis. This value is then input into each of the equations producing the desired frequency for the oscillators (Figure.17).

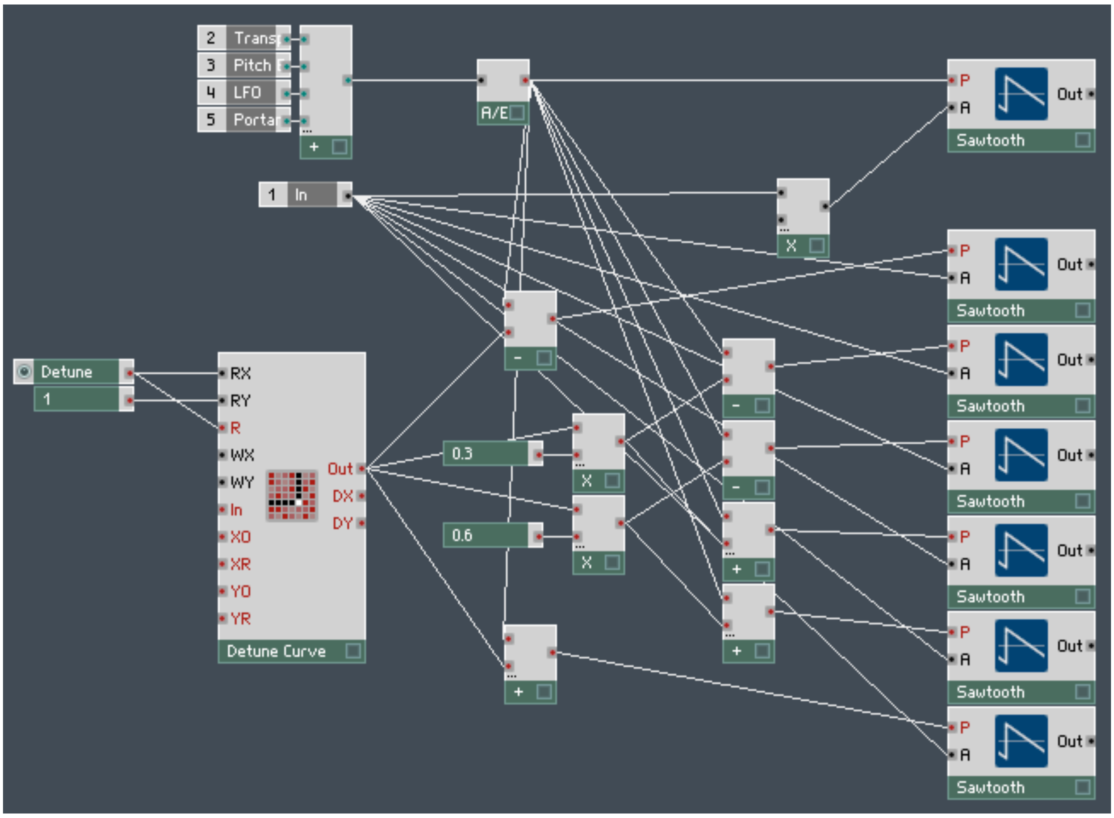


Figure.17 – Implementation of detune control within Reaktor patch, the lone sawtooth waveform being the centre (C) oscillator.

6.2.3 Implementation of Mix Control

The mix control was also constructed using event tables as it too required the ability to plot a graph from which values could be read. Although Roland (1998) suggests that the mix controls the level of the slave oscillators in relation to the centre, this is in fact not the case. An FFT analysis was conducted with the hardware unit set to maximum detune with the mix set a zero (Figure.18) and full (Figure.19). The spectrum shows the mix control also lowers the level of the centre oscillator to avoid clipping when the slave oscillators are increased in volume.

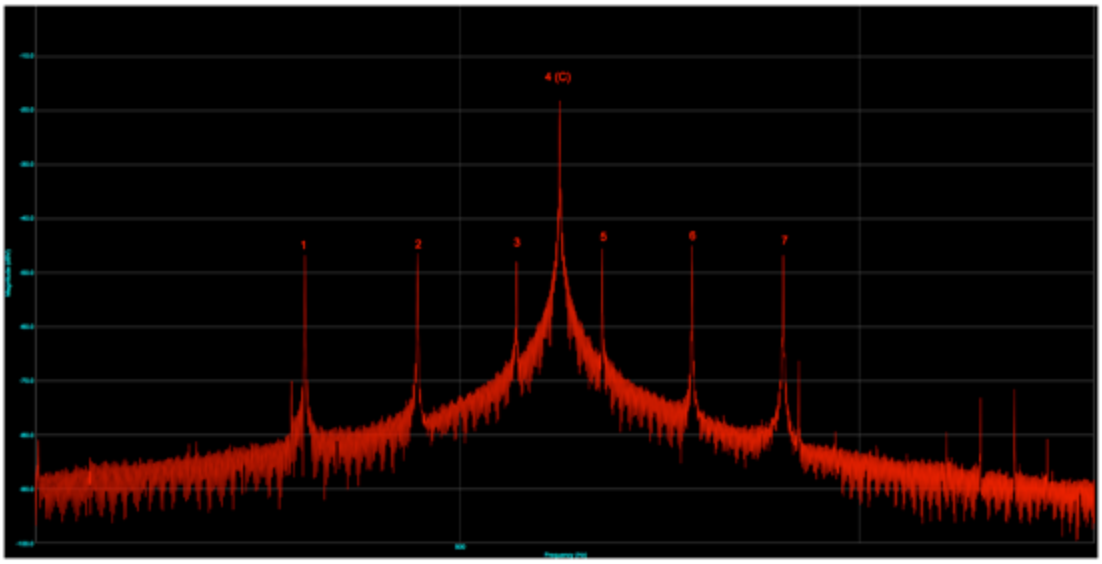


Figure.18 – FFT spectrum with mix set to zero with maximum detune at 523 Hz (C5).

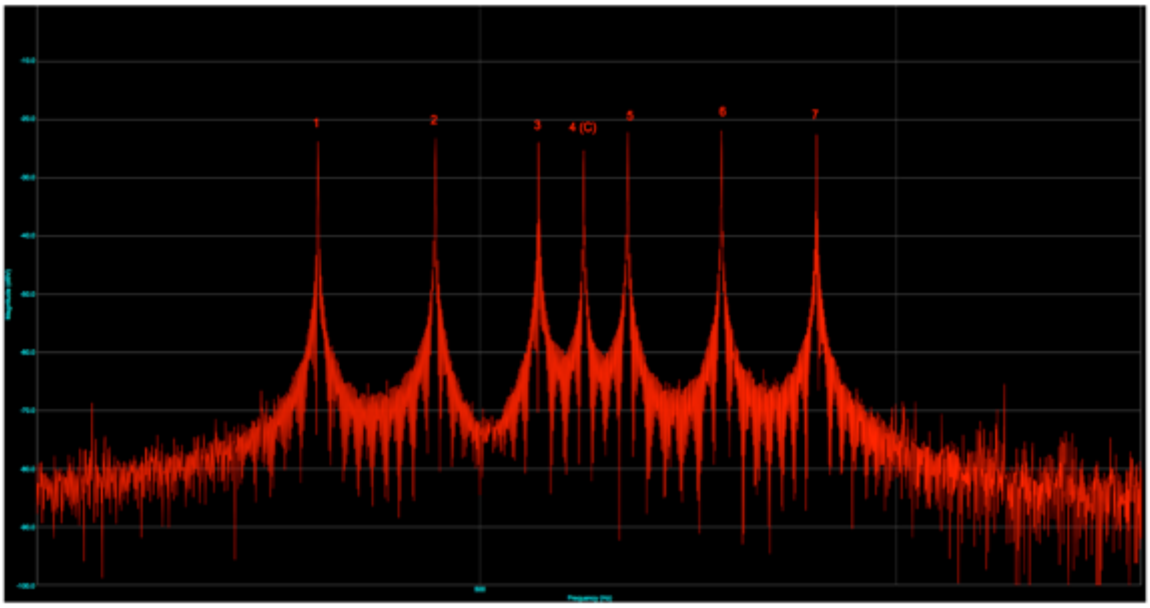


Figure.19 – FFT spectrum with mix set to full with maximum detune at 523 Hz (C5).

Similar to before, mix amount values were entered into an event table (Table.5). Two event tables were used to create the graphs needed for adjusting the parameter, one controlling the amplitude of the centre oscillator (Figure.20) and another controlling the amplitude of the slave oscillators.

Data Point	Value – Centre Osc..	Value – Side Osc.
1	1	0.03836
2	0.965	0.12
3	0.93	0.19
4	0.901	0.25
5	0.86	0.31
6	0.83	0.37
7	0.795	0.42
8	0.76	0.46
9	0.72	0.5
10	0.69	0.53
11	0.65	0.56
12	0.62	0.58
13	0.585	0.59
14	0.55	0.6
15	0.51	0.605
16	0.48	0.6
17	0.445	0.59

Table.5 – Values used to create mix curves.

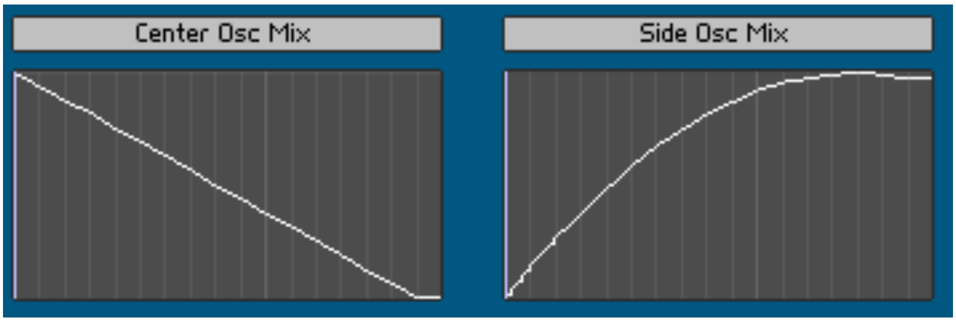


Figure.20 – Graphs plotted in Reaktor representing mix values for both the centre and side oscillators.

From these values equations for adjusting oscillator mix levels were determined below:

- Equation for centre oscillator mix: ax
- Equation for side oscillator mix: $(\sum_{i=1}^{n} y_1)x$

a = *Amplitude* / x = *Mix amount* / y = *Slave oscillator output*

The patch control simultaneously selects the mix values along the *x*-axis and then outputs the corresponding mix amount from the event tables along the *y*-axis. The centre mix table outputs the value for the central wave that is then multiplied by the amplitude and input into the oscillator. The side mix table outputs the value that is multiplied by the summed six waves (Figure.21). These two signals are then added together to give the final output.

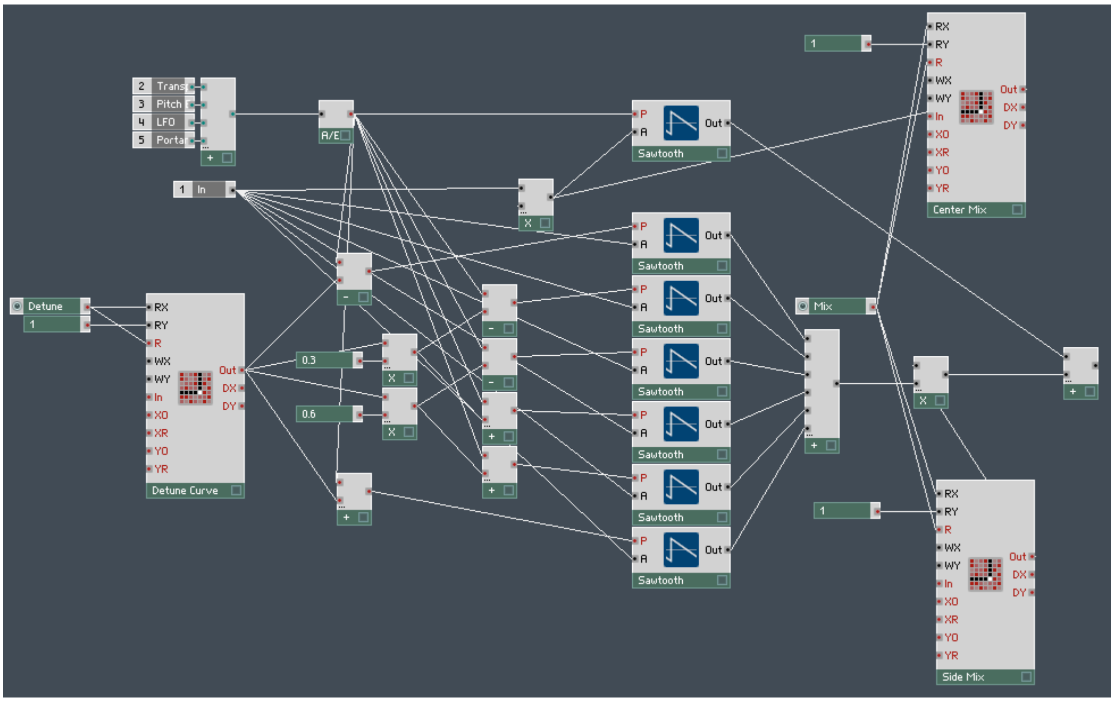


Figure.21 – Implementation of mix control within Reaktor patch.

6.2.4 Implementation of Oscillator Shape

An FFT spectrum of the Super Saw was produced showing the effect of the high pass filter that smooths the shape of the waveform, with the noise cut below the fundamental harmonic in dBV (Figure.22). The effect can be seen much clearer when viewing the signal in a linear scale (Figure.23).

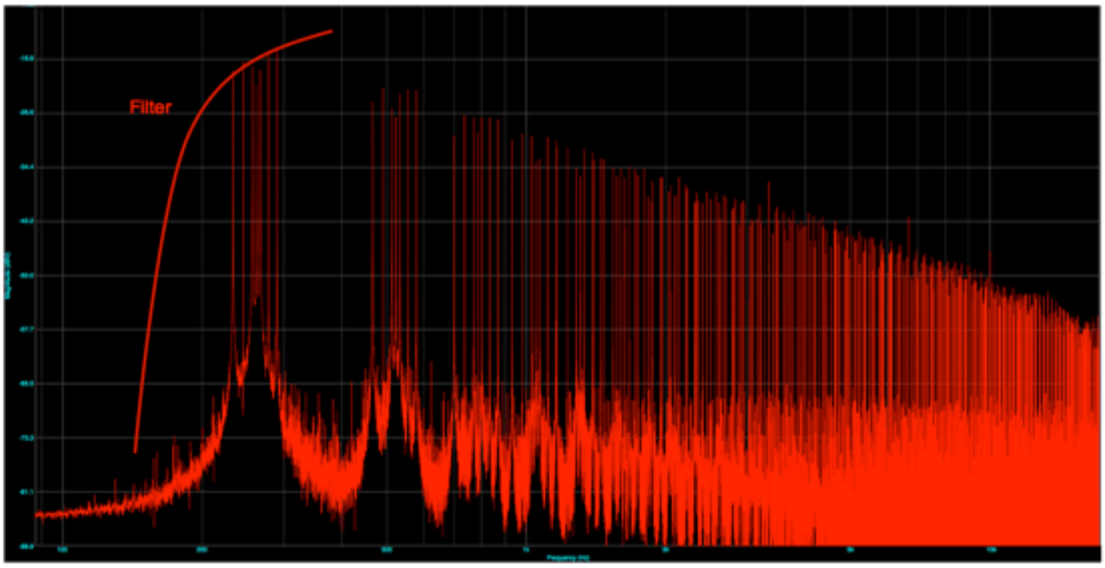


Figure.22 – FFT spectrum demonstrating noise cut below the fundamental harmonic in decibels (dBV).

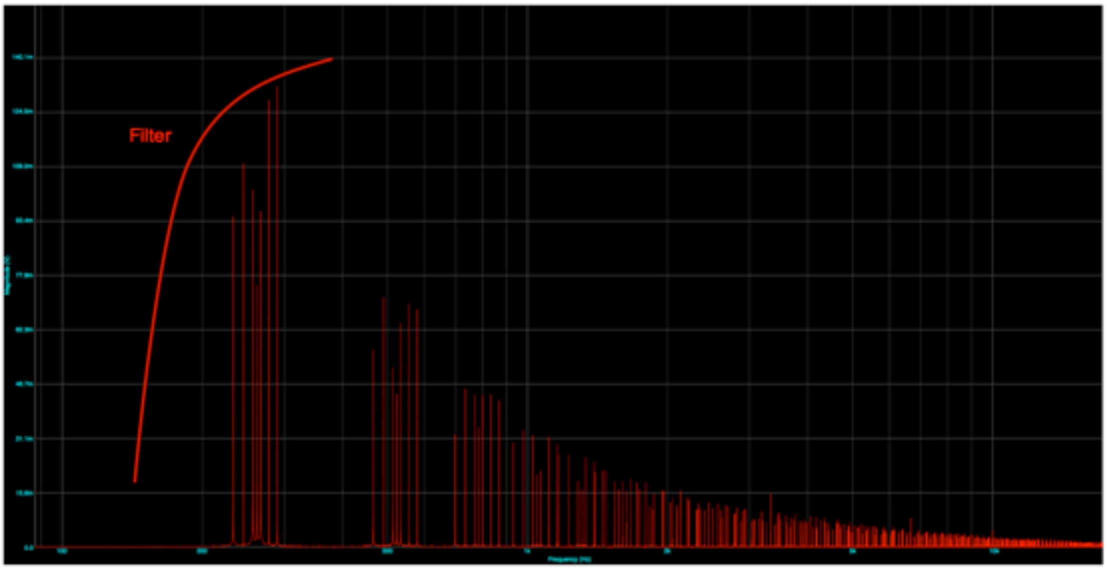


Figure.23 – Linear FFT spectrum emphasising noise cut below fundamental harmonic (V).

To reproduce this element within the emulation a fourth order (24/dB per octave) high pass filter was added at the end of the signal chain to replicate the steep cut off slope (Figure.24). In order for the filter to only cut the frequancies below the fundamental, a key-follow system was implemented using the note in frequency as the cut off input. The waveform was compared with the original in an oscilloscope to check the correct shaping of the oscillator through the filter (Figures. 25 & 26).

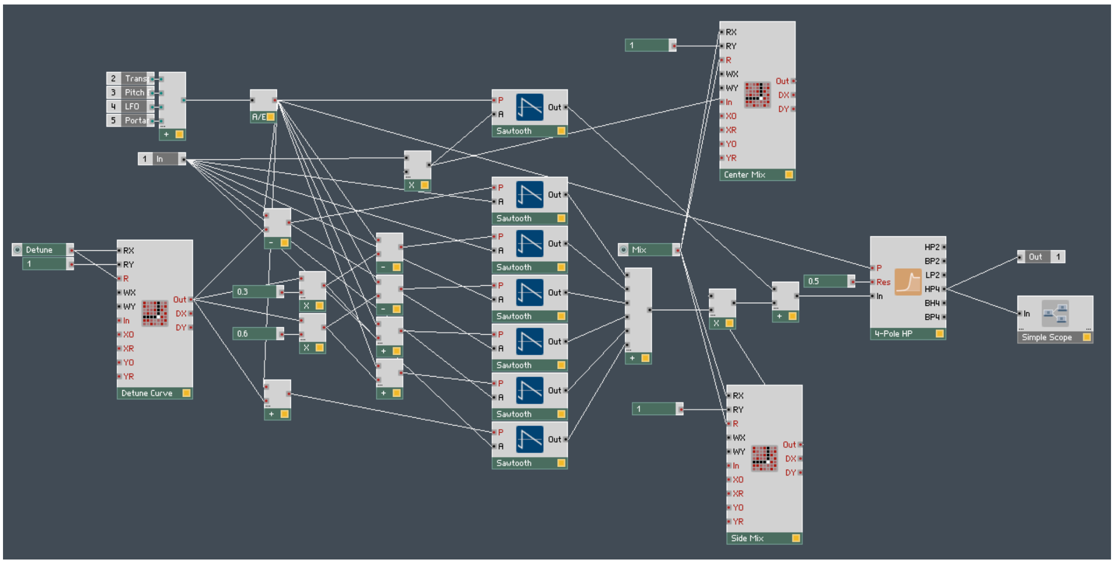


Figure.24 – Implementation of high pass filter within Reaktor patch.

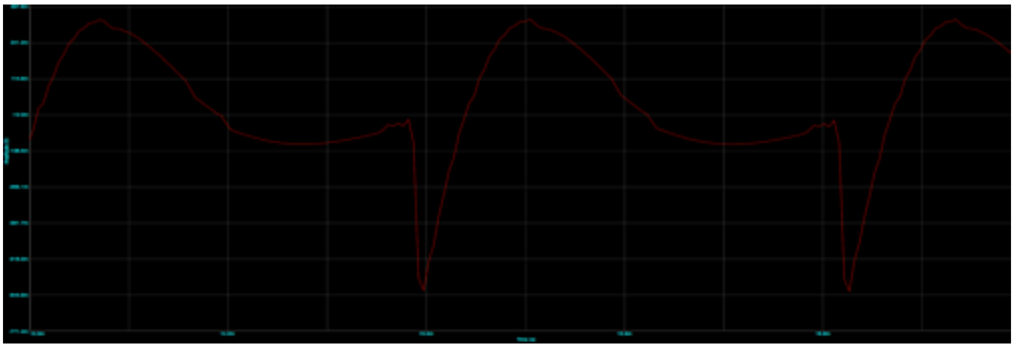


Figure.25 – Waveform shape viewed through oscilloscope from original hardware.

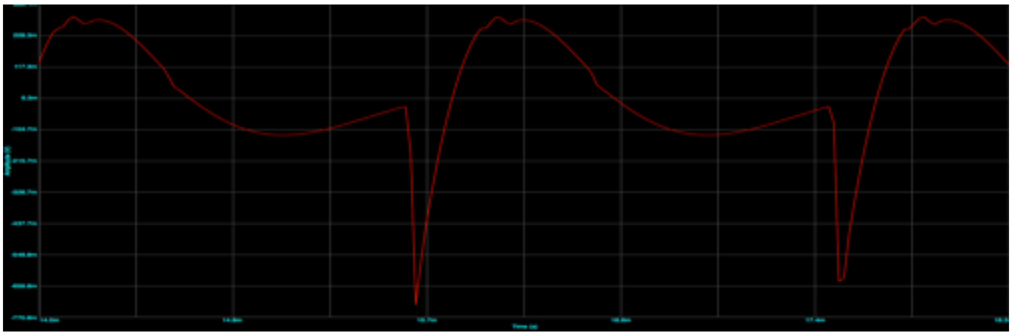


Figure.26 – Waveform shape viewed through oscilloscope from emulation.

6.3 Modifications for Oscillator Tests

6.3.1 Three Oscillators

Patch was modified with the number of sawtooth oscillators used reduced from seven to three. The three oscillators consisted of the centre oscillator and two closest side oscillators (Table.6).

Oscillator	1	2	3	4 (C)	5	6	7
Included in Patch	✕	✕	✓	✓	✓	✕	✕

Table.6 – Table showing oscillators used in modified patch.

6.3.2 Five Oscillators

Patch was modified with the number of sawtooth oscillators used reduced from seven to five. The five oscillators consisted of the centre oscillator and four closest side oscillators (Table.7).

Oscillator	1	2	3	4 (C)	5	6	7
Included in Patch	✕	✓	✓	✓	✓	✓	✕

Table.7 – Table showing oscillators used in modified patch.

6.3.3 Nine Oscillators

Patch was modified with the number of sawtooth oscillators used increased from seven to nine. The oscillators consisted of the original seven oscillators with another sawtooth oscillator added at either end. In order to correctly implement another set of sawtooth waveforms it was necessary to figure out the detune relationship between the oscillators. When designing the initial master patch, equations were formulated to calculate the applied detune amounts needed to achieve the correct frequency spread of the oscillators.

- Equation for oscillators three and five: $f(1 \pm \frac{1}{3}x)$
- Equation for oscillators two and six: $f(1 \pm \frac{2}{3}x)$
- Equation for oscillators one and seven: $f(1 \pm x)$

  f = Fundamental frequency / x = Detune amount

It can be seen here that the detune amounts result in the sequence, 0.3, 0.6, 1 (after rounding). Multiplying these numbers by a figure of 10 to give the values 3, 6, 10, it became apparent that the numbers follow Pascal's triangle sequence, starting from the second term. The triangle rule was used to calculate the next detune amount with the result then divided by ten to give the relative value.

- Triangle rule calculation for detune amount: $x_n = \frac{n(n+1)}{2}$

$$x_5 = \frac{5(5+1)}{2} = 15$$

$$x_5 = \frac{15}{10} = 1.5$$

6.4 Modifications for Shape Tests

6.4.1 Filter Removed

Patch was modified with the 24/dB per octave high pass filter at the end of the signal path completely removed.

6.4.2 First Order Filter

Patch was modified with the 24/dB per octave high pass filter at the end of the signal path changed to a first order 6/dB per octave high pass filter.

6.4.3 Second Order Filter

Patch was modified with the 24/dB per octave high pass filter at the end of the signal path changed to a first order 12/dB per octave high pass filter.

6.5 Modifications for Detune Tests

6.5.1 Detune Amount Doubled

Patch was modified so that values output from the detune table were multiplied by two, henceforth doubling the amount of detune applied to the oscillator. New equations were as follows:

- Equation for oscillators three and five: $f(1 \pm \frac{1}{3}(2x))$
- Equation for oscillators two and six: $f(1 \pm \frac{2}{3}(2x))$
- Equation for oscillators one and seven: $f(1 \pm 2x)$

f = Fundamental frequency / x = Detune amount

6.5.2 Detune Amount Halved

Patch was modified so that values output from the detune table were divided by two, henceforth halving the amount of detune applied to the oscillator. New equations were as follows:

- Equation for oscillators three and five: $f(1 \pm \frac{1}{3}(\frac{1}{2}x))$
- Equation for oscillators two and six: $f(1 \pm \frac{2}{3}(\frac{1}{2}x))$
- Equation for oscillators one and seven: $f(1 \pm \frac{1}{2}x)$

f = Frequency / x = Detune amount

6.5.3 Linear Detune Amount

Patch was modified so that detune amount would follow a line as opposed to a curve. Changes were made to the detune values within the event table, creating values of equal increments while still retaining the 17 data points (Table.8). Boundary points of 0 – 2 were also kept constant to the curve, otherwise this would have allowed for any line to be drawn that would have produced unfair results. Calculations for the linear detune values are as follows:

- Equal increments: $x = \frac{2}{17} = 0.11764706$
- Equation for linear detune values: $v\ (n) = xn$

v = Value / n = Data point

Data Point	Value
1	0.11764706
2	0.23529412
3	0.35294118
4	0.47058824
5	0.58823529
6	0.70588235
7	0.82352941
8	0.94117647
9	1.05882353
10	1.17647059
11	1.29411765
12	1.41176471
13	1.52941176
14	1.64705882
15	1.76470588
16	1.88235294
17	2

Table.8 – Detune amounts used to create linear detune.

7. STIMULI PREPARATION

7.1 Sample Preparation

A Roland JP-8080 was used to record the original sources. Emulation patches were built in Reaktor v5.8. All settings were initialised prior to recording ensuring equality between both products. Samples were recorded and bounced at 44100Hz, 24Bit Stereo into Logic Pro using an Apogee Duet 2 for conversion. Samples were peak normalised, with Sonalkis FreeG plugin used for level matching to ensure effective means of calibration for signals that are spectrally and temporally similar as stated by Bech and Zacharov (2006). See Appendix 13.4 for additional test setup information.

7.2 Testing Sources

ITU-R BS.1284-1 specification recommends that audio excerpts should be no longer than 15 to 20 s due to limitations in short-term human memory (International Telecommunication Union, 2003). The test method was split into four areas, with variables outlined below based on the different parameters, with a unique four-chord progression for each at 135bpm with no source time beyond 12 s. See Appendix 13.5 for progression notations.

- Number of Oscillators: Testing modified emulation with three, five and nine oscillators at three detune intervals (high, medium and low). Mix Control set to full, [Progression 1].
    - Dependant Variable: Perception of thickness and fullness

- Shape of the Waveform: Testing modified emulation with 1-Pole filter, 2-Pole filter and filter removed at three detune intervals (high, medium and low). Mix Control set to full, [Progression 2].

    - Dependant Variable: Perception of brilliance and sharpness

- Detune amount: Testing modified emulation with detune amount doubled, detune amount halved and a linear detune amount at five detune intervals (very high, high, medium, low and very low). Mix Control set to full, [Progression 3].
    - Dependant Variable: Perception of roughness and dirtiness

- Accuracy: Testing unmodified emulation at five detune intervals (very high, high, medium, low and very low). Mix Control set to full, [Progression 4].
    - Dependant Variable: Perception of accuracy

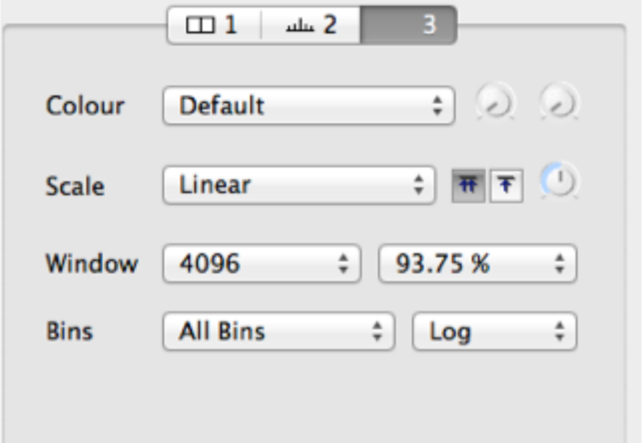
7.3 Observations

Audio from the emulations was compared against the original sources recorded from the hardware unit. Subjective and objective testing was initially undertaken with conclusions drawn from the results. A subjective listening test involving other audiophiles was later conducted based on observation findings.

8. TESTING STRUCTURE AND SETUP

8.1 Objective Measurements

Audio analysis tools can give a visual representation of recorded material, providing the ability to understand an audio signal. Samples were imported into Sonic Visualiser and spectrograms produced for analysis purposes. Figure.27 shows software settings used to generate these images.



Colour: Default
Scale: Linear
Window: 4096 93.75 %
Bins: All Bins Log

Figure.27 – Sonic Visualiser sonogram configuration.

Despite many definitions, no standardised methods for measurements of timbre currently exist. In order to produce a more definitive measure of the chosen timbral attributes, a selection of equalisation guides were consulted as reference points for the analysis of frequencies (IRN, 2013), (Baker, 2009), (Stereoklang Productions, 2013), see Appendix 13.6. From these it was possible to observe general areas of the spectrum where characteristics of audio are described:

- Thickness and Fullness: Found in the low-mid ranges around 250 Hz to 500 Hz

- Brilliance and Sharpness: Found in higher frequencies around 5 kHz – 12.5 kHz
- Roughness and Dirtiness: Found in the high-mid around 1 kHz to 6 kHz

8.2 Subjective Testing

Although objective measurements can provide meaningful results, not all audible characteristics can be visually reproduced in a way which correlates with what is heard, simply because no measurements for them exist (Lipshitz and Vanderkooy, 1980). A personal subjective listening test was conducted based on critical analysis in accordance with ITU-R BS.1284-1 specification for comparison tests.

Testing took place in an acoustically treated studio (Redrooms Two at Huddersfield University) in accordance with ITU listening room standards. Tests were conducted using a pair of Sennheiser HD 280 PRO's, closed-back, circumaural headphones designed for professional monitoring applications; see Appendix 13.7 for full specification. A seven-point absolute grading scale was devised based on ITU-R BS.1284-1 specification to record audible differences, see Appendix 13.8 for grading scales.

9. RESULTS AND DISCUSSION

9.1 Oscillator Test Results

Source audio used for the comparison test are found in the following folders:

- [JP-8080 > Recordings > Progression 1 > Dry]
- [Reaktor > Osc (Progression 1) > Modification > Recordings > Dry]

Recordings from the JP-8080 were compared against Reaktor emulations described in Chapter 2.3 with the number of sawtooth waveforms used modified. This test measured thickness and fullness using chord progression one. Results are presented in Table.9.

Test Parameter	**Detune Setting**	**Grade**
3 Osc	Quarter	-1
	Half	-3
	Three Quarter	-2
5 Osc	Quarter	-2
	Half	-3
	Three Quarter	-1
9 Osc	Quarter	0
	Half	-1
	Three Quarter	1

Table.9 – Results from subjective tests varying number of waveforms used.

9.2 Filter Test Results

Source audio used for the comparison test are found in the following folders:

- [JP-8080 > Recordings > Progression 2 > Dry]

- [Reaktor > Filter (Progression 2) > Modification > Recordings > Dry]

Recordings from the JP-8080 were compared against Reaktor emulations described in chapter 2.4 with the high pass filter type modified. This test measured sharpness and brilliance using chord progression two. Results are presented in Table.10.

Test Parameter	Detune Setting	Grade
No Filter	Quarter	3
	Half	1
	Three Quarter	1
1-Pole/6dB	Quarter	1
	Half	-2
	Three Quarter	2
2-Pole/12dB	Quarter	1
	Half	0
	Three Quarter	0

Table.10 – Results from subjective tests varying filter type used.

9.3 Detune Test Results

Source audio used for the comparison test are found in the following folders:

- [JP-8080 > Recordings > Progression 3 > Dry]
- [Reaktor > Detune (Progression 3) > Modification > Recordings > Dry]

Recordings from the JP-8080 were compared against Reaktor emulations described in chapter 2.5 with the detune control modified. This test measured roughness and dirtiness using chord progression three. Results are presented in Table.11.

Test Parameter	**Detune Setting**	**Grade**
Linear	Low	2
	Quarter	3
	Half	3
	Three Quarter	3
	Full	1
Half Amount	Low	-3
	Quarter	-3
	Half	-1
	Three Quarter	2
	Full	-1
Double Amount	Low	-1
	Quarter	1
	Half	1
	Three Quarter	1
	Full	2

Table.11 – Results from subjective tests varying detune amount used.

9.4 Accuracy Test Results

Source audio used for the comparison test are found in the following folders:

- [JP-8080 > Recordings > Progression 4 > Dry]
- [Reaktor > Accuracy (Progression 4) > Recordings > Dry]

Recordings from the JP-8080 were compared against unmodified Reaktor emulation originally constructed based on previous research. This test measured the overall accuracy of the emulation using chord progression four. Results are presented in Table.12.

Test Parameter	Detune Setting	Grade
Accuracy	Low	0
	Quarter	0
	Half	0
	Three Quarter	0
	Full	1

Table.12 – Results from accuracy test for original emulation.

9.5 Discussion

Initial test results indicated that although there appeared to be some slight differences in sound from the majority of changes, the most prominent differences were found between the linear and curved detune amounts.

Figure.28 compares sonograms from the linear test at low detune amount setting. Results from subjective analysis found that the sound was perceived to be rougher, with a distinct difference between the two examples, with the original sounding a lot smoother and static in comparison. Analysing the sonograms produced similar findings, with the original appearing smoother, showing much less movement between the waves in comparison to the modified emulation in which the beating of waves can be observed. Also note the slight increase of high-frequency content with the Reaktor example around 1 - 4kHz in accordance with the general descriptive area of roughness.

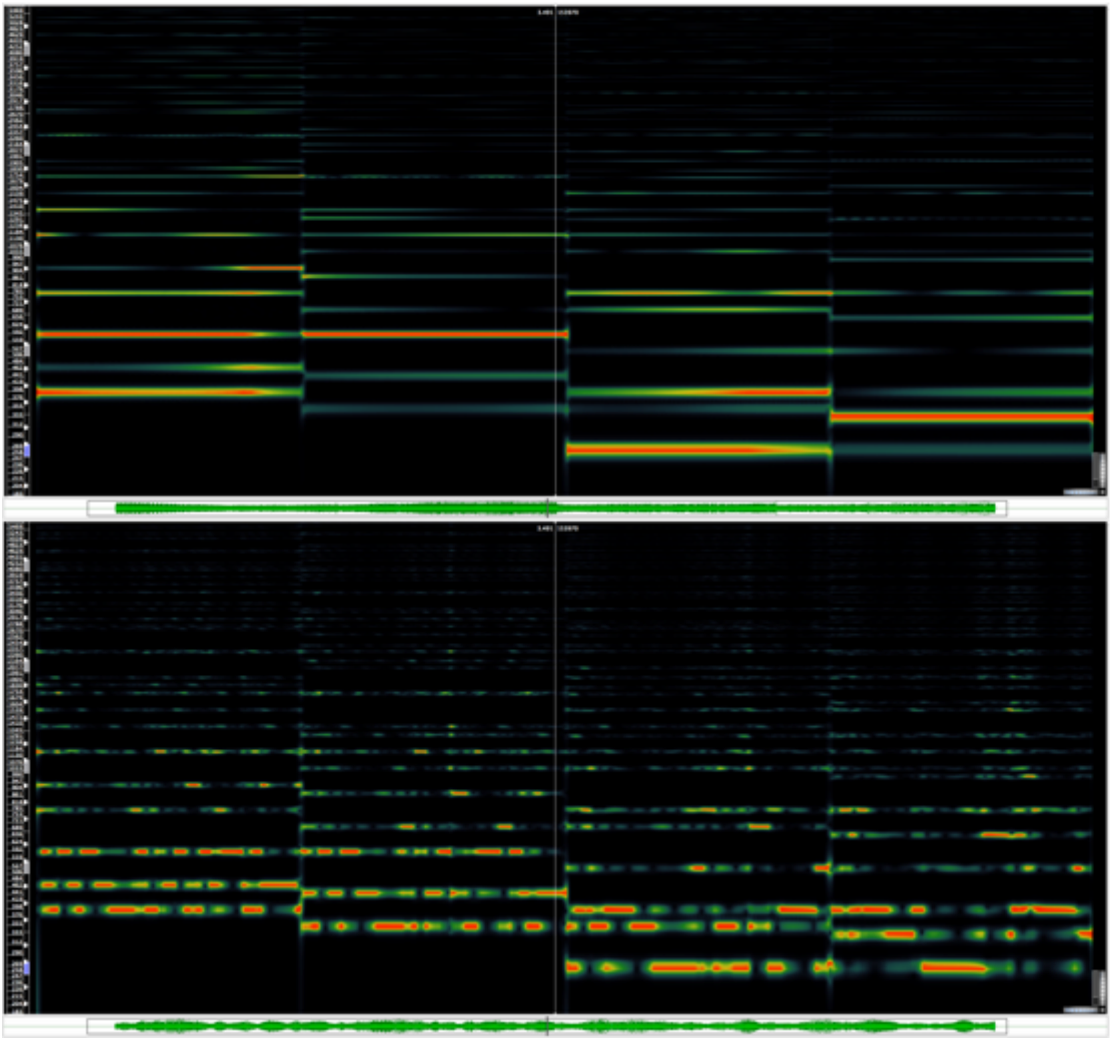


Figure.28 – Sonograms for linear detune test with control set to low (top is original JP-8080, bottom is modified Reaktor example).

Increasing the detune control to a quarter produced a large difference in texture, with the Reaktor example sounding much rougher and thicker in comparison, with a noticeably large frequency spread. Figure.29 shows the objective analysis from this test in which the greater spread of frequencies can be clearly observed with the modified emulation producing a more intense beating effect with a greater consistency of frequencies in the mid-range of 1 – 3kHz. However, the JP-8080 exhibits a larger concentration of higher frequencies, between 3 – 6kHz, which although found to be described as roughness are more possibly being perceived as sharpness.

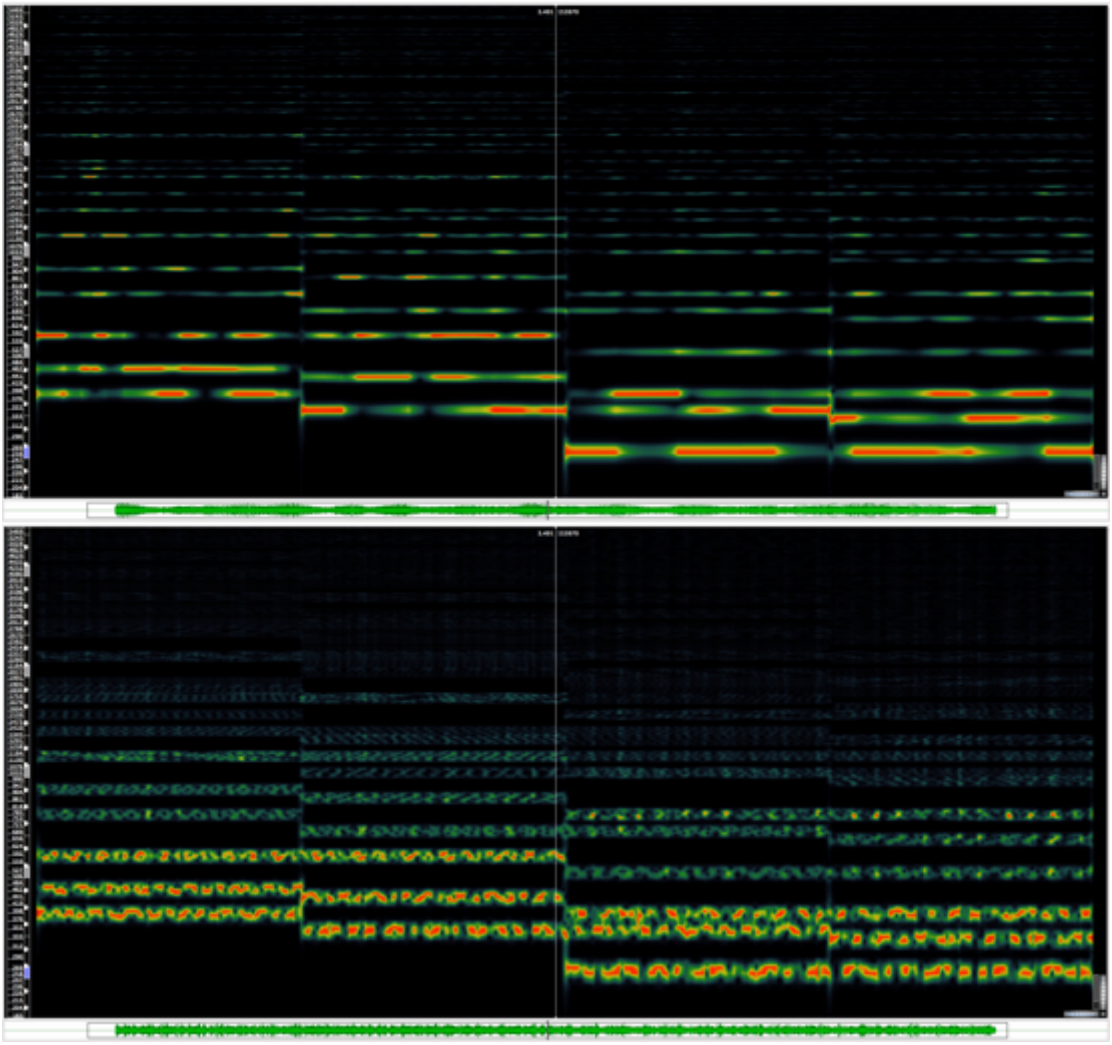


Figure.29 – Sonograms for linear detune test with control set to quarter (top is original JP-8080, bottom is modified Reaktor example).

Similarly, with the detune control set to half there is a considerable difference. Subjective analysis found the Reaktor example sounded much rougher with a greater spread of waves perceptible, beginning to sound harsh and unpleasant. Sonograms in Figure.30 indeed demonstrate a much wider spread of frequencies and increased beating effects, with some of the fundamental note frequencies starting to merge together, being particularly visible in the last chord around 250 - 400Hz.

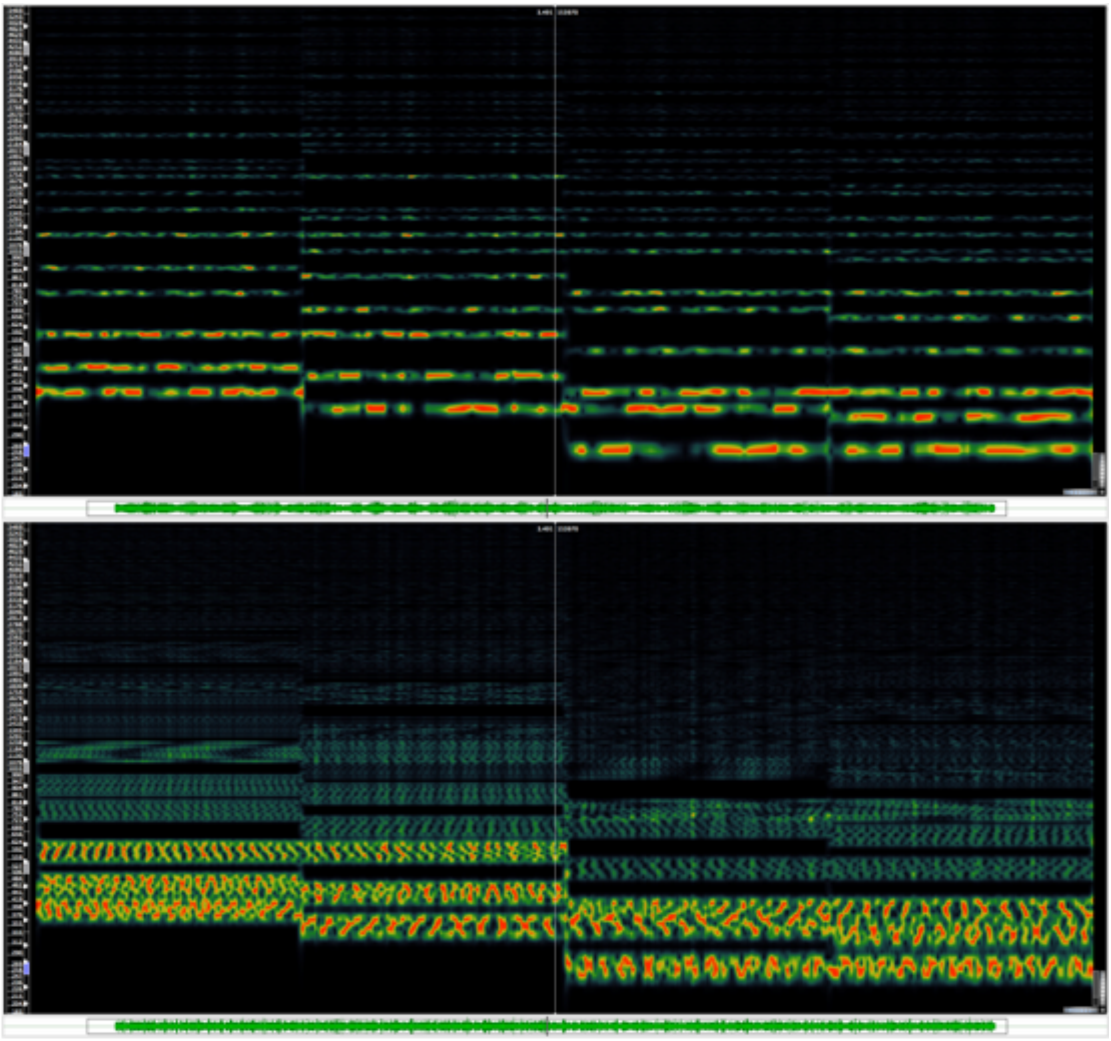


Figure.30 – Sonograms for linear detune test with control set to half (top is original JP-8080, bottom is modified Reaktor example).

With the detune control set to three-quarters subjective analysis again perceived a large difference in sound, the Reaktor example being much rougher. It was still possible to hear the basic chords in the original example, where as they were much harder to distinguish in the Reaktor example. This effect can be seen from the sonograms presented in Figure.31, with the modified emulation showing a merging of rough frequencies across the spectrum 300Hz – 4kHz, creating an extremely dissonant tone in comparison to the original in which it is still possible to view the fundamental notes.

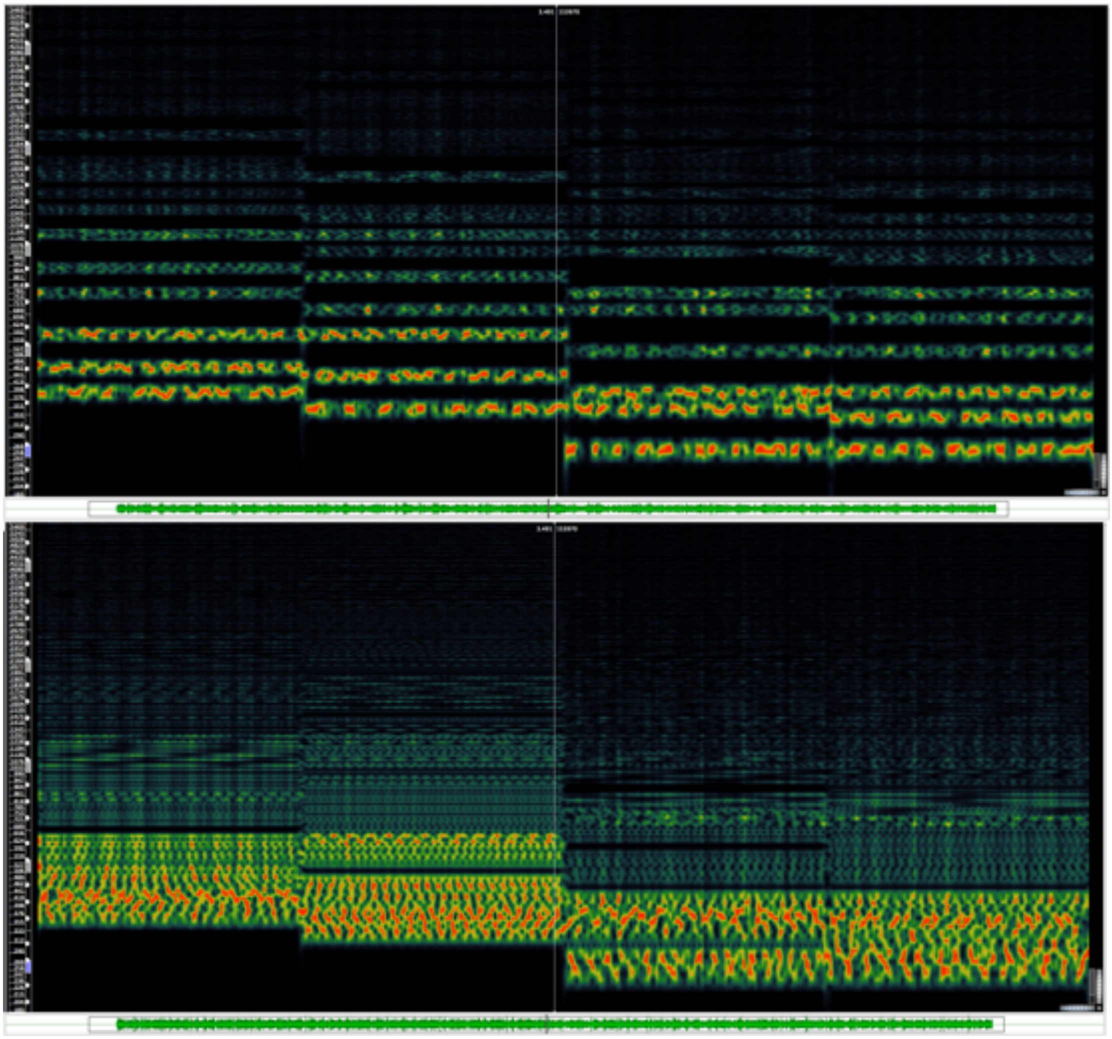


Figure.31 – Sonograms for linear detune test with control set to three-quarters (top is original JP-8080, bottom is modified Reaktor example).

At full detune the differences were much smaller, with the Reaktor example sounding slightly rougher than the original. Both sounded very dissonant with it being no longer possible to hear the original chords in either example. Objective measurements in Figure.32 show a similar pattern with the modified emulation exhibiting a dirtier sound due to the slightly wider frequency spread and more constant amount of frequencies in the 700Hz – 1.5kHz range.

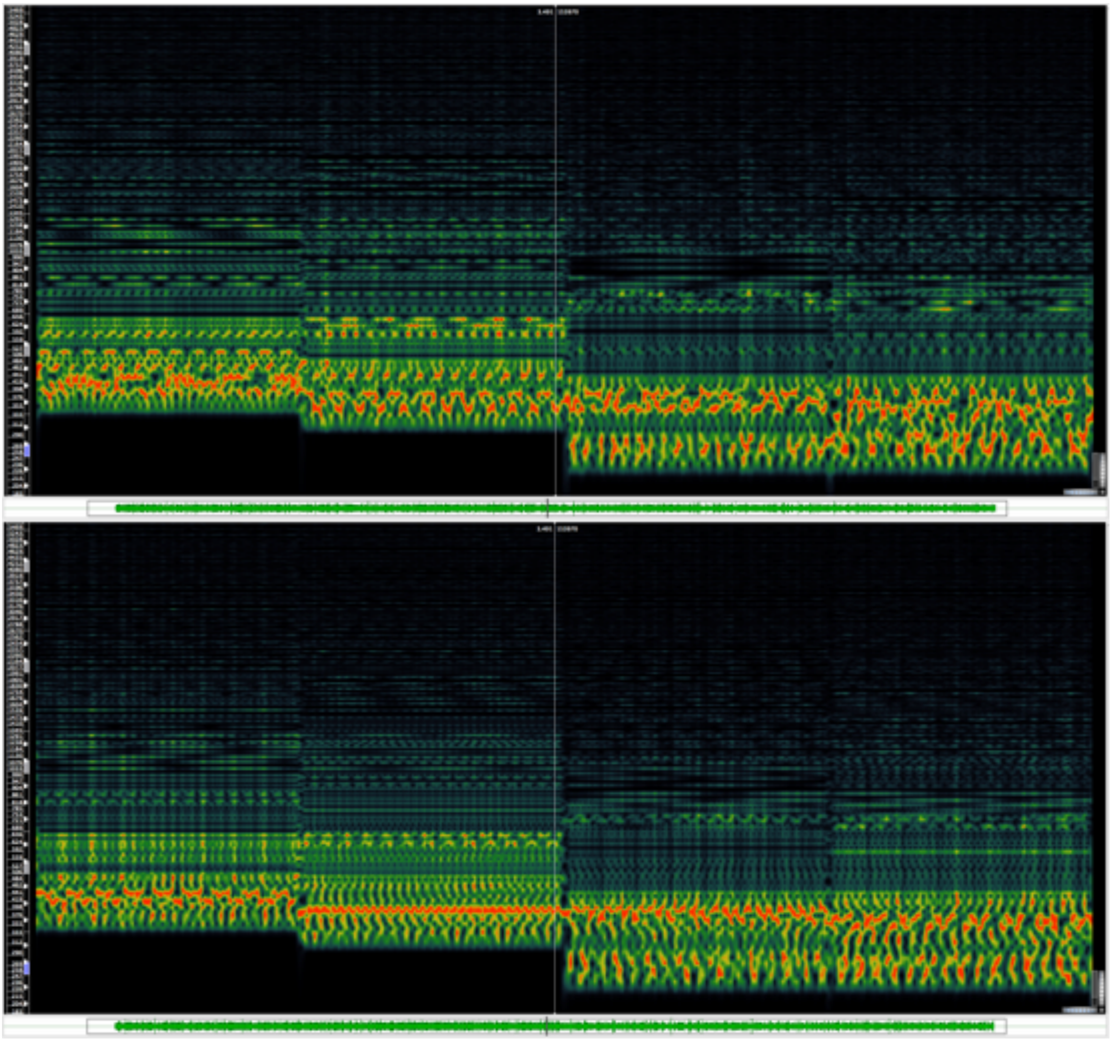


Figure.32 – Sonograms for linear detune test with control set to full (top is original JP-8080, bottom is modified Reaktor example).

In conclusion, it can be observed that although before at three-quarters detune the original example showed a much more conserved spread of frequencies, this has suddenly become more intense. This outcome is due to the sharp ramp upwards in the curve towards the end of the detune control. The earlier tests with lower detune show the controls ability to gradually increase the spread of the waves through much finer increments, allowing for the creation of very smooth sounding timbres, whereas the linear control only gives the option of having a very rough and wide spread of frequencies.

10. ADDITIONAL LISTENING TEST

10.1 Test Design

Results from initial analysis seemed to show that the most important element is the detune curve as this test showed the largest differences. In an attempt to clarify this observation further, an additional listening test was conducted with a selection of experienced subjects. The ITU (1998) categorises experienced subjects as being able to describe auditory events in detail, able to describe subjective impressions and having previous experience in subjective listening but do not regularly conduct subjective evaluations. The test format was conducted with the exact same conditions and equipment as the personal subjective test, detailed in chapter 4.2, with a total of eleven subjects having participated.

Subjects took part in two tests, detailed below. Subjects were asked to compare audio from the emulations against the original sources and evaluate the difference in roughness and dirtiness of A to B using the same grading scale (Appendix 13.8). See Appendix 13.9 for additional test setup information.

- Accuracy (Test 1): Testing unmodified emulation at five detune intervals (very high, high, medium, low and very low). Mix Control set to full, [Progression 4].
    - Dependant Variable: Perception of roughness and dirtiness

- Detune (Test 2): Testing modified emulation with linear detune amount at five detune intervals (very high, high, medium, low and very low). Mix Control set to full, [Progression 3].
    - Dependant Variable: Perception of roughness and dirtiness

10.2 Hypothesis

Perhaps the most important part of a scientific experiment is the formulation of a hypothesis, initial conditions and the testable statement. These determine the scientific quality of the results and statistical analysis of the data depends on the form of the testable statement (Bech and Zacharov, 2006). The subsequent criteria were formulated below:

- Hypothesis: The perception of roughness will be greater using the linear detune rather than the curve
- Initial Conditions: Two tests are conducted at different detune amounts. One using the original polynomial curve, with the other using a line
- Testable Statement: The curved detune control is the most important element that contributes to the Super Saw's sound

10.3 Results

Accuracy Test (Table.13).

Accuracy Test	**1A**	**1B**	**1C**	**1D**	**1E**
Amount of Detune (%)	**0**	**25**	**50**	**75**	**100**
Subject 1	1	2	1	-2	3
Subject 2	-1	2	0	-1	3
Subject 3	-1	2	-1	1	-1
Subject 4	1	2	1	0	0
Subject 5	1	2	1	-1	1
Subject 6	-1	2	0	0	1
Subject 7	-2	1	-1	0	1
Subject 8	-1	2	0	-2	2
Subject 9	0	2	-1	-1	1
Subject 10	-1	1	0	-1	2
Subject 11	-1	2	0	1	2

Table.13 - Results from Accuracy Test (Test 1).

Detune Test (Table.14).

Detune Test **Amount of Detune (%)**	**2A** **0**	**2B** **25**	**2C** **50**	**2D** **75**	**2E** **100**
Subject 1	2	3	3	3	2
Subject 2	-2	3	3	3	2
Subject 3	2	3	2	3	0
Subject 4	3	3	3	2	0
Subject 5	2	3	3	2	0
Subject 6	-2	2	3	3	0
Subject 7	-3	2	3	3	1
Subject 8	-2	2	3	3	1
Subject 9	-2	3	3	3	2
Subject 10	-1	2	3	3	1
Subject 11	-3	2	3	3	-1

Table.14 - Results from Detune Test (Test 2).

10.4 Statistical Analysis and Discussion

To correctly demonstrate the truth of a hypothesis through statistical analysis Kranzler (2003) explains that a null hypothesis (H_0) should be generated, as it is almost impossible with most statistical techniques to demonstrate that something is true. If the probability (p) of this is < 0.05 then the null hypothesis is rejected demonstrating the original hypothesis to be indeed true (Clarke and Cook, 1998). In this case:

- $H_0 =$ Using a linear detune amount, the perception of roughness will be equal to or less than using a curve

In accordance with the ITU-R BS.1284-1 test specification, the presentation of mean values can provide a sufficient overview of the data. Along with this the inclusion of confidence intervals and error bar graphs, with the significance levels stated, are concluded as acceptable forms of data analysis. As the grading scale used intermediate anchor points for evaluation normalisation of results was not applicable. See Table.15 and 16 for analysis.

Accuracy Test **Amount of Detune (%)**	**1A** **0**	**1B** **25**	**1C** **50**	**1D** **75**	**1E** **100**
Mean	-0.45	1.82	0.00	-0.55	1.36
Standard Deviation	1.03573	0.40452	0.7746	1.03573	1.20605
Standard Error	0.31228	0.12197	0.23355	0.31228	0.36364
Confidence ±	0.61206	0.23905	0.45775	0.61206	0.71271

Table.15 – Statistical Analysis of Accuracy Test (Test 1).

Detune Test **Amount of Detune (%)**	**2A** **0**	**2B** **25**	**2C** **50**	**2D** **75**	**2E** **100**
Mean	-0.55	2.55	2.91	2.82	0.73
Standard Deviation	2.29624	0.52223	0.30151	0.40452	1.00905
Standard Error	0.69234	0.15746	0.09091	0.12197	0.30424
Confidence ±	1.35697	0.30861	0.17818	0.23905	0.5963

Table.16 – Statistical Analysis of Detune Test (Test 2).

The arithmetic mean for the accuracy test seems to suggest that listeners did seem to perceive slight differences in sound with test 1B showing the most significant difference suggesting people found that particular example to be rougher. It is worth noting that although listeners perceived test 1E to have a slightly rougher sound, this was to be expected, as a similar result was heard in critical listening conducted by the author. The mean results from the detune test suggest that there is indeed a much bigger difference in sound, with test 2B, 2C and 2D all showing that subjects found that using a linear control created a much rougher sound. Again, it is worth highlighting that there should not have been a big difference in sound perceived for test 2A and 2E due to the same boundaries used for the control as described in Chapter 2.5.3. It can also be concluded that the likelihood that subjects will find a much bigger difference in sound with tests B, C and D as the standard deviation is much lower in comparison to tests A and B.

With the standard error of the estimate, it is possible to report the accuracy of predictions made Kranzler (2003). This measure estimates the variance of answers based on the data, with smaller values proving more accurate. Figure.33 shows the standard error graph plot of mean values with the standard error bars for both tests. From this it can be concluded that tests B, C and D show that there is a significant difference in sound with the linear detune control stimuli perceived to be rougher than the original due to the non-overlapping error bars, none of which have a standard error > .32 which provides an accuracy level of more than two-thirds. Tests A and B shows that listeners perceived these to be relatively similar in sound which is also correct.

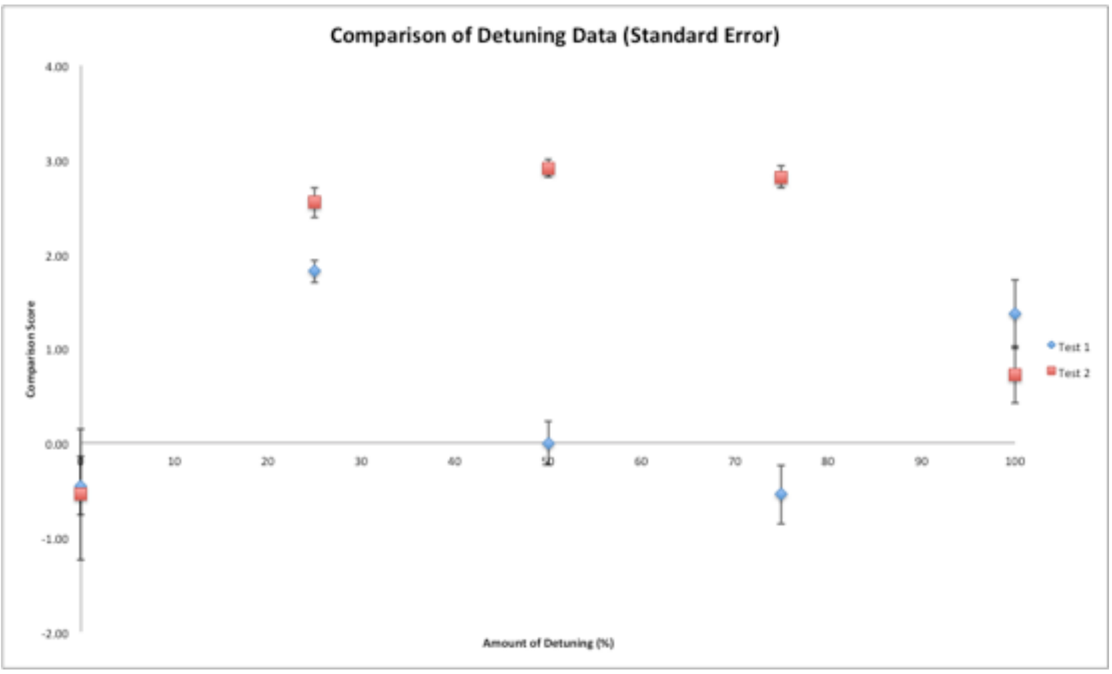


Figure.33 – Graph plot for comparison standard error.

Confidence intervals for tests were calculated as suggested in the ITU-R BS.1284-1 test specification with a significance level of 0.05, of which they are used to estimate the probability of an outcome. Figure.34 shows the graph plot of both tests mean values with their 95% confidence intervals. From here it can again be concluded that subjects perceived stimuli in tests B, C and D with the linear detune control to be rougher than the original due to the non-overlapping confidence limits. This means that $p < 0.05$ and the null hypothesis can be rejected. Again, tests A and B correctly show an overlap due to the boundaries of the detune control.

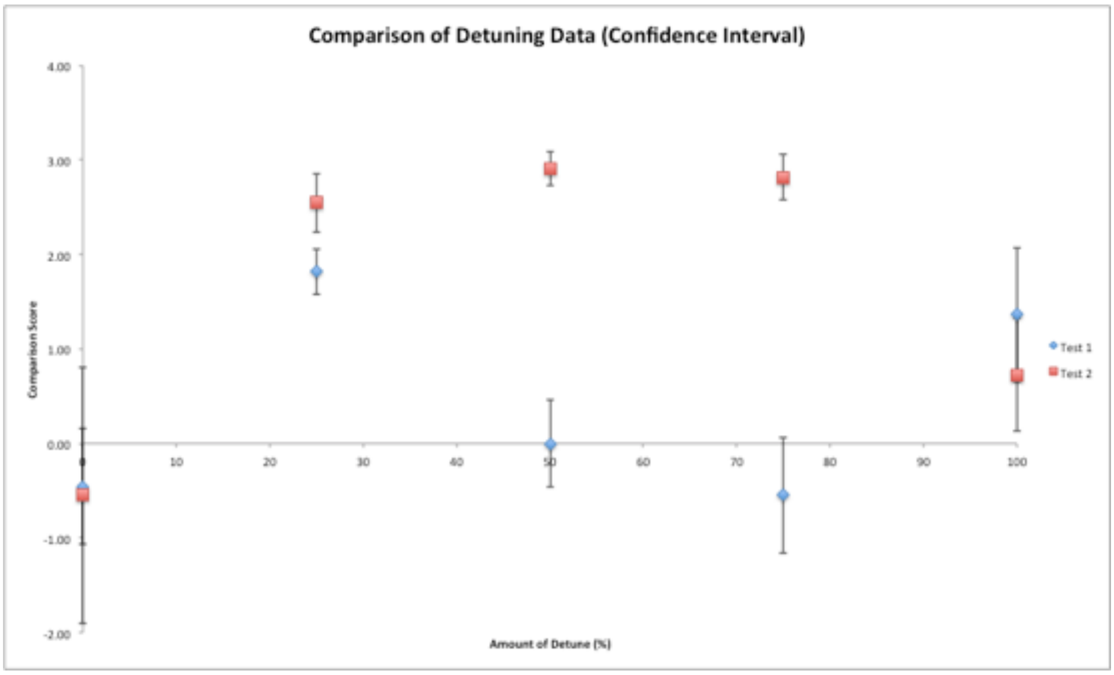


Figure.34 – Graph plot for comparison 95% confidence interval.

11. SUMMARY AND CONCLUSIONS

11.1 Conclusions

- Super Saw oscillator is commonly used for synthesising trance pads
- Super Saw oscillator has been thoroughly researched allowing for a greater understanding of its parameters
- Oscillator exhibits interesting characteristics within parameters of detune, mix, phase and shape
- Effective emulation of the oscillator was constructed within Reaktor based on measurements and research conducted by the author
- Both subjective and objective test methods were researched and conducted in accordance with existing specifications
- Initial observations demonstrated most noticeable sound difference in detune control curve through a series of critical listening and objective measurements
- Additional listening test with expert subjects produced similar results showing considerable difference in sound between linear and curved control
- Project results show the detune curve appears to be the most important element contributing to the Super Saw sound
- Curve allows for subtle detune amounts providing the ability to create unique textures and smooth sounding pads

11.2 Further Work

- Greater understanding of statistical data analysis techniques leading to more conclusive findings. Original analysis limited by the author's knowledge of the subject
- Advanced analysis of original waveform looking into possible effects of THD.
- Increase accuracy of emulations further with inclusion of oscillator phase and additional elements
- Additional methods of objective measurements to increase validity of results.
- Further subjective tests conducted on other elements with a greater number of test subjects
- Listening tests conducted with audio within the context of a track.
- Listening tests using ABX method

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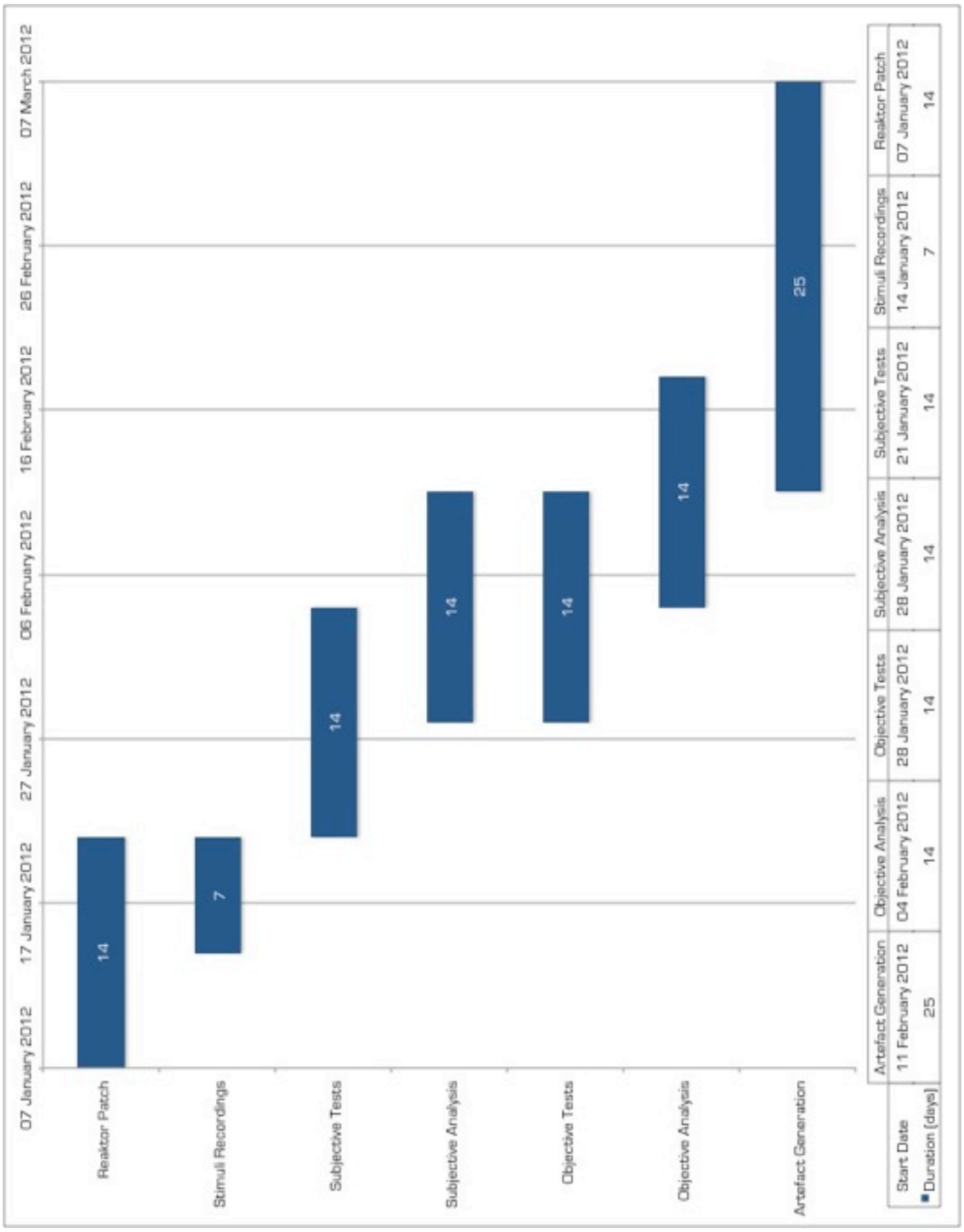
13. APPENDIX

13.1 Gannt Chart Plan



	Reaktor Patch	Stimuli Recordings	Subjective Tests	Subjective Analysis	Objective Tests	Objective Analysis	Artefact Generation
Start Date	07 January 2012	14 January 2012	21 January 2012	28 January 2012	28 January 2012	04 February 2012	11 February 2012
■ Duration (days)	14	7	14	14	14	14	25

13.2 ITU-R Five-Grade Impairment Scale

ITU-R five-grade impairment scale for subjective assessment of sound quality (after International Telecommunication Union, 1997).

Impairment	**Grade**
Imperceptible	5.0
Perceptible, but not annoying	4.0
Slightly annoying	3.0
Annoying	2.0
Very annoying	1.0

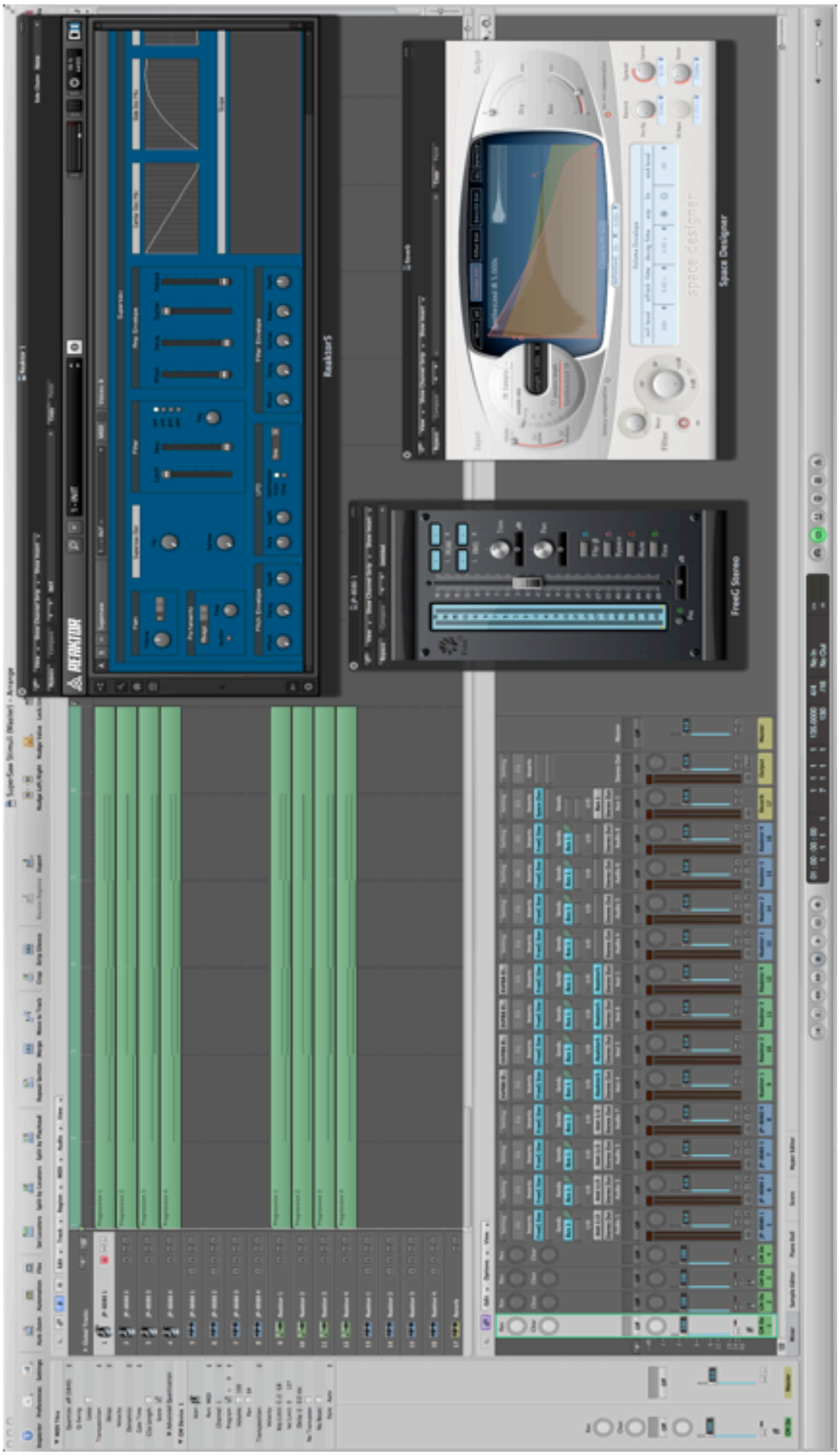
13.3 ITU-R Seven-Grade Impairment Scale

ITU-R seven-grade comparison scale for subjective comparison methods (after International Telecommunication Union, 2003).

Comparison	**Grade**
Much better	3
Better	2
Slightly better	1
The same	0
Slightly worse	-1
Worse	-2
Much worse	-3

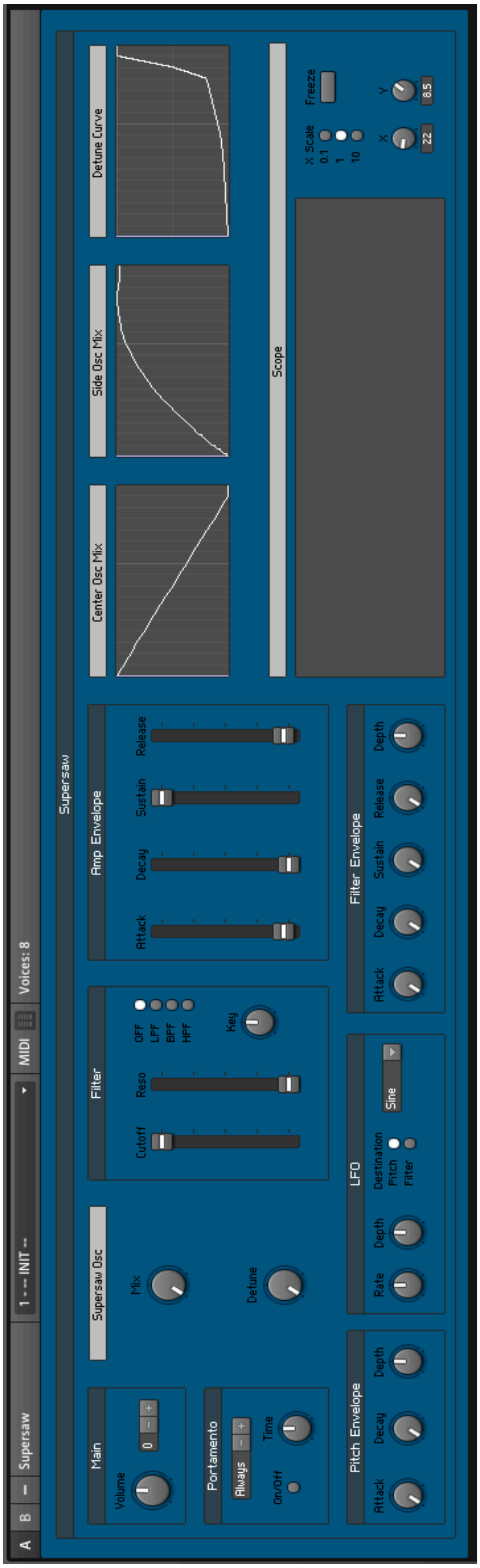
13.4 Additional Testing Setup Information for Source Recordings

13.4.1 Master Logic project used for recording audio samples



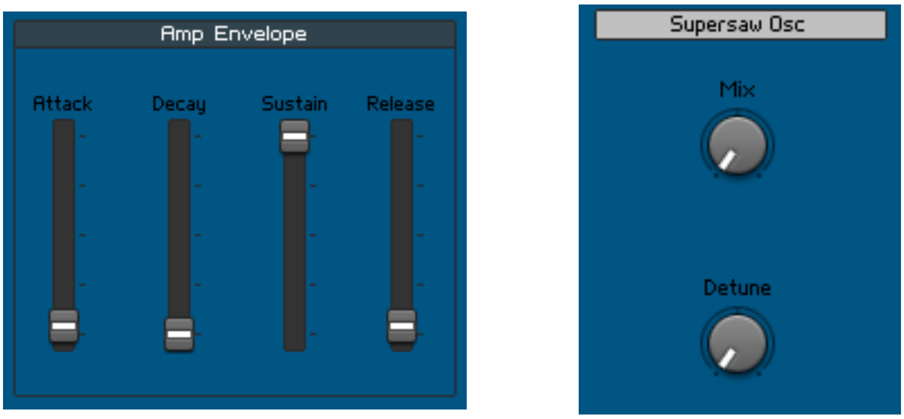
13.4.2 Initial settings for JP-8080 recordings

13.4.3 Initial settings for Reaktor patch recordings



13.4.4 Initial ADSR and oscillator settings for JP-8080 recordings

13.4.5 Initial ADSR and oscillator settings for Reaktor patch recordings



13.4.6 Apogee Duet 2 audio interface recording settings

Sample Rate 44.1 kHz
Clock Source Internal
System Status Ready

ANALOG	Inst 1	Inst 2
Analog Level	Inst	Inst
Soft Limit	Off	Off
	0	0

13.4.7 Detune settings for JP-8080 recordings

Very Low -

Low -

Medium -

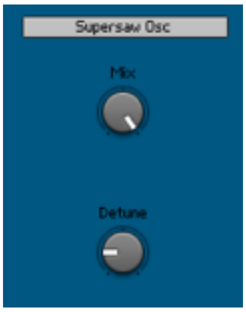
High -

Very High -

13.4.8 Detune settings for Reaktor recordings

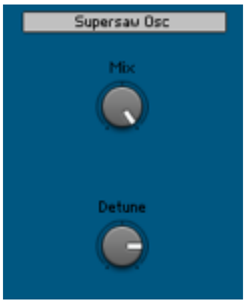
Very Low -

Low -

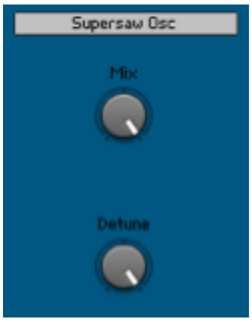


Medium -

High -



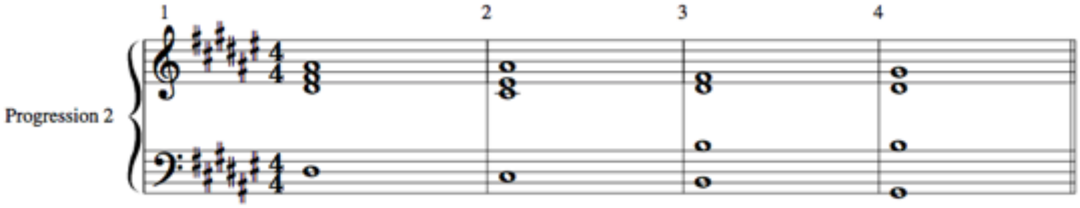
Very High -



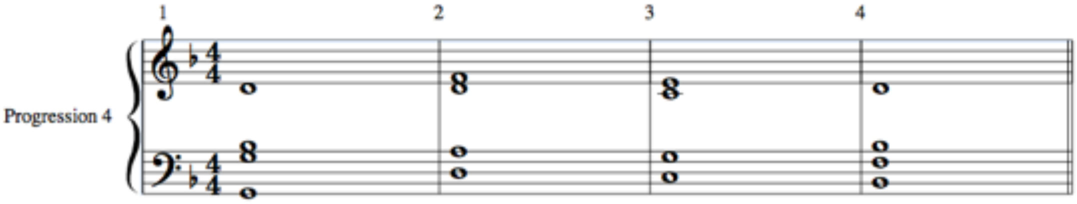
13.5 Test Chord Progressions

13.5.1 Progression one notation used for oscillator test samples.

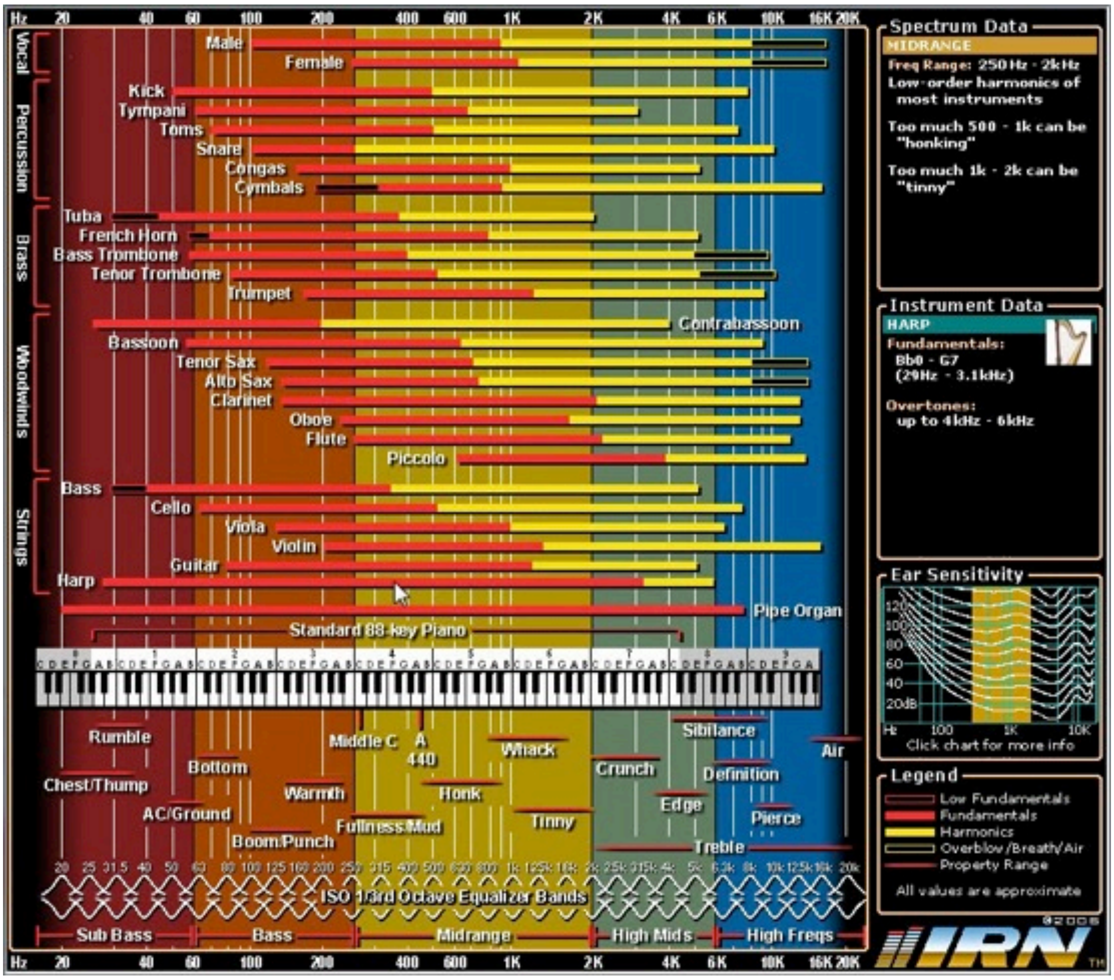
13.5.2 Progression two notation used for filter test samples.



13.5.3 Progression three notation used for detune test samples.

13.5.4 Progression four notation used for accuracy test samples.

13.6 Equalisation Guides

13.6.1 EQ guide from Independent Recording Network (IRN, 2013)



MIDRANGE

Freq Range: 250 Hz - 2kHz

Low-order harmonics of most instruments

Too much 500 - 1k can be "honking"

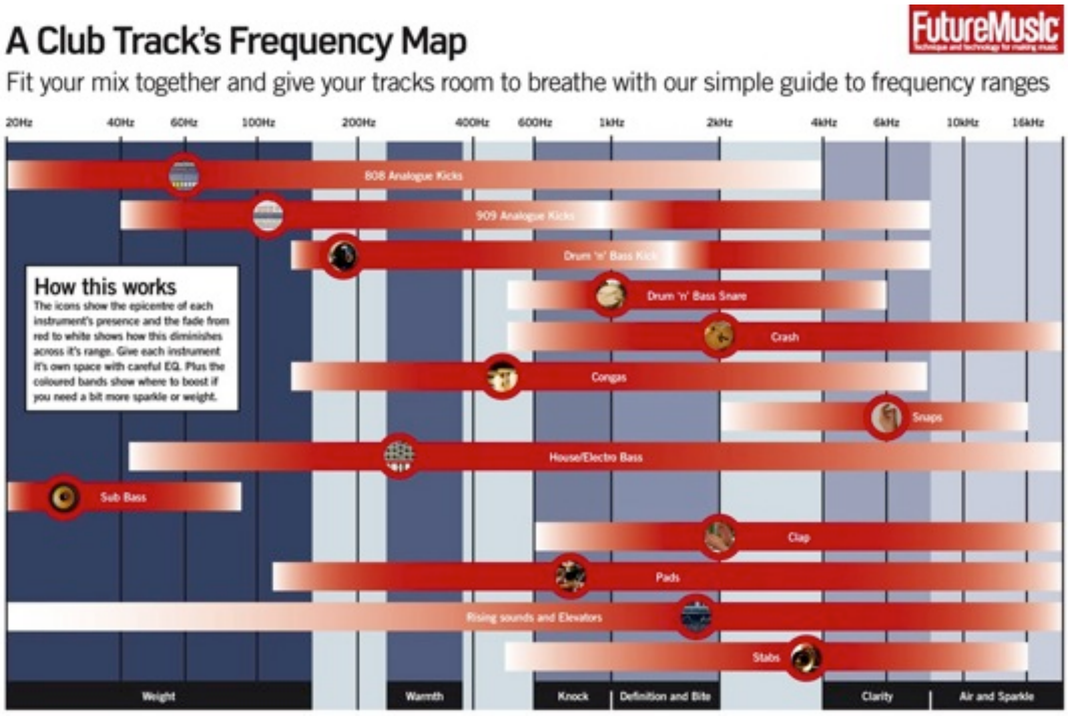
Too much 1k - 2k can be "tinny"

HARP

Fundamentals: Bb0 - G7 (29Hz - 3.1kHz)

Overtones: up to 4kHz - 6kHz

13.6.2 EQ guide from Music Radar (Baker, 2009)

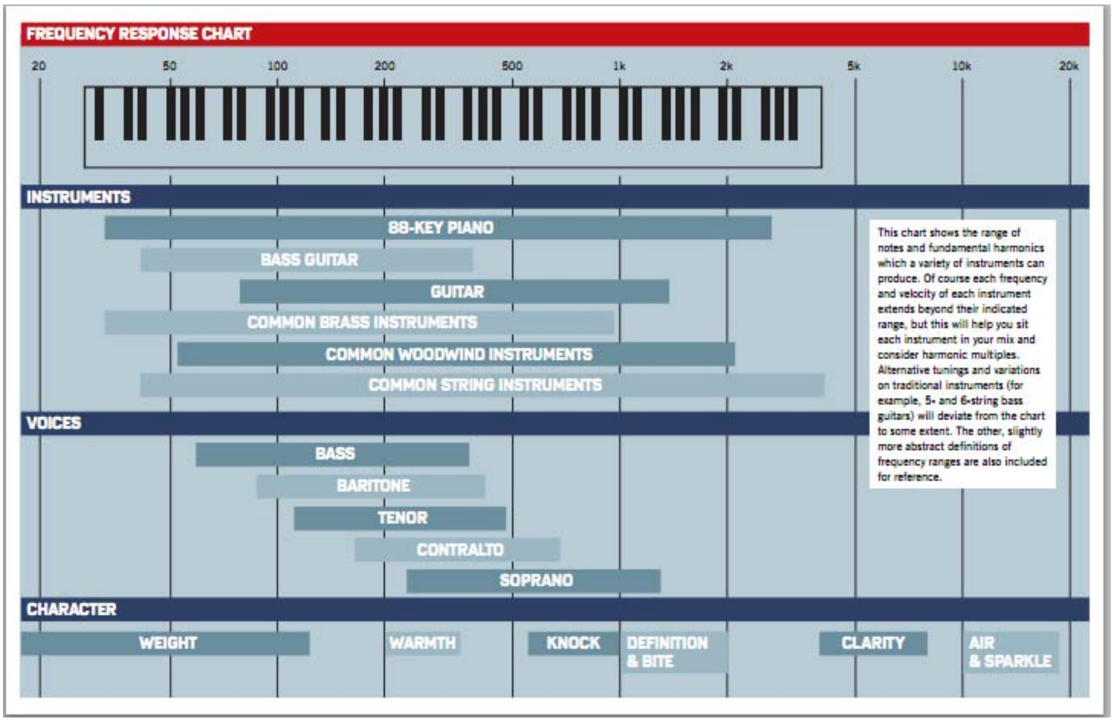


A Club Track's Frequency Map

Fit your mix together and give your tracks room to breathe with our simple guide to frequency ranges

How this works
The icons show the epicentre of each instrument's presence and the fade from red to white shows how this diminishes across it's range. Give each instrument it's own space with careful EQ. Plus the coloured bands show where to boost if you need a bit more sparkle or weight.

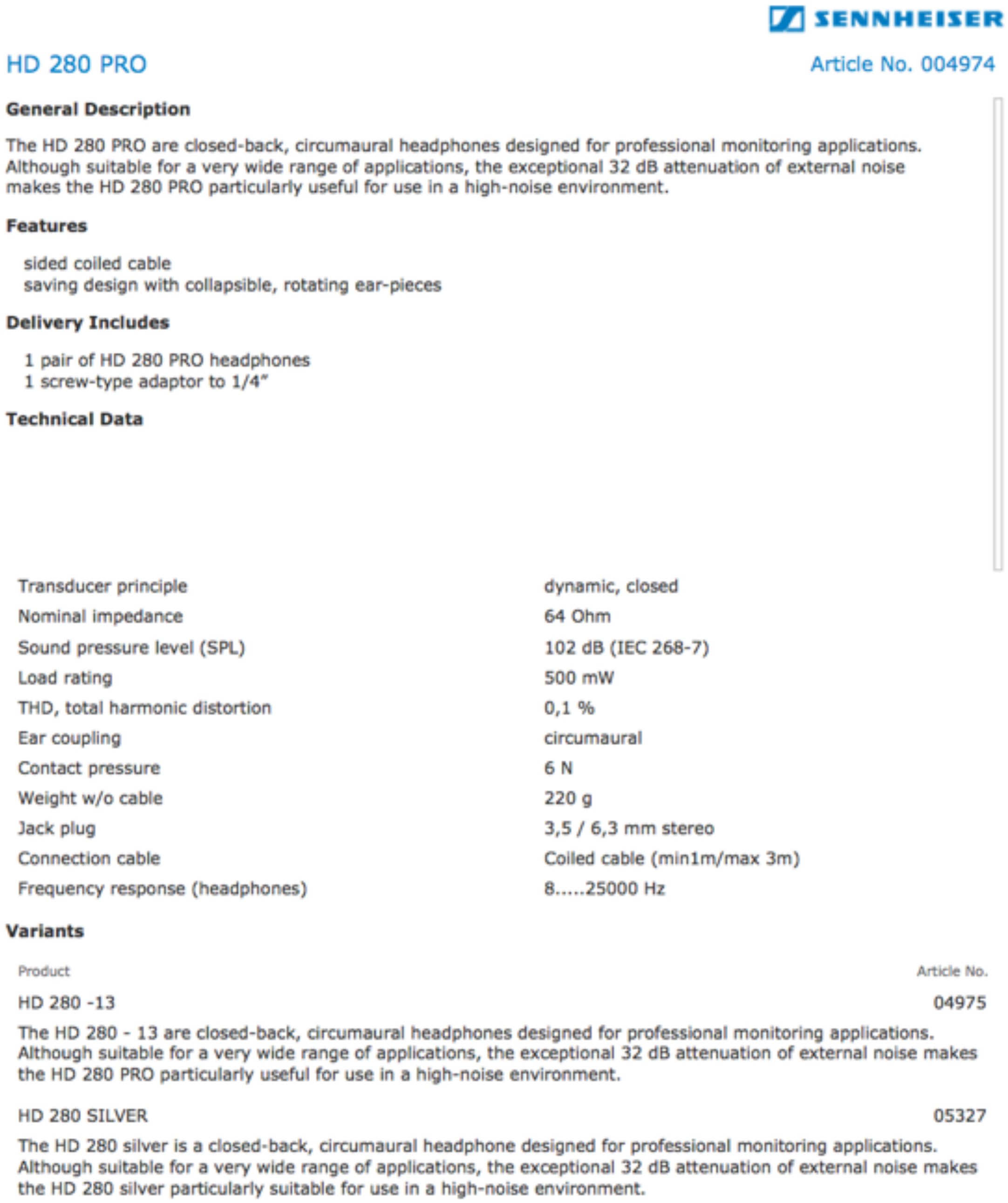
13.6.3 EQ guide from Stereoklang (Stereoklang Productions, 2013)



FREQUENCY RESPONSE CHART

This chart shows the range of notes and fundamental harmonics which a variety of instruments can produce. Of course each frequency and velocity of each instrument extends beyond their indicated range, but this will help you sit each instrument in your mix and consider harmonic multiples. Alternative tunings and variations on traditional instruments (for example, 5- and 6-string bass guitars) will deviate from the chart to some extent. The other, slightly more abstract definitions of frequency ranges are also included for reference.

13.7 Sennheiser HD 280 PRO Technical Specifications



SENNHEISER

HD 280 PRO

Article No. 004974

General Description

The HD 280 PRO are closed-back, circumaural headphones designed for professional monitoring applications. Although suitable for a very wide range of applications, the exceptional 32 dB attenuation of external noise makes the HD 280 PRO particularly useful for use in a high-noise environment.

Features

sided coiled cable
saving design with collapsible, rotating ear-pieces

Delivery Includes

1 pair of HD 280 PRO headphones
1 screw-type adaptor to 1/4"

Technical Data

Transducer principle	dynamic, closed
Nominal impedance	64 Ohm
Sound pressure level (SPL)	102 dB (IEC 268-7)
Load rating	500 mW
THD, total harmonic distortion	0,1 %
Ear coupling	circumaural
Contact pressure	6 N
Weight w/o cable	220 g
Jack plug	3,5 / 6,3 mm stereo
Connection cable	Coiled cable (min1m/max 3m)
Frequency response (headphones)	8.....25000 Hz

Variants

Product	Article No.
HD 280 -13	04975

The HD 280 - 13 are closed-back, circumaural headphones designed for professional monitoring applications. Although suitable for a very wide range of applications, the exceptional 32 dB attenuation of external noise makes the HD 280 PRO particularly useful for use in a high-noise environment.

HD 280 SILVER	05327

The HD 280 silver is a closed-back, circumaural headphone designed for professional monitoring applications. Although suitable for a very wide range of applications, the exceptional 32 dB attenuation of external noise makes the HD 280 silver particularly suitable for use in a high-noise environment.

13.8 Listening Test Grading Scales

- Detune: Measure of texture (concluding of roughness and dirtiness)

Comparison	Grade
Much Rougher	3
Rougher	2
Slightly Rougher	1
The same	0
Slightly Smoother	-1
Smoother	-2
Much Smoother	-3

- Osc (Mix): Measure of mass (concluding of thickness and fullness)

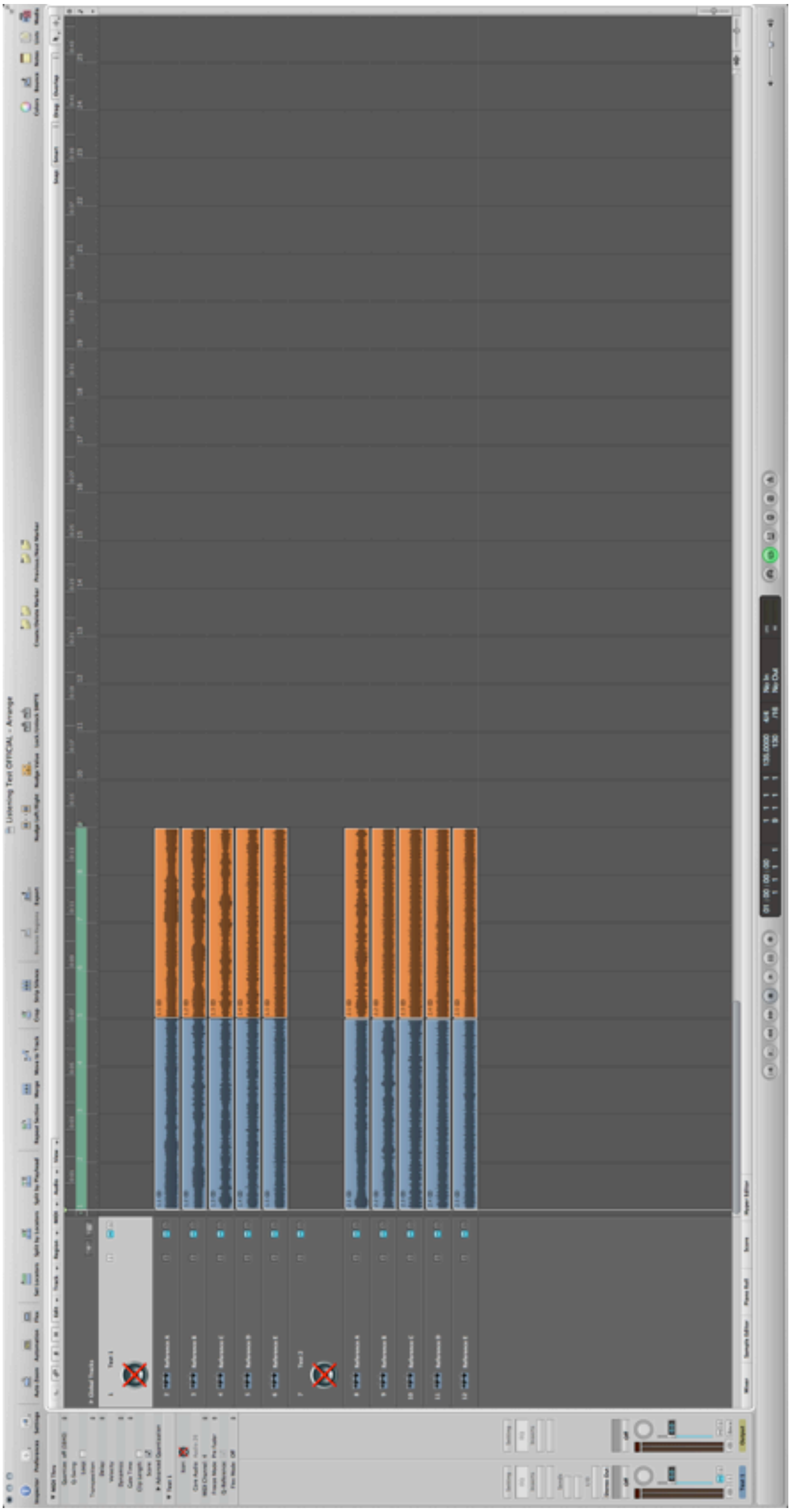
Comparison	Grade
Much Thicker	3
Thicker	2
Slightly Thicker	1
The same	0
Slightly Thinner	-1
Thinner	-2
Much Thinner	-3

- Filter (Shape): Measure of luminance (concluding of brilliance and sharpness)

Comparison	Grade
Much Sharper	3
Sharper	2
Slightly Sharper	1
The same	0
Slightly Duller	-1
Duller	-2
Much Duller	-3

13.9 Additional Testing Setup Information for Additional Listening Test

13.9.1 *Master Logic project used for test*



13.9.2 Test Instructions

Instructions

Compare the two references.

Using the scale below evaluate the texture of A (blue) compared to B (orange).

Texture = roughness/dirtiness

Thank you for taking part.

Grading Scale

Orange regions is -	**Grade**
Much Rougher	3
Rougher	2
Slightly Rougher	1
The same	0
Slightly Smoother	-1
Smoother	-2
Much Smoother	-3