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Bacheloroppgave

Lage en lydsyntese

Creating a sound synthesis

Musikkproduksjon

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Preface

Following a long term personal interest in synthesizers, a module, Sound Synthesis and MIDI, was undertaken during my exchange year at Staffordshire University in the UK to explore the subject further. In this module, the class learned about sound synthesis using Native Instruments Reaktor software, which was used to construct synthesisers using a variety of different synthesis types. I found this subject to be very interesting and decided to base my bachelor on the different studies made in that module.

I want to thank everyone that helped me with this bachelor. A special thanks to my former tutor in Sound Synthesis and MIDI, Bob Lewis. A big thanks to Ola Haampland for the support he gave me during the entire process of writing from start to finish. Last but not least a big thanks to the amazing ladies working in the library at Høgskolen i Hedmark, Campus Rena.

The task has a background in music production course at Hedmark University College, Campus Rena, Department for economics and management.

Ole Anders Danielsen

Rena, May 2014.

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Attachment 1: CD containing Reaktor ensemble, pictures of the synthesizer, and Logic project for snapshots.

Sammendrag

Denne bacheloroppgaven tar for seg *subtractive* syntese og *frequency modulation* syntese. Mitt mål var å vise hvordan disse to lydsyntesene fungerer og hvordan man kan lage en synthesizer bestående av disse to syntesene i Native Instruments Reaktor.

Den teoretiske delen tar for seg oppdagelsen og utviklingen av de to syntesetypene og forklarer hvordan de fungerer. Til sammen utgjør dette bakgrunnsstoffet grunnlaget for gjennomføringen av den praktiske delen.

Den praktiske gjennomføringen viser hvordan man lager de to lydsyntesene via bilder og tekst, og hvordan de kobles innad i synthesizeren for å gi lyd. Igjennom forhåndslagde lyder vises det også hvordan de to lydsyntesene kan fungerer i sammen og hvilke bruksområde som passer best for den individuelle syntesen.

Abstract

This bachelor assignment examines *subtractive* synthesis *and frequency modulation* synthesis. My goal was to show how these two types of sound synthesis works and how one can create a synthesizer consisting of these synthesis in Native Instruments Reaktor.

The theoretical part examines the discovery and development of the two synthesis types and explain how they work. Together, this background material creates the foundation for executing the practical part.

The practical completion shows how one can create the two types of sound synthesis through pictures and text, and how they are connected inside the synthesizer to generate sound. Through pre-made sounds it also shows how the two synthesis types can work together and what area of use best suits the individual type of synthesis.

Chapter 1: Introduction

1.1 Introduction:

The era of computer-music has given the possibilities for manufactures and business companies to digitalize their products to reach an even bigger amount of people to buy and use their products, and in the 21st century a great amount of music genres use software synthesized sounds in their productions. Producers and artists spend vast amounts of time trying to create the perfect sound and create something new for the music market and the sounds used are getting more obscure, more specialised and more complex. The use of more powerful synthesizers, analogue and digital, with endless effects and routing capabilities combined with a simple *hands on* interface makes it easy and fun to experiment with sounds and more rewarding trying to create new "special" sounds. What is interesting is that a lot of people prefer the use of software versions rather than hardware versions. There are of course several reasons for this such as costs and space the equipment takes in a room, but most importantly the quality between the two versions are not significantly different anymore. Software synths are getting more powerful as time passes and people develop new products with new attributes and this gives the customers that use these products even more possibilities when it comes to the creation of sounds. Some products actually allow you to not only create a sound in a synthesizer, but construct the synthesizer itself. For this paper I shall use such a tool to create two types of synthesis into a single synthesizer and explain how the synthesis methods work and how you can construct them.

1.2 Motivation:

There are large numbers of synthesizers, both software and hardware, which features different types of synthesis. These synthesizers can be bought from companies and distributors, and usually costs a lot of money. The price you pay for this synthesizer defines what you get for it. With that I mean you pay what is costs and that is all you get. So my question is why not just create the synths yourself?
Reaktor is the tool I have used for this assignment and it can create basically any synthesizer you could wish for. The possibilities are endless with this plug-in and when you get tired of the synthesizer you've made, you can just add new synthesis methods in it and expand the synths capabilities even further.

1.3 Thesis:

The objective of this thesis is to construct a synthesizer in Reaktor featuring subtractive and frequency modulation synthesis.

Chapter 2: Theory

2.1.1 What is synthesis?

"´Synthesis´ is defined in the 2003 edition of the Chambers 21st Century Dictionary as ´building up; putting together; making a whole out of parts´" (Russ, 2009. Pp. 3).

Martin Russ continues with defining synthesis as being frequently used in two major contexts: production of electronic sounds and the creation of chemical compounds (Russ, 2009).

Even though synthesizers have a similar concept, bringing something together as a whole, there are many different types of synthesizers not only described as "musical synthesizers". To mention some, there are video synthesizers used for processing and producing video signals, there are texture synthesizers used in the graphics industry and of course the sound synthesizers used in music for programming and creating sounds (Russ, 2009).

All the types of different synthesizers share two fundamental building blocks: a control surface where a person define the values of different parameters and a "synthesis engine" that interprets these values and produces the output. Russ Martin (2009, pp. 4) also mentions an important aspect of the differences between the "engine" itself and the control interface. That the level of complexity in the synthesis process is usually very high so the need of a more control friendly interface is needed. E.g. if you want to programme a single note in a DAW[1] using for example Nexus[2], it would be easier to just programme the C-note in a "piano roll" than to define the value of the note in hertz. It would also be difficult and time-consuming to connect a low-pass filter to the right parameters to be able to use it rather than just selecting it in a scroll-down menu inside the control interface. By doing these things easier the consumers do not need a deep level of knowledge of how the equipment works to use it and this makes synthesis more fun than hard work.

[1] Digital Audio Workstation
[2] Nexus is a Software synthesizer

2.1.2 Sound synthesis:

“Sound synthesis is the process of producing sound“ (Russ, 2009. Pp. 4).

There are many types of synthesis that are utilized in the musical world. The **granular** synthesis-model for example uses the same principle as sampling, but instead of taking a sample and reuse 2 seconds of it as a whole, granular synthesis relies on taking very short parts called *grains*, and their duration varies between 10-100ms (Russ, 2009, pp. 294).

Additive synthesis is a model where multiple sine-waves are put together to create different timbres. Each sine-wave plays at different amplitude and frequency to create overtones and harmonics. The principle of this synthesis model is that it can create any sound, harmonically rich or not, with layers of sine-waves (Reid, 2000, chapter 1.).

We also have **Amplitude Modulation**(AM) synthesis, **Ring Modulation**(RM) synthesis, **Sample replay**, **S&S**, **Wavetable** and **Physical Modelling** synthesis, but as for this paper there are two types of sound synthesis we want to take a closer look at: **Subtractive** synthesis and **Frequency Modulation** synthesis.

2.1.3 Frequency Modulation synthesis (FM):

“FM is a digital form of synthesis that can create complex sounds and aural elements with a very small number of fundamentals involved” (Prager, 2004, pp. 329).

FM synthesis was discovered and developed by John Chowning in 1973. His work was based on data collected and analysed by Max Mathews and Jean Claude Risset at Bell Labs. Mathews and Risset analysed frequencies from acoustic instruments and then tried to re-synthesize it with computers, using the data they had collected. The FM synthesis was meant to emulate acoustic sounds, for example the “Bell” which is one of the classic examples of FM synthesis (Prager, 2004).

As mentioned by Prager, FM started to lose its popularity in the late 80s, but because of the virtual instrument era it soon became a welcomed and integrated component in electronic music (Prager, 2004).

FM is a digital form of synthesis, but it can be compared in ways to the subtractive synthesis. A subtractive synthesis relies on stringing a set of modules together to generate a signal.

An example explained by Prager in "*Sampling and Soft Synth Power*":

> 1. You began with the oscillator by selecting a waveform.
>
> 2. That signal would then be routed to a filter to manipulate the frequencies of the oscillator.
>
> 3. The output of the manipulated signal could then be patched to an envelope that could adjust the attack, decay, sustain, and release of the filter.
>
> 4. The output of the envelope would then be routed to an amplifier with an envelope that would adjust the attack, decay, sustain, and release of the amplitude.
>
> 5. The resulting signal would then be routed to a hardware mixer or external amplifier (2004, pp. 282).

A FM model that is most basic is built upon two fundamentals:

-The Carrier(C) – This is the waveform that has modulation applied to it by the modulator.

-The modulator(M) - This is the modulating/manipulating frequency in FM. "It is routed into the input of the carrier frequency in order to merge them into a single composite waveform that contains several additional frequencies i.e. frequency deviation" (Prager, 2004, pp. 330).

The modulator is a subsonic low-frequency oscillator which is used to modulate the carrier which is the oscillator in audible frequency range. The audible oscillator(C) is modulated by the subsonic low-frequency oscillator(M) to change the pitch. "A typical result was a slightly rising and falling periodic pitch variation (vibrato) in the carrier" (Holmes, 2012, pp. 333).

Holmes continues,

> Using a computer to simplify the management of complex frequency modulation, Chowning found that, when the modulating frequency entered the audio range (upwards from about 20Hz), the rising and falling of the pitch of the carrier was no longer heard as vibrato, but rather as a complex change in its spectrum (Holmes, 2012, pp. 333-334).

He continues to say that this technique is called *non-linear* because it produces a wide spectrum of overtones from the modification of few inputs and parameters. FM also produces *sidebands* around the carrier frequency, which can also be seen upon as harmonics. The amount of harmonic content is determined by the ratio of the C, M and the depth of modulation. “In practice, this means that the timbre of one oscillator could be controlled by the another, and that both could be managed using envelope generators to dynamically modify the timbre and shape of a note” (Holmes, 2012, pp. 334).

2.1.4 Subtractive synthesis:

According to Jussi Pekonen and Vesa Välimäki (2011, pp. 1) most of the early music synthesizers in the 1960s and 70s were based on analogue electronics used in subtractive synthesis.
Later on when arriving to the 1980s and 1990s there was an increasing interest in the introduced frequency modulation and sampling synthesis, and subtractive synthesis was no longer the most popular sound production principle (Pekonen & Välimäki, 2011).
Even though the different sound presented in the different types of synthesis, in the mid 1990s the “warm” sound of the subtractive synthesis were shown interest by musicians, and to meet the increasing demand the Swedish company Clavia released the *NordLead* synthesizer (1995) using subtractive synthesis for generating sound, but using digital signal processing tools. As Pekonen and Välimäki states, “The NordLead synthesizer was, in fact, the very first digital synthesizer that emulated the sound generation principle of analog synthesizers” (Pekonen and Välimäki, 2011, pp.1).

Subtractive synthesis is a method of synthesis where harmonic content is removed from a source that is harmonically rich. This source can contain waveforms such as sine, triangle, square, pulse and saw-tooth waveforms. The **Sine** wave consists of only a fundamental with no other harmonic content. The Sine wave is often used in synthesis in combination with other waveforms. When used alone, the Sine wave is often used as bass or sub either beneath a kick drum or a bass line layered with say a saw-tooth. When layered, the sine wave can give more depth to a sound and make it sound bigger. A **Triangle** waveform in terms of harmonics contains only the fundamental frequency plus odd-numbered harmonics. The **Pulse** wave has only the fundamental plus odd harmonics. Compared to the sine, saw and triangle, the pulse wave instantaneously jumps from apex to base, and the duty cycles can vary where 1:3 is the typical value. The duty cycle of the Pulse wave determines the harmonic content (Holmes, 2008, pp. 182). **Square** waves have odd harmonic content such as 3rd, 5th etc. These harmonics play back at an amplitude of 1/F, i.e. a frequency of 100Hz has the 3rd harmonic of 300Hz. The **Saw-tooth** contains both even and odd harmonics, and is the waveform with the most harmonic content, giving the advantage of being able to create clear and bright sounding pads, strings and brass. **White noise** are unlike any other waveforms in terms of frequencies and tonality. White noise do not produce any actual tones, but create a random mixture of all frequencies and when listening to white noise on its own it is very "sharp" and rather unpleasant to listen to. **"**White noise contains equal amounts of energy at every frequency and is comparable to radio static.." (Snoman, 2009, pp. 12*).* In other words this means that no matter what key you press on a MIDI-controller, white noise will always play at the same pitch and the frequencies will remain the same. So if you play a C note or a G note the wave will play back exactly the same way. White noise can be used for several things in a song, but it is mostly used for percussive elements, snares, cymbals, hi-hats and more (Snoman, 2009, pp. 10-12).

Subtractive synthesis operates by filtering undesired harmonic content produced by the oscillators. With the use of modifiers such as envelopes and LFOs, this synthesis method can change and shape the timbre of an existing sound into something new.

2.1.5 Oscillators:

> **"**An oscillator of some kind is the sound-producing mechanism in nearly every music-making device, including most Western acoustic orchestral instruments, many from outside the Western tradition, electromechanical instruments, analogue synthesizers, and digital synthesis algorithms" (Bilbao, 2009, pp. 45).

There are many types of oscillators and circuit configurations that produce oscillations. As Guillermo Gonzalez states in the book "*Foundations of Oscillator Circuit Design"*:

> "Some oscillators produce sinusoidal(SO) signals, others produce nonsinusoidal(NSO) signals. NSO oscillators, such as pulse and ramp (or saw-tooth) oscillators, find use in timing and control applications. Pulse oscillators are commonly found in digital-systems clocks, and ramp oscillators are found in the horizontal sweep circuit of oscilloscopes and television sets" (Gonzalez, 2006, pp. 1).

Gonzalez continues to mention that SO oscillators are used in many consumer electronic equipment such as radios, TVs and VCRs.

As for this paper we will take a look at the oscillators used in music. There are many types of oscillators such as: Voltage Controlled Oscillator(VCO), Digitally Controlled Oscillator(DCO), Low Frequency Oscillator(LFO), Audio Oscillator(AO), and several others used for different applications. For example a VCO is used in a hardware synthesizer such as the Prophet 5 which is fully analogue, and DCOs are used in the prophet 12(P12) which is a hybrid synth. It is called a hybrid because of its use of analogue and digital components, which means that the P12 is a hybrid for its digital oscillators and analogue filters, voltage controlled amplifier and so on. The reason for using different oscillators for different synthesizers may vary. As for the Prophet 12 the reason for choosing DCOs was that they are more stable and will not fall out of tune, which has been seen upon as a problem for many users. The reason for DCOs not falling out of tune is that it’s an analogue oscillator circuit controlled by a digital microchip. This makes the oscillator less likely to overheat and go out of tune (Vintagesynth, n.d., sixth para.). Some argue that this is in fact the “charm” of having a fully analogue synthesizer and it gives the synths character (Snoman, 2009).
For other usage opportunities we have the LFO which can be used as a modulation tool for other parts of a synthesizer, such as filter modulation or pitch modulation (Wiffen, 1997, chapter three).

The oscillator starts to play when a key is pressed on the keyboard. Pressing a key sends a signal to the oscillator alongside with a control voltage(CV). This signal provides the oscillator with a pitch value so it will oscillate at the appropriate frequency. As said by Rick Snoman, writer of “*Dance Music Manual”:* “The CV that is sent is unique to the key that is pressed, allowing the oscillator to determine the pitch it should reproduce” (Snoman, 2009, pp. 9). This means that the CV determines that when you press the C key on a keyboard it plays a C and not a different note. With VCOs this isn’t always the case though. As mentioned above, synthesizers that are fully analogue can sometimes “detune” themselves. That means that when you press a key, the note that are supposed to be played is out of tune, so you have to tune it back in. A lot of people like a slightly detuned oscillator in mix with other sounds played in their songs and let it stay out of tune. The detune “problem” is solved by using digitally controlled oscillators which is more reliable when it comes to tuning. Since they are digital, they will never go out of tune and they are more accurate in their cycles than VCOs.

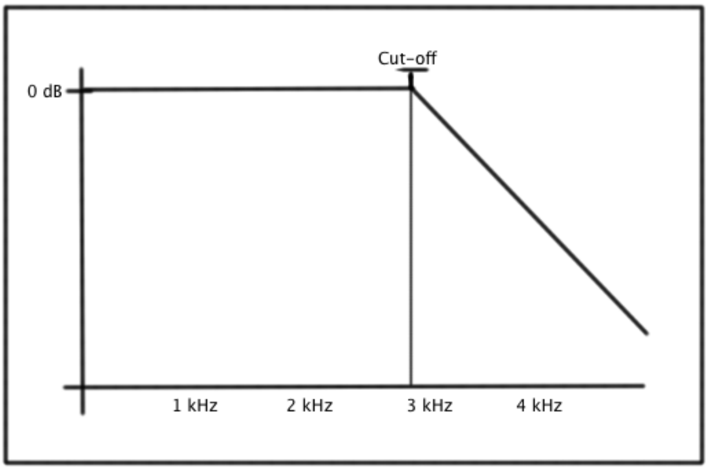
2.2.0 Filters:

> "A filter is an amplifier whose gain changes with frequency. It is usually the convention to have filters whose maximum gain is one, and so it is more correct to say that for a filter, the attenuation changes with frequency.. Filters are powerful modifiers of timbre, because they can change the relative proportions of harmonics in a sound" (Russ, 2009. Pp. 113).

Filters are in some ways equalizers because they both attenuate frequencies of sounds. The difference between them is that equalising is used mostly for modest cuts or boosts of amplitude across the frequency spectrum (Robjohns, 2005, first para.).
The filter is a tool for removing portions of the frequency spectrum, e.g. harmonics from 1kHz and up if that is the chosen setting.
When we look at synthesizers such as Omnisphere by Spectrasonics, Sylenth by LennardDigital or the Prophet 12 by Dave Smith Instruments, we see a number of different filter settings that can be chosen. In fact, most synthesizers have (especially software synthesizers) more than two settings on the filter that can be utilized for filtering.

In subtractive synthesis especially, the filters are an important component that fulfils the criteria of a synth being subtractive. As mentioned earlier in 2.1.4, subtractive synthesizers work by removing something from a sound, e.g. harmonic content from frequencies above or beneath a given cut-off value. There are many types of filters that attenuates frequencies in different ways and we shall take a look at some of these.

2.2.1 Low-pass :



“A low-pass filter has more attenuation as the frequency increases” (Russ, 2009. Pp. 114).

The point where the attenuation first becomes apparent is called the cut-off frequency. On a low-pass filter the frequencies bellow the cut-off point has no alteration because the filter allows the lower frequencies of a sound to pass through. “Above the cut-off frequency, the attenuation increases at a rate which is called a slope” (Russ, 2009. pp. 114). This happens because of the constant change in the frequency spectrum as a sound plays in higher pitch. Say a pianist plays a low note on the piano, there are more lower frequencies on that key than if he/she plays a higher note. So the filter will not attenuate the signal as much when the lower key is played as the filter is set to a cut-off at for example 3kHz, but as the pianist plays higher and higher notes the pitch will increase and change the frequencies, so the filter will attenuate the frequencies above the cut-off. This results in the sounds reaching frequencies above the cut-off to sound “muffled” and “closed”.
Filter designs varies and the slope of the attenuation varies with it. The most basic filters will have one resistor and one capacitor, also known as RC. These filters will have slopes of 6 dB/octave, which means the attenuation increases by 6 dB for each doubling of frequency (Russ, 2009). This is also called a 1-pole low-pass filter. Each pole represents one pair of RC elements and when the slope increases the number of poles increases.

Example:

1-pole = 6 dB/octave attenuation

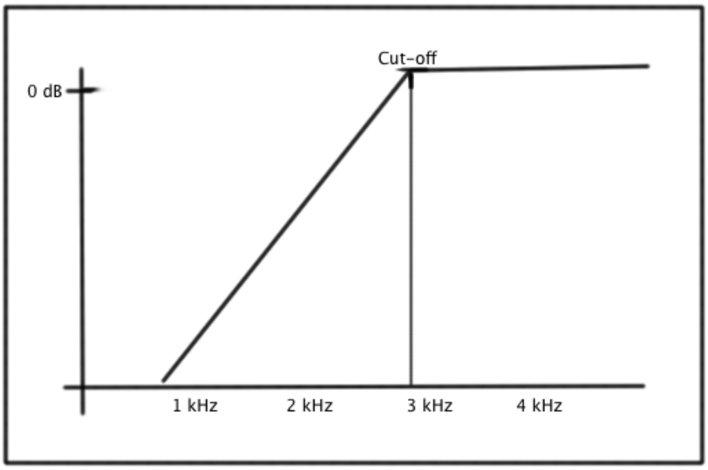
2-pole = 12db/octave attenuation

4-pole = 24 dB/octave attenuation

The more poles the filter uses the more it affects the sound and changes its timbre. It makes the slope more steep and "sharp" so the harmonics you want to remove will be cut in a more audible way (Mellor, 2009). If a filter is set to a 4-pole low-pass rather than a 2-pole low-pass it will sound more synthetic than "natural". Different settings may work for different purposes, e.g. a steep curve could work better to create an effect, where the more gentle slope will make it sound less "chopped" off (Reid, 1999, chapter 3.).

In Subtractive synthesis low-pass filters are very important. The reason for this is that the low-pass filters remove harmonics from the high end of the frequency spectrum, which in most cases are the harmonics you would want to remove/attenuate (Buchanan, 2011, chapter 3.). When setting a low-pass filter to its minimum value you can control the amount of harmonics you want to pass through, as the first sound you will hear whilst "opening" the filter is usually the fundamental. From there the first harmonic will appear and continue until all harmonics are heard: "As the cut-off frequency of a low-pass filter is raised from zero, the first frequency that is heard is usually the fundamental. As the frequency rises, each of the successive harmonics (if any) of the sound will be heard" (Russ, 2009. pp. 114).

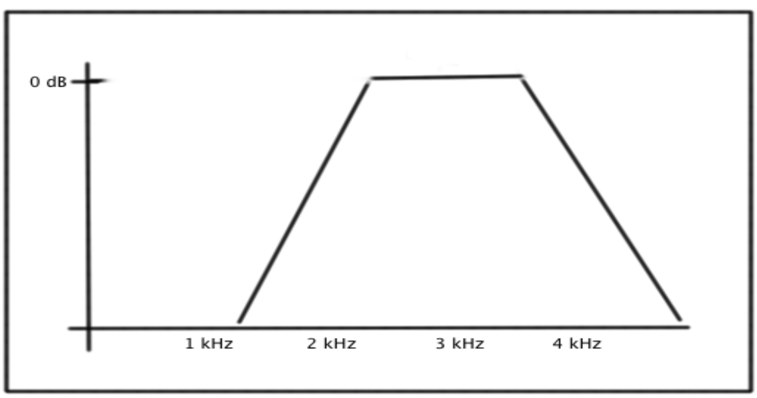
2.2.2 High-pass:



A high-pass filter has the same job as any other filter: to filtrate unwanted frequencies away from a sound, but opposed to the low-pass filter it attenuates frequencies bellow the given cut-off point.

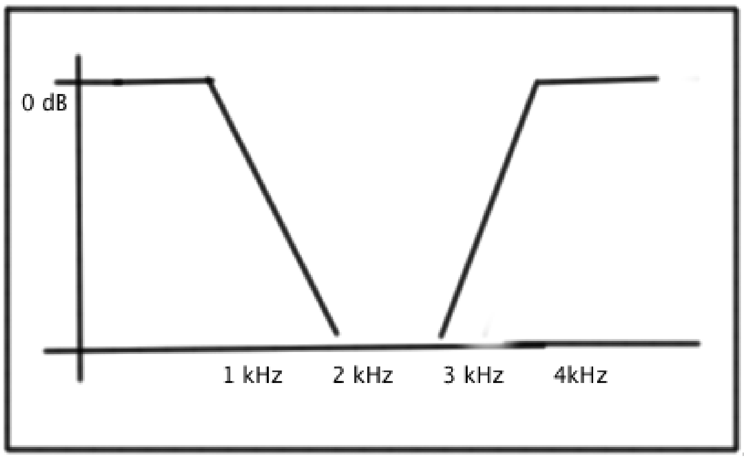
High-pass filters can be used for removing lower frequencies from a kick drum so it doesn't "crash" with the sub to mention one example. These filters remove harmonics from a signal waveform (Russ, 2009. Pp. 114), but as the filter raises from zero it is the fundamental that is removed first. The timbre of the sound will change gradually as you remove more low-end frequencies, and it will sound "brighter" and thinner.

2.2.3 Band-pass:



A band-pass filter is basically a low-pass and a high-pass filter put together. The band-pass filter only allows a set range of frequencies to pass through unchanged, and all other frequencies are attenuated (Russ, 2009). The frequencies that are allowed to pass through is called bandwidth or pass-band. This type of filter can drastically change the timbre of a sound depending on the “narrowness” on the filter setting. The wider the band-pass filter is, the less it affects the timbre of the sound because it barely attenuates the frequencies around the bandwidth. Very narrow band-pass filters can also be used as examination tools for listening to the frequency content of waveforms and isolate harmonics (Russ, 2009).

2.2.4 Notch:



“A notch filter is the opposite of a band-pass filter” (Russ, 2009. Pp. 116).
Taking this citation into consideration, the notch filter does everything the other way around. The notch filter is basically a band-pass filter turned upside down. When a band-pass filter is used it attenuates all frequencies around the bandwidth, whereas the notch filter attenuates only these frequencies and lets all the other frequencies to pass through unaffected. Where the band-pass filter can be used to listen to individual harmonics, the notch filter allows for single harmonics to be removed, given that the notch filter setting are set to a narrow value.

2.2.5 More about filter slopes:

The slope on these filters determines the steepness of the filter curve (Izhaki, 2008, pp. 220). The steepness affects how the filter will attenuate the frequencies from the given cut-off value. As Roey Izhaki states in "*Mixing audio*"; "With an aggressive slope of 30 dB/oct, it will only take two octaves before frequencies are attenuated by more than 60 dB (Izhaki, 2008, pp. 220). When using steep slopes like this, the filter can create unwanted effects on the sound such as perceived muting. Slopes like this may also be recognised with the *Brick Wall* filter, where the filter cuts all frequencies from the set cut-off value. This filter type can be used for creating effects rather than "normal" filtering because of its way of dividing frequencies to such an extent that the sounds frequencies is put to a dead stop. It should be clarified that this type of filter only exists in theory, but there has been made some filters to try and create this effect such as the *Brick Wall* filter in *Alloy* by iZotope. If a 12 dB/octave or 6 dB/octave filter is used it presents a more gentle slope compared to the 30 dB/octave.

The number of poles determines the ultimate filter slope rate. Each pole contribute 6 dB to a low or high-pass filters ultimate slope rate (SoundFirst, 2013, section 3). This means that when a filter is a 2-pole high-pass it has a 12 dB/octave gain loss because of the number of poles and the 4-pole low-pass has 24 dB/octave gain loss.

Chapter 3: Practical work

3.0.1 Reaktor

For this assignment I used Native Instruments "Reaktor" software. This is a construction tool for making various virtual devices. Reaktor can be used for making synthesizers from scratch, create effects such as reverbs and delays and it can be used as a regular synthesizer. Reaktor has also been given its own community webpage on Native Instruments homepage. On this webpage the Reaktor community can discuss and exchange their experiences and upload their self-made synthesizers, as well as giving feedback on how they liked it and give help with problems occurred during the creation process.

Using Reaktor is a great way of learning how different synthesis models work and discover how the different components used affects the end result. By including information tabs on almost all of the building blocks, macros[3] and core cells, they give the users a nice introduction as to what the components do and how they work.

3.0.2 The synthesizer:

The constructed synthesizer (presented in figure 1*)* has 4 oscillators, 2 filters, 2 LFOs[4], and a small effect rack consisting of a *Micro Space Reverb* and a *Delay*.

[3] Empty "containers" that can be used for fitting filters, oscillators and all other components in the synthesizer.

[4] Low Frequency Oscillator. Used for modulating different parameters.

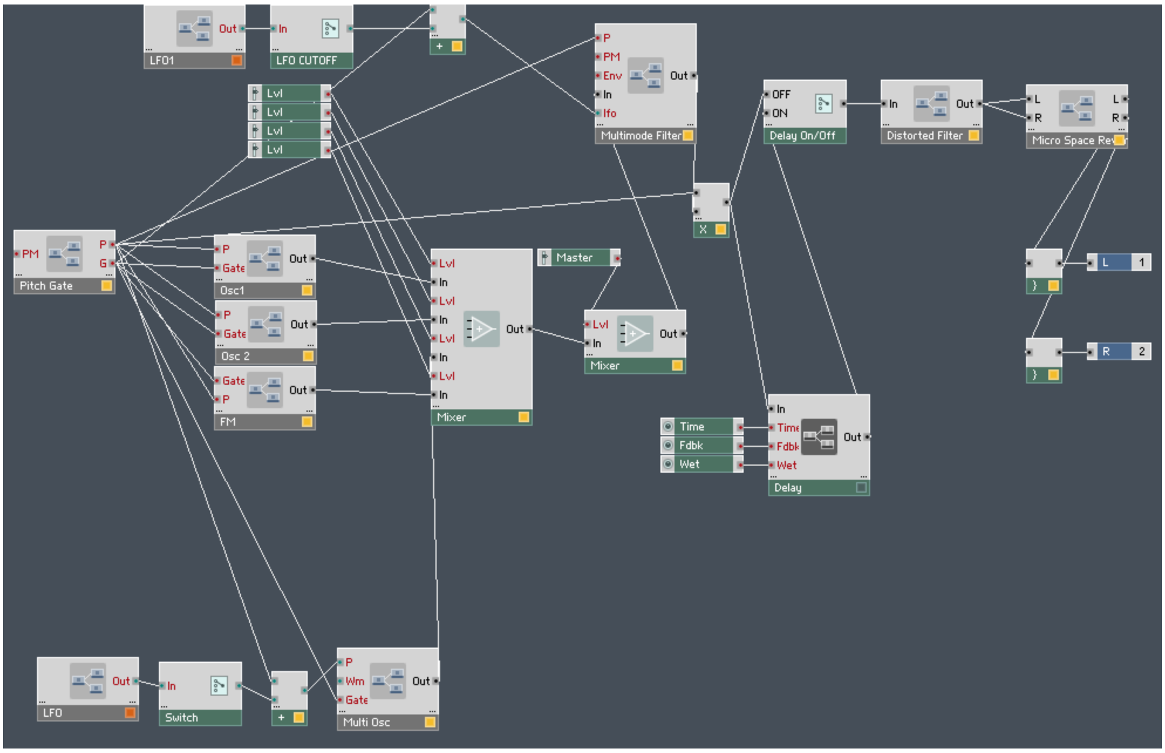


Figure 1, construction of the synthesizer

3.0.3 Oscillators:

The presented synthesizer has 4 oscillators – two of which are identical – giving the user many possibilities of combining different waveforms to create dynamic and various sounds. To create a signal that triggers the oscillator, you must add a "Pitch/Gate"(P/G) function in Reaktor (figure 1). This device sends out a signal to each oscillator in the synthesizer, causing the oscillators to make sound. The P/G function works as a trigger, so when you press a key on a midi controller that is when the P/G receives and sends out its signal. The G signal is sent to each oscillator and is received by the Gate, and the P signal is sent to each oscillator and received by the P.

Oscillator 1 and 2 (see figure 2*)* consists of 4 different waveforms:

- Sine(sin)
- Triangle(tri)
- Sawtooth(saw)
- Pulse(pul)

Oscillator 1 and 2 (figure 2) are self-made. These two oscillators have four waveforms, without any noise elements. On the left side of figure 2 you can see three boxes named: Pitch, Fine and P. These three signals gives the user the opportunity of fine tuning and octave pitching each oscillator. To do this the *Add*[5] function is used multiple times. First you assign one *Add* to adjust both pitch and fine tuning, then the signal is sent into an additional *Add*, which receives the main pitch (key pressed on controller) and blends with this signal to make it possible to tune the oscillators (see figure 2).

The oscillators also have built in dedicated envelopes. The reason behind this is for the user to have full control over each oscillator. By having dedicated envelopes you can control the length and amplitude of each sound *(ADSR*[6]*-*[7]*).*

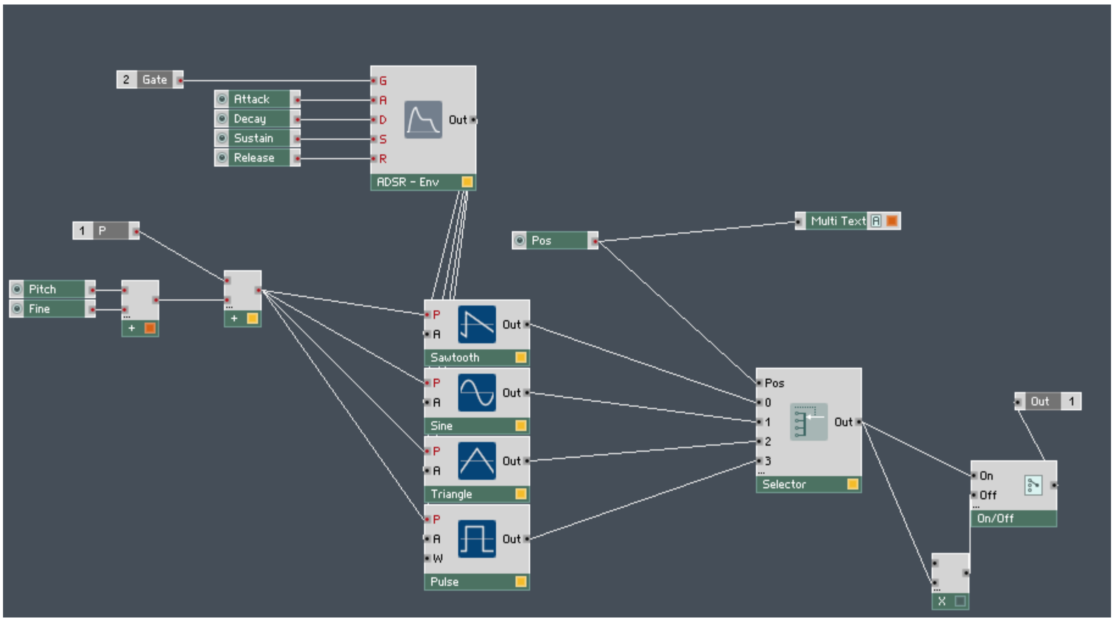


Figure 2, oscillator 1&2

3.0.4 The Multi-oscillator:

The Multi-oscillator (Multi-osc) is the only oscillator out of the four that can be modulated by both LFOs. The dedicated LFO for the Multi-osc applies a signal to the "Digitally Controlled

[5] *The "Add" function adds multiple signals. The Add functions symbol is "+".*
[6] *ADSR abbreviation for: Attack, Decay, Sustain and Release.*
[7] ***Attack** – Time taken to reach highest point. **Decay** – Time taken to decay to sustain level. **Sustain** – Sustain level. **Release** – Time taken to drop back to zero.*

Oscillator"(DCO) and pitch modulation comes to effect. "You get frequency (or pitch) modulation, where the sound wavers back and forth between two pitches.." (Casabona, 1988, pp. 25).

Using an LFO this way can create for example the well known siren-like sound as well as many others, like vibrato or a trill effect, all depending on wave shape utilized.

The Multi-osc consists of 6 waveforms, four of which are the same as oscillator 1 and 2, but in addition this oscillator has white noise and a 2-saws waveform.

3.0.5 Frequency Modulation oscillator(FM):

This is a simple Sine FM oscillator with frequency modulation, created to modulate other waves. This oscillator was built using "core-cells". (see figure 3)

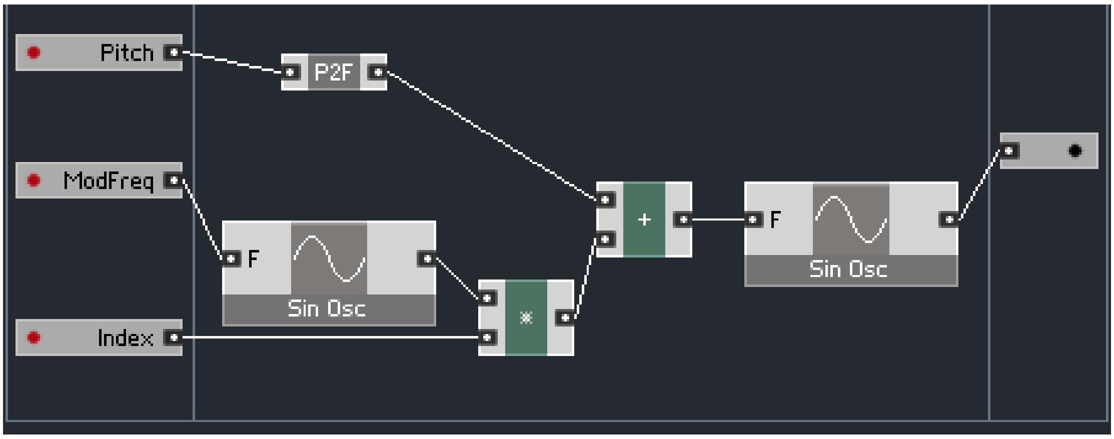


Figure 3, core-cell

In the core-cell above the *Index* is the FM depth by which the modulator modulates the carrier. The "*ModFreq*" regulates what frequencies that are being modulated.

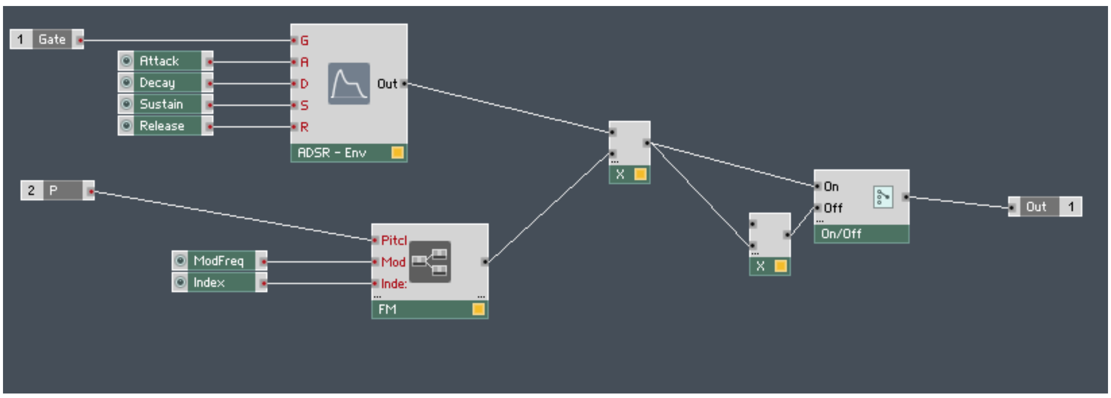


Figure 4, FM macro

“The original FM synthesizers did not have filters” (Cann, 2005, pp. 105).
This synth on the other hand has filters. By having filters on top of the FM one can adjust the sound of it even more giving countless possibilities, and adding a FM part to the Subtractive synthesis combines well. As Simon Cann states “FM and Subtractive sounds can be perfect compliment for each other, with the characteristics allowing the respective sounds to work together perfectly into a mix” (Cann, 2005, pp. 105).

3.1.0 Filters:

“A **filter** is a specialized amplifier that controls the amount of gain to prescribed frequency ranges of a sound” (Holmes, 2012, pp. 226).
There are two filters in this synthesizer. One Multimode filter and one Distortion filter. Both filters can be utilized the way that the user wants. By inserting a switch on the distortion filter and a bypass button within the multimode filter, the user can decide whether one of the filters, both or none shall be active.

3.1.1 Distortion filter:

The distortion filter is used for effect purposes. With this filter you can create “dirty” sounding synths as shown in the snapshot[8] “Guitar/Bass lead”. This filter is a combination of a 2-pole 12dB/octave high-pass and a 4-pole 24dB/octave low-pass, which can both be overdriven, thereby giving the distorted sound.
To adjust the amount of distortion you want on a sound, simply adjust the amount of gain on the *in gain* knob on the panel of the synth. The *out gain* knob is a gain for the filtered and distorted signal.

3.1.2 Multimode filter:

The multimode filter is more traditional and used for “standard” filtering. It has 10 different filter configurations. Some of these are: High, Low, Notch and Band-pass filters with multiple choices of poles.

[8] Snapshot is a pre-set in Reaktor.

Examples

- **The 2-pole** high-pass filter allows higher frequencies to pass through from a set cut-off value. This means that the filter attenuates the lower frequencies from a sound, and let the frequencies above the set cut-off value to pass.
- **The 4-pole** low-pass filter allows the low frequencies to pass through from a set value on the cut-off. Frequencies above the set value on the cut-off are attenuated.

There are also several kinds of notch filters and bandpass filters with different attenuation settings.

- **Notch 4** which attenuates only the frequencies near to the cut-off (band-reject).
- **LP Notch** which is a 4-pole low pass with a notch at the cut-off frequency.
- **Bandpass 1/3** that attenuates low frequencies with 6 dB/oct and high frequencies with 18 dB/oct.
- **Bandpass 4** is a 4-pole filter which attenuates both high and low frequencies with 12 dB/oct.

The multimode filter is used for shaping the sound and decide what frequency range should pass through. As mentioned earlier, a subtractive synthesizer uses filters to remove harmonic content, so when you put on a low-pass filter you remove harmonics from the higher frequencies to a given value on the filter cut-off. Of course this is given that the sound source is harmonically rich.

3.1.3 Creating the multimode filter and filter envelope:

The multimode filter is a “pre-set” in Reaktor. To add the filter into the synthesizer you simply right click an empty space inside the instrument and click the following:

Macro -> *building blocks* -> *filters* -> *multimode filter.*

When the multimode filter has been selected it will become visible inside the synthesizer and only needs to be connected to work. This involves connecting all the components that is going to pass through the filter, shown in figure 5.

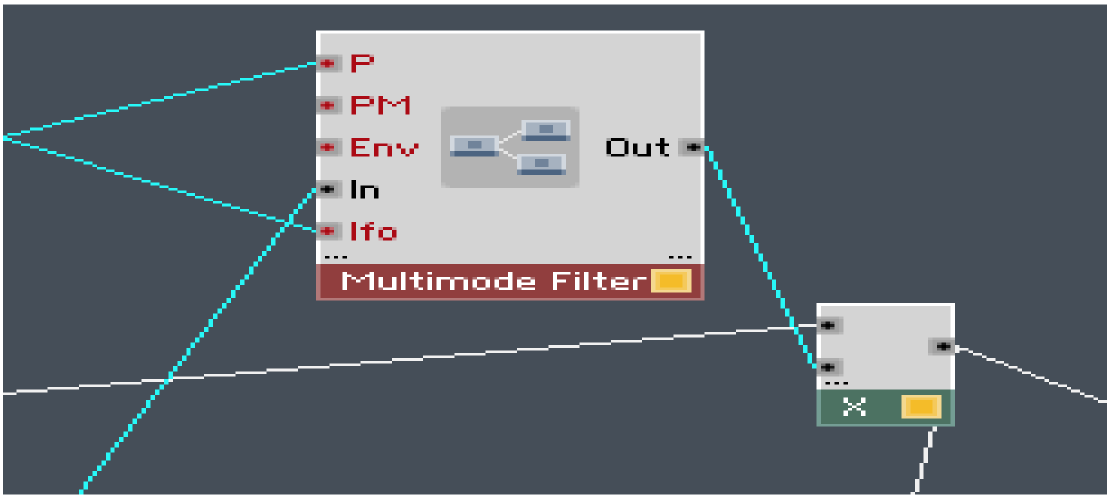


Figure 5, Multimode filter connections

In the picture above you see the connections on the multimode filter:

- The **In** connection receives the signal from all of the oscillators
- The P connection receives pitch information from the pitch/gate
- The lfo connection receives its signal from the LFO/FreqCut

After all these connection are done, the *out* sends all its information to a *multiply* where it is joined by another pitch signal and sent through the rest of the synth.

The multimode filter has two modulation tools connected to the filter cut-off. It has an LFO for frequency modulation (figure 7) and an ADSR-envelope. In the picture bellow you can see the ADSR-envelope connected to the cut-off:

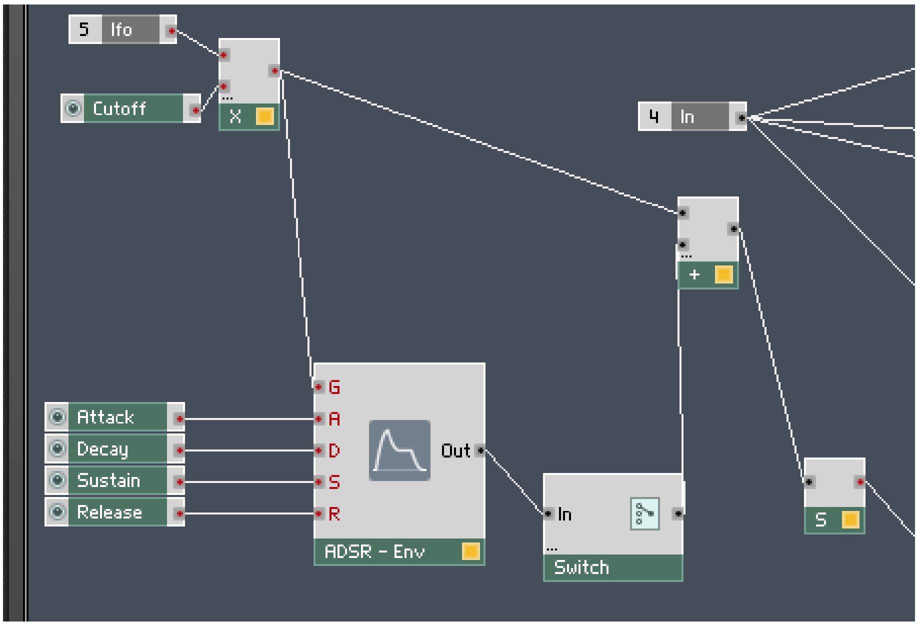


Figure 6, filter envelope

This envelope gives further possibilities to modulate the filters cut-off and control the cut-off amplitude. It works in the same way as the envelopes on the oscillators: ***Attack*** *– Time taken to reach highest point.* ***Decay*** *– Time taken to decay to sustain level.* ***Sustain*** *– Sustain level.* ***Release*** *– Time taken to drop back to zero.*

3.2.0 LFOs:

Connected to the filter cut-off on the multimode filter, an LFO is connected giving modulation possibilities to the cut-off. By doing this the LFO modulating the filter causes the filter to open and close as the LFO cycles, creating a "Wah Wah" effect. This is also how a lot of the basses in Dub-step are created, but in that environment they call it "wobble" rather than "Wah Wah", all depending on how the LFO is utilized. There are 5 different waveforms that can be used to modulate the frequency cut-off. These are: Sine, Triangle, Pulse, Brown, and S&H. Each waveform represents a different curve as to how the LFO will affect the frequency cut-off and gives possibilities to have more "interesting" sounds rather than just the "pulsating" or "wobbly" effect.

3.2.1 Creating the LFO for frequency cut-off:

Macro -> Building Blocks -> LFO etc -> LFO

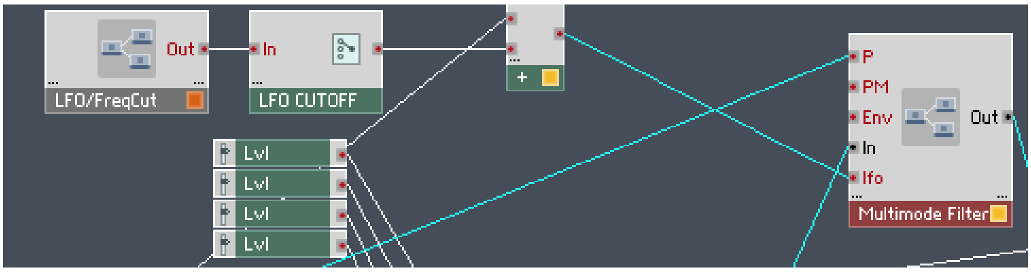


Figure 7, LFO/FreqCut connection

After creating the LFO/FreqCut the signal is sent to a switch called *LFO CUTOFF*. This switch enables the modulation to be bypassed if the user doesn't want it on and after the switch the signal is sent into an *Add* where it is met by the *Gate* signal before being sent into the LFO-connection on the multimode filter.

As mentioned in the "Multi-oscillator" section, there is another LFO in this synthesizer. This LFO is connected into the multi-osc and modulates the pitch. It was created exactly the same way as the LFO/FreqCut, but instead of sending it into a filter, it's sent into a switch followed by an *Add* where a pitch signal is also received, and from there the signal goes to the P connection on the multi-osc (see figure 8).

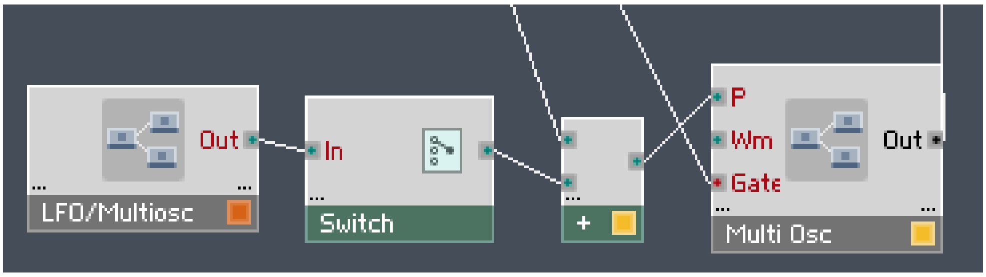


Figure 8, LFO/Multi-osc

3.3.0 Delay:

"The simplest units will allow you to delay the incoming audio signal by a predetermined time, which is commonly referred to in milliseconds or sometimes in note values" (Snoman, 2004, pp. 55).

Bellow you can see a picture from the created delay for this synthesizer. It is a simple, yet effective delay.

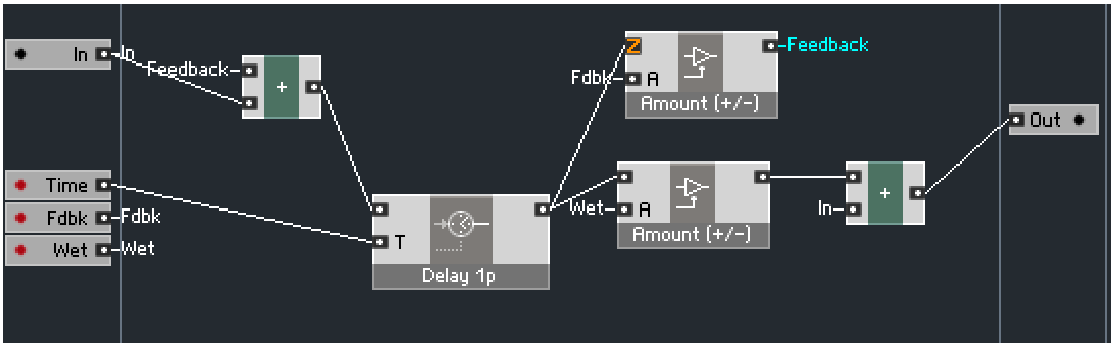


Figure 9, delay

There are several components that are needed for this delay to work properly.
The **Feedback** function on the delay sends the output signal back to the input so the sound passing through is repeated/delayed again. This makes it possible to have almost endless amounts of "delays" by adjusting the value of the feedback.

The signal flow is explained on the picture above. We see that the "Feedback/Fdbk" starts right after the input, it then moves on to a "standard macro", the amount(+/-) cell, and is from here sent back to the first feedback through wireless routing.

Among the feedback knob we have a **Wet** function as well. The feedback takes care of the repetitions and the wet knob controls how much of the effected sound is heard alongside with the dry sound. If the wet knob is turned all the way down to "0" value, it is only the dry sound that will go through.

The third knob is called **Time**. By adjusting this parameter, you define the time it takes for each repetition to happen. If a low value is chosen the cycle of which the delay repeats, will be short. If a high value is chosen the cycles will take longer time.

Chapter 4: Results and Conclusion

4.1 Why create two types of synthesis?

The reason for creating two types of synthesis for this assignment was to have a synthesizer that could create all types of sounds. Even though subtractive synthesis manages to create a range of "interesting" sounds on its own, the addition of FM takes it one level higher. As you can hear whilst listening to some of the snapshots created with my synthesizer, taking out the FM oscillator would strip down the depth and leave some of the sounds rather naked. Take "Horror spaceship" as an example. If you turn off the FM oscillator the sound loses all its depth, and doesn't impact the same way. This snapshot was made to give the feeling of something peculiar or "freaky" happening, and personally I think it loses that feeling when used without the FM oscillator.

Another reason for having the FM in this synthesizer is that the FM oscillator is used mostly to create effects. When creating more tone based synth-sounds like the snapshot "Pulsating Pad", the subtractive part of the synth is used, but when you want to create a weird effect the FM oscillator became a personal "go-to" oscillator.

4.2 Conclusion:

When it comes to the subtractive part of the presented synthesizer there were several things that was needed to create it. Firstly a harmonically rich sound source was needed, in this case oscillators with multiple waveforms. The reason for this is that subtractive synthesis works by removing harmonic content from harmonically rich sounds. If the synthesizer had only one waveform which was a sine wave, there would be no harmonics to remove because the sine wave only consists of one harmonic, also known as the fundamental. So the sound source has to consist of more than a fundamental with no harmonics, e.g. a saw-tooth waveform because it has the most harmonics. This is why I used several oscillators with many different waveforms containing different harmonics. It was done so the synthesizer actually had something to subtract. If this synth didn't have any harmonically rich waveforms it could not be described as a subtractive synthesis.

The subtractive synthesizer also needs a way of removing these unwanted harmonics, and can do so by the use of filters. By having a wide range of optional filter types and settings, the user can choose what harmonics of a sound they would want to remove, and this is the reason for choosing the multimode filter for this synth. Because of the multimode filters many filter types you can remove the harmonics you want. For example removing lower frequencies with a high-pass filter or removing high and low frequencies with a band-pass filter. On the other hand, through the research I've done for this paper it seems to me that the most important filter type in a subtractive synthesizer is the low-pass filter. To be able to remove harmonics from the higher range of frequencies seems more useful than to only remove lower harmonics which in most cases starts with the fundamental. When Pekonen and Välimäki states that musicians started to take new interest in the "warm" sound given by subtractive synthesizers in the mid 90s, it seems evident that the low-pass are the filter that can bring this timbre to a synthesizer. In addition to filters, envelopes connected to the filters are important sound shaping tools as well. Not a fundamental component compared to the filters, but it gives more modulation possibilities.

Overall I would say that for a synth to be a subtractive one it needs the following:

- Oscillators containing several waveforms that are harmonically rich
- A low-pass filter
- The filters in the synth needs to have several filter settings (1-pole low-pass, 2-pole etc.)

So I would say I have covered what is needed for my synthesizer to be a subtractive one. Of course there are always new components and features you would want to add to your synthesizer to make it even better, some of which I will discuss in the improvement section on page 31.

The simple sine FM is a small part of this synthesizer. Even though it's only one oscillator it delivers great sounds by the use of only two knobs. I think that for making effects this is a great addition to the presented synthesizer and as Simon Cann states "FM and Subtractive sounds can be perfect compliment for each other, with the characteristics allowing the respective sounds to work together perfectly into a mix" (Cann, 2005, pp. 105). But as with the subtractive part of my synthesizer there are several things I would change if I were to redo the FM.

4.3 What improvements could have been done?

If I were to create the synthesizer all over again I would have added more filters. If I had set up one filter per oscillator, instead of one filter for all oscillators, it would have been a lot more powerful synthesizer. This would have made the synthesizer extremely versatile when it comes to filtering harmonics of different waveforms individually rather than collectively.
The addition of auxiliary envelopes with routing capabilities to pitch modulation would also have given the synth a new "lift". Assigning an auxiliary envelope to the pitch will result in a quick drop or rise in pitch if the attack is set to a low value. If the attack is set to a high value the pitch drop or rise would take a longer.

Regarding the simple sine FM oscillator it works as more of an effect-making oscillator and it can't really be used in the way that "proper" FM synths should. Because I have only one oscillator I can't layer the FM oscillator on top of several others. If I added 3 more oscillators with more waveforms the possibilities of making complex sounds and emulations would be there.
If I were to redo the FM section of the synthesizer I would have two oscillators which both contained sine and triangle waveforms. These oscillators would be connected so one could oscillate the other. By doing this I could have emulated a typical "Bell FM" sound amongst other things and the FM part of my synthesizer would be more useful and versatile.

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Appendix

Snapshots

For this paper a number of snapshots were made. This is sounds that I have created with the presented synthesizer. I chose to create 5 different “pre-sets”, to show the diversity of sounds you can make with a basic synth built in Reaktor.

Snapshot one, *Pulsating Pad*, uses 2 waveforms, Saw and Pulse. The oscillators work to blend with each other reasoned long attack on the Pulse-wave. Lowpass filtering is utilised by the multimode filter with the cut-off opened letting more high frequencies through. The lowpass is filtering away the highest frequencies in the sound. This is just personal preference, but can be easily adjusted by turning the “Cut-off” knob on the front panel of the synthesizer. Some reverb is used to give the illusion of space around the sound, giving it a bit of room to not sound too flat. There is also added an LFO, that modulates the cut-off to give the sound a pulsating effect.

Number two is called *Guitar/Bass* lead. This is created with the intent to have a bass that can play both low and high notes and still give the feel of power. The Guitar/Bass lead contains 3 oscillators all on Saw/2Saws waveforms with different octave pitch. All the envelopes are set to relatively high values on sustain, making the sounds louder and giving it constant power that will not fade away immediately. It also uses both filters (Dist & Mult), thus creating the distorted feeling of the instrument. At the end of the chain a small sized reverb is used to create a metallic feel when the notes are released.

Number three is called *Plucked synthesizer*. A Saw and Pulse wave is used to give a sharp and aggressive sound. There is no attack applied on the envelopes, some decay, but most importantly no sustain. By not having sustain to a certain degree it gives the impression of “plucking” an instrument, or gives the staccato feel. This snapshot uses a high-pass filter, cutting away the lower frequencies and by using a reverb with gained high notes it completes the sound of big spaces.

The fourth one is called *Horror Spaceship*. This sound tries to show how versatile this synth can be. Here a FM Osc is used, layered with a Triangle that is pitched 13+ semitones. The oscillators are sent through the multimode filter and 2 LFO’s. The Multi-osc is fed through

both LFO’s giving it a very fast rate that modulates both filter cut-off and pitch modulation. The LFO/FreqCut are set to shape S&H, this shape provides a varied pattern. Playing this sound using the higher notes shows the effect that LFO-pitch modulation has on the Multi-osc.
This snapshot varies from the high notes to the low ones and gives out small adjustments at different octaves.

Incoming Waterfall is the fifth one and it used all four oscillators where FM is the only one without attack. A noise source is used as well for this sound, providing the “water” effect. The point is to adjust the attack, and by time it goes from dominated FM sound type of drone, to a waterfall sounding atmosphere following the horror theme from the example above. The reverb used in this pre-set has boosted the lows and highs and set to a big distance, so this would convey for a scene outside.

Last snapshot, number six, is called *Aliens are marching*. This snapshot uses all four oscillators and a band-pass filter. This filter attenuates low and high frequencies with 12 dB/octave filter (2-pole) added with the distortion filter that makes the snapshot sound closed off, and a LFO on the cut-off creates the marching effect.