Prototyping a Modular Analog Synthesizer

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Introduction

This paper constitutes an attempt to summarize the most important facts and information on the topic of analog, modular synthesizers. The range of discussed subjects involves a series of various perspectives, including historical, theoretical, electronic and practical viewpoints.

Its goal is to convey an understanding of the inner workings of electronic synthesizers and their components. Moreover the reader is guided through the process of creating a small but functional modular synthesizer setup that is fun to play and experiment with. The intention was to investigate the possibilities and limits in designing and building an analog sound device for someone who had not been in contact with analog synthesizers before, let alone building electronics devices. The project was inspired by the film *moog* (Fjellestad, 2004) , a documentary about Dr. Robert Moog, electronic instrument pioneer and inventor.

Chapter Overview

The chapter *Historic Evolution of the Synthesizer* represents the research on the historical background of analog synthesizers since the beginning of the twentieth century. It was tried to outline important milestones in the historic development from the first electronic sound generating devices until a point in time when manufacturers of modular synthesizers have developed a profitable market.

Subsequently the most important concepts of subtractive synthesis are summarized. A general overview over common sound generation and processing methods is given, whereby all concepts are applicable to both analog and digital synthesis. In chapter three these concepts are taken one step further and discussed in the context of electronic circuitry. Lastly the process of building an electronic synthesizer prototype is described.

2.1 Early Development Milestones

Around 1900 american Thaddeus Cahill initiated a new era of music by inventing a machine known as the Dynamophone or Thelharmonium (Humpert, 1987, p. 19). Working against incredible technical difficulties he succeeded to create an electrical sound generator with a weight of around 200 metric tons. It produced alternating sine wave shaped currents of different audio frequencies. A modified electrical dynamo was used in conjunction with several specially geared shafts and inductors to create the signals. The Dynamophone could be played with a polyphonic keyboard and featured special acoustic horns to convert the electrical vibrations into sound (Manning, 1985, p. 1). The timbre of the instrument was manually shaped from fundamentals and overtones. This is known as the principle of additive synthesis (Bode, 1984, p. 730).

In 1924 the russian inventor Leon Theremin created the Aetherophone (see figure 2.1), which would later be known as the Theremin. Unlike most electric instrument developed around that time, the Theremin had no keyboard. It was played merely by hand motion around two capacitive detecors, that generated electrical fields. These were affected by the electric capacity of the human body. One of these detectors was a vertical rod to control dynamics and the other a horizontal loop to change the pitch (Manning, 1985, p. 3). “The theatricality of its playing technique and the uniqueness of its sound made the Theremin the most radical musical instrument innovation of the early 20th century.” (Dunn, 1992, p. 6)

Figure 2.1: Leon Theremin performing the Aetherophone

Some organ-like precursors to the synthesizer were the Ondes Martenot and the Trautonium, which were devised just a few years later. The Ondes Martenot is one of the few early electric instruments, that are still in concert- and theatre use in their original design today (Humpert, 1987, p. 20).

The Givelet (1929) was a commercially more successful instrument, since it was designed as a cheap alternative to pipe organs. These instruments were polyphonic and unified the concepts of the Pianola - a self-playing piano, controlled by pre-punched tape - with electronic sound genaration. The ability to program electronic sounds should lead the way for future devices such as the RCA synthesizer or computer music production in general. However, the Givelet was about to take a back seat, when Laurens Hammond published his Hammond Organ in 1935. Its technical operation principle is reminiscent of the Dynamophone, since it also involved rotating discs in a magnetic field (Manning, 1985, p. 3).

The german engineer Harald Bode contributed to the design of several new instruments from the 1930's on, like the warbo formant organ (1937) or later the Melochord (1949) (see figure 2.2). He was primarily interested in providing tools for a wide range of musicians, which is why his contributions straddled between the two major design traditions of new sounds versus imitation of traditional ones. He turned out to be one of the central figures in the history of electronic music, since he was also one of the primary technicians in establishing the classic tape music studio in Europe (Dunn, 1992, p. 9).

Figure 2.2: Harald Bode tuning his first Melochord

Bode was one of the first engineers to grasp the significance of the invention of the solid state transistor for sound synthesis. In an article published in 1961 he draws particular attention to the advanteges of modular design. "The versatility of transistor-based electronics made it possible to design any number of devices which could be controlled by a common set of voltage characteristics." (Manning, 1985, p. 117). But it was not until the early 1960's that major advances in electronic design took shape (Dunn, 1992, p. 19).

2.2 The First Synthesizers

In 1955 the laboratories of the Radio Corporation of America (RCA) introduced a new and very advanced machine to the public named the Olson-Belar Sound Synthesizer, later known as the RCA Mark I Music Synthesizer. It combined many means of tone generation and sound modification known at the time and is considered the first synthesizer. Mark I was built with the specific intention of imitating traditional instrument sounds and to reduce the costs of the production of popular music by replacing musicians. However, the machine proved unsuitable for its original intent and was later used completely for electronic music experimentation and composition (Dunn, 1992, p. 15-16). The synthesizer could not be played in the conventional sense in real time. Instead musical information had to be pre programmed as punched holes in a large paper tape. Harry Olson and Herbert Belar produced an improved Mark II Synthesizer in 1957, which the nickname *Victor* was given.

Around the same time the outstanding guitarist and inventor Les Paul became famous with his multitrack guitar recordings. He stimulated many innovators not only with the success of his multitrack recorder, but also with his methods of electronic sound processing. Harald Bode was so impressed and inspired by his work, that he built a system consisting of a number of electronic modules for sound modification in late 1959 through 1960. His system featured ring modulator devices, envelope followers and generators, voltage-controlled amplifiers, filters and other (Bode, 1984, p. 733). The modular concept of his device had proven attractive due to its versatility and predicted the more powerful modular synthesizers that emerged in the early 1960's (Dunn, 1992, p. 20).

In 1963 Robert Moog, a passionate inventor from Ithaca, New York, United States, was selling kits of transistorized Theremins (Dunn, 1992, p. 20). In the documenteray *moog* (Fjellestad, 2004) he states that he had been completely obsessed with building and later designing Theremins since the age of 14. A year later he built a transistor based voltage-controlled oscillator and amplifier for the composer Herbert Deutsch. This led Moog to the presentation of a paper entitled *Voltage-Controlled Electronic Music Modules* at the sixteenth annual convention of the Audio Engineering Society, which had stimulated widespread interest (Manning, 1985, p. 117-118).

Figure 2.3: Donald Buchla with a Series 100 system in the 1960's

Similar developments had been taking place at the west coast of the united states. Morton Subotnick and Ramon Sender started their career in electronic music experimentation and became increasingly dissatisfied with the severe limitations of traditional equipment at the San Francisco Tape Music Center. They sought out to hire a competent engineer and met Donald Buchla (see figure 2.3) (Manning, 1985, p. 117-118). Their discussions resulted in the concept of a modular voltage-controlled system. Buchla's design approach differed significantly from Moog. He rejected the idea of a synthesizing familiar sounds and resisted the word *synthesizer* ever since. To him it seemed much more interesting to emphasize new timbral possibilities and stress the complexity that could arise from randomness. At the same time Buchla was fascinated with designing control devices other than the standard keyboard, which Moog decided to use for playing (Dunn, 1992, p. 20). "Buchla has maintained his unique design philosophy over the years producing a series of highly advanced instruments often incorporating hybrid digital circuitry and unique control interfaces." (Dunn, 1992, p. 23)

In 1966 Robert Moog's first production model was available from the business R.A. Moog Co. that he had founded (Dunn, 1992, p. 20). At this time Walter Carlos, an audio engineer from New York who advised Robert Moog while perfecting his system, worked with Benjamin Folkman to produce an album of titles by Johann Sebastian Bach interpreted only with Moog synthesizers (see figure 2.4). With the title *Switched-on Bach* they demonstrated the performance of the system so convincingly, that they hit the pop music charts and sold a million LP's (Ruschkowski, 1990, p. 45).

Figure 2.4: Switched-On Bach LP artwork

By the end of the decade two other manufacturers entered the market: ARP in America and EMS Ltd. in England. They had become major rivals for Moog and Buchla. Synthesizer production was dominated by these four companies for several years, whereby each firm struggled for a major share of a highly lucrative, rapidly expanding market (Manning, 1985, p. 118).

During the last century different kinds of synthesis methods have emerged. The subtractive synthesis however, appears to be the most common one. It is a method of sound synthesis, where partials of the initially generated signal are attenuated with a filter to change the timbre of the sound. But how can these sounds actually be synthesized? And what exactly does a filter do? These questions will be examined in the following.

3.1 Sources

Acoustic events can generally be divided in two groups: noises and tones. Whereas tones - as opposed to noise - are classified as sound waves, that have determined set of frequencies, which oscillate in a sinoidal manner. However this is only a theoretical classification, since most natural sounds are a combination of the two (Ruschkowski, 1990, p. 52).

3.1.1 Wave Oscillation

At the root of every artificial tone generating system there is an element that produces an oscillation. This element is mostly described as the oscillator, which represents the very source of what can be heard eventually. The oscillator produces a wave, that moves between an amplitude-minima and -maxima. Its waveform (shape of the wave) determines the overtone structure and therefore the timbre of this basic source sound. Oscillators often provide several waveforms between which it is possible to switch back and forth (see figure 3.1). The pitch of the output signal is defined by the frequency of the wave and must oscillate between 20 Hz and 20 kHz in order for it to be audible to humans (Friesecke, 2007, p. 124). The output signal can later be processed and modulated in several ways.

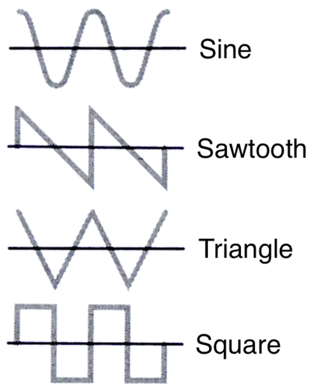

Figure 3.1: Common oscillator waveforms

Oscillators that swing at an infrasonic frequency - meaning a frequency so low, that it is not hearable anymore - are called low frequency oscillators (LFO). They are used to control parameters of different components of the synthesizer periodically. For example to influence the pitch of another oscillator to get a vibrato - or the amplitude to get a tremolo effect. Some oscillators frequencies range from very low to very high, in which case a distinction between oscillator and LFO is unnecessary.

Characteristics of Common Waveforms

Sine — The most basic waveform is the sine wave. It contains no overtones at all and sounds round and dull.

Sawtooth — The sawtooth, also known as saw or ramp waveform sounds very bright, sometimes described as trumpet-like (Anwander, 2011, p. 49). It consist of a complete series of harmonics and is therefore well suited for subtractive synthesis. There are two types of sawtooth waves: rising and descending.

Triangle — Composed of only odd harmonics, the triangle wave has a much softer, flute-like sound.

Square — Also known as rectangle, the square wave also consists of odd harmonics only, but the level distribution of its harmonics is different. Its timbre reminds of woodwind instruments (Ruschkowski, 1990, p. 55). A true square wave has a 50 % duty cycle - equal high and low periods. However, oscillators often feature a pulse width parameter, trough which the high-low time ratio can be accessed. This has a distinct influence on the wave's timbre. In this case, the square becomes a pulse waveform.

3.1.2 Noise Generation

A different approach on the creation of source audio material is resembled by noise generators. "In very loose terms, noise is a complex waveform with components of every frequency, each attaining a completely random amplitude at any give instant." (Henry, 2009, p. 1)

Noise Types

White — Equal power density in any band of the frequency spectrum

Pink — Power density decreases by 3dB per octave; also referred to as 1/f noise

Brown — Power density decreases by 6dB per octave; also referred to as $1/f^2$noise

The names of these noise types were derived from the spectral distribution of the correspondingly colored light (Friesecke, 2007, p. 155).

3.1.3 Triggering Notes

In order to use the previously discussed signal generators in a musical context, it is necessary to cut off their stationary signals when no note is being played. This is accomplished by routing the output signal of the generator to an amplifier and providing it with a gate signal. The source of the gate signal can be a keyboard or a sequencer, which would also send a pitch value to the oscillator to set its frequency (Anwander, 2011, p. 36).

3.2 Signal Processing

In their raw shape the mentioned source signals sound rather underwhelming, since they produce fixed timbres lacking of distinctive qualities (Manning, 1985, p. 49). To get a more interesting sound, the signal can be manipulated in acoustic color or dynamics by one or more processing units.

3.2.1 Dynamic Envelopes

The most important component responsible for shaping the dynamic structure of a note is the envelope. It is triggered by the gate on/off signal and outputs a control signal that fades between the different state phases of a note. The rapidity of these changes is adjusted by parameters, that represent these states. Its output signal can be used to control an amplifier and therefore shape the dynamic structure of the note. The most common envelope type is the ADSR, which stands for attack, decay, sustain, release.

Attack	Sets how long the envelope signal rises after a note was triggered
Decay	Sets how long it takes for the envelope signal to drop from its maximum to the sustain level after the attack phase was completed
Sustain	Sets the output level for the time period after the decay phase and before the gate signal was terminated
Release	Sets the length of the fade out after the note has ended

Envelopes can also be used to control other parameters, for example the cutoff frequency of a filter (see chapter 3.2.2).

3.2.2 Filtering

The filter is the processing component responsible for the sound changes, that people associate with *the typical synthesizer sound* (Anwander, 2011, p. 53). They remove a spectrum of frequencies from their input signal above or below the cutoff frequency and are often used in conjunction with an envelope or LFO modulation on the cutoff. This cutoff frequency is an important parameter determining the frequency at which the signal begins to be attenuated. The slope of the filter determines how abrupt frequencies are being cut. Another common parameter is called resonance. It controls the level of a feedback loop, emphasizing the timbre at the cutoff frequency.

Filters can generally be divided into two categories: Low pass and high pass (also called high cut and low cut). To get a band pass, low- and high pass are connected in series. When connected in parallel, they become a band stop or band reject filter. Lastly the all pass filter should be mentioned, which does not change the frequency spectrum but merely influences the phase of the signal around its cutoff (Anwander, 2011, p. 55).

3.3 Controllers

Controllers can be characterized by the way of how humans interact with them and how their output signal is applied in controlling other components of the system (Hutchins, 1975, Ch. 1A, p. 5). A keyboard for example is a manual controller, since it is the movement of the players fingers which are translated into a voltage or control value and then used to control pitch and amplitude of a note. The same applies for rotary knobs and faders or touch sensitive surfaces.

Sequencers on the other hand are programmable controllers. They are not dependent upon a manual interaction except for their programming and activation.

3.4 The Modular Approach

A modular synthesizer is an instrument, where sound generators, processors and control facilities are presented in separate independent entities called modules. These modules are not wired in a preconceived way, but are connected with patch chords. The second essential aspect is the concept of inter-modular controllability, with which modules may modulate or regulate other modules.

The previously discussed theoretical concepts can generally be applied to both analog and digital synthesizers. In this chapter some of these concepts are transferred into the electronic context.

4.1 General

In digital synthesis audio signals and control signals are often being differentiated. This is because computing power can be saved by limiting the sample rate for control signals, since they usually only require a fraction of the amount of sampling values to get acceptable results.

In the area of analog modular synthesis it is important to know that signals are nothing but alternating and/or direct currents usually with voltages ranging from -5 to +5 volts. That is why there is no difference between audio and control signals. This has a great impact on the extend of possibilities especially in modular systems. One of the consequences is that an oscillator can control the frequency of a second oscillator with its output signal. This is known as frequency-modulation or FM synthesis.

However, when combining or mixing control and audio signals one should be aware of the following phrase, stated by Donald Buchla: “DC offset doesn't make any difference in the sound domain but it makes a big difference in the structural domain, whereas harmonic distortion makes very little difference in the control area but it can be very significant in the audio areas.” (Dunn, 1992, p. 23)

4.2 Modules

In a modular synthesizer setup, all sound generating and processing modules are aggregated and connected to a central power supply with a common ground. Among the modular components the signals are transferred via patch chords. The modules should be engineered to have the lowest possible output impedance and a relatively high input impedance. A common impedance ratio is 1:100.

Below, the basic circuitry principles of some essential synthesizer components will be examined.

4.2.1 Oscillator

Natural Resonance

The most simple form of an oscillator circuit producing a sawtooth signal can be realized with just a few parts (see figure 4.1). A current charges a capacitor at a certain rate. Between the electrodes of the capacitor a voltage potential rises. Meanwhile the voltage is constantly compared with a reference voltage by a detector (Schmitt trigger). Once it reaches the predefined threshold, an electronic switch (transistor) is activated. This switch short-circuits the capacitor and discharges it, causing the output voltage to drop back to its initial potential. This is happening continuously, whereas the rate of the repetition determines the frequency of the generated signal.

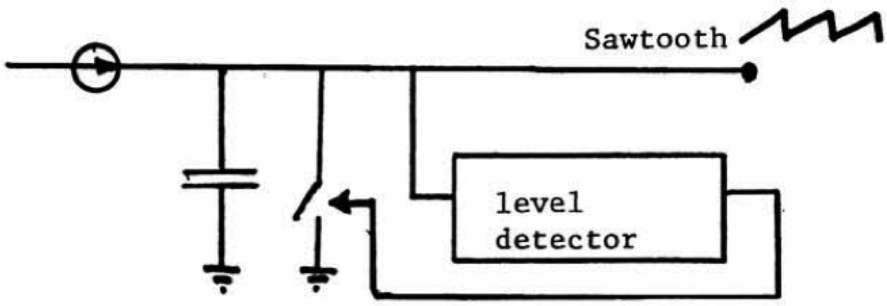

Figure 4.1: Abstract circuit scheme of a basic sawtooth oscillator

To get a triangle shaped output signal, the current source can be reversed in response to the detector (see figure 4.2) (Hutchins, 1975, Ch. 5B, p. 3).

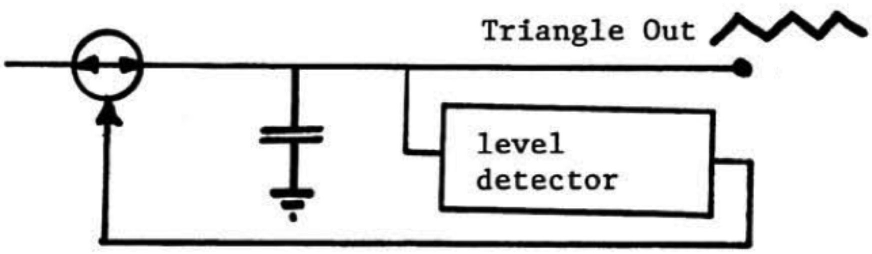

Figure 4.2: Circuit scheme of a simple triangle oscillator

These two waveforms can be wave-shaped into to other common basic waveforms. All required wave shaping methods are fairly simple and their accuracy is sufficient for musical purposes. The sine wave is an exception, because there is no simple electronic method of creating a pure sine wave. What is generally used is a rounded triangle, which gives at least 1 % of harmonic distortion. To get a purer sine wave, the harmonics can be filtered out with a voltage-controlled filter following in the signal chain (Hutchins, 1975, Ch. 5B, p. 3-4).

Voltage Control

How fast the capacitor is charged is determined by the intensity of the current that it is being charged with. This current is being extracted from a voltage, which of course can be a control voltage.

However, creating a stable voltage-controlled oscillator design presents one of the toughest challenges for musical engineers, because the human ear is extremely sensitive to pitch changes. Also compositional processes like multitrack recording plainly reveal errors in pitch relationships (Hutchins, 1975, Ch. 5B, p. 1). The voltage to pitch distribution curve of an oscillator is determined by its exponential response to the voltage at its CV input. In a volt per octave system an increase by 1 volt at the input must result in a doubling of the oscillation frequency at the output. Thus octave purity is achieved.

A major issue that oscillator designers have to face are temperature influenced variations of the electrical parameters of the system, causing the oscillator pitch to drift when temperature changes occur. To overcome this problem some kind of temperature compensation should be implemented.

4.2.2 Filter

Passive Filter

Filter circuits are generally based upon the fact that the transfer of alternating currents through a capacitor becomes increasingly weak below a certain frequency. A signal simply passed through a capacitor resembles a high pass filter. If the capacitor is connected to the ground the high frequencies are short circuited and low frequencies remain in the signal path. To control the charging time of the capacitor and therefore the cutoff frequency a resistor is added. This is referred to as a resistor-capacitor (RC) element. This simple circuit represents a passive, first-order filter with a roll-off of 6 dB per octave. To increase the slope of the filter multiple RC elements can be connected in series (Anwander, 2011, p. 57).

Nevertheless this poses some disadvantages, because the filter stages influence each other. The resistor of the first filter affects the resistance of the second. This can be dealt with by designing the second filter to have a much higher impedance. Unfortunately a high impedance circuit is much more prone to interferences and signal noise (Sontheimer, 2004, p. 97).

Active Filter

A better and more flexible solution is the usage of an active filter design, like the Sallen-Key filter (see figure 4.3). This is a common second-order filter, which is applicable as a low or high pass, whereby only resistors and capacitors have to be swapped (Sontheimer, 2004, p. 97). Resistor (R_2) and capacitor (C_2) can be identified to form a low pass section (Hutchins, 1975, Ch. 4B, p. 4).

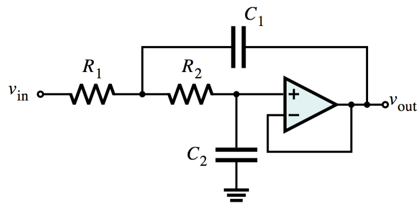

Figure 4.3: Active Sallen-Key low pass filter circuit scheme

To be able to change the cutoff frequency via voltage control the frequency defining resistor could be replaced with a voltage controlled resistor (Lancaster, 1975, p. 198).

4.2.3 Envelope Generator

The envelope generator is based on the principle of slew-rate limiting. The rapid changes in the level of the gate voltage can be decelerated by again using a resistor-capacitor element. By using a potentiometer in the place of the resistor the charging time and therefore the speed of the voltage alteration can be adjusted. To get a typical attack-release (AR) envelope the charge and discharge times should be adjusted separately. To achieve this two preceding potentiometers with a diode each should be connected. The diodes must be pointed in opposite directions (see figure 4.4).

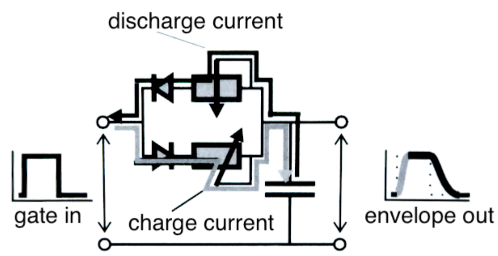

Figure 4.4: Principle circuit diagram of an AR envelope

5.1 Introduction

It is relatively easy to find circuits to construct simple oscillators and filters based on the fairly comprehensible concepts of resonant circuits and RC elements (see chapter 4.2.1 and 4.2.2). However, as their flexibility and capabilities increase (e.g. controlling the frequency of an oscillator with 1 volt per octave), the circuits tend to get exceedingly complex, requiring solid expertise in electronics.

This is why it was decided to switch to the usage of pre-designed, professionally manufactured circuit boards for this project as opposed to elaborating all the circuits on perfboards as originally intended. This made the goal of intermodular controllability attainable more easyily. The downside of this approach are higher costs for boards and parts. However, the quality of the end-product is impressing. Also the time saving using this strategy is not to be underestimated.

During the research phase of this project it came to the authors attention, that a modular synthesizer building workshop would be taking place in Berlin monthly. It is organized by a spanish collective from barcelona called *befaco* (`http://befaco.org`). At the workshop it is possible to acquire various module kits containing all necessary parts and also receive tips and support while assembling them.

Due to budget limitations, it was tried to arrange a smaller setup that would still offer lots of sound design possibilities.

5.2 Formats and Interfaces

There are several formats for module sizes, power supply plugs or patch chord connectors which emerged out of the production lines of various module manufacturers. For example Doepfer's modules are only compatible with their EuroRack cases, with a height of 128.5mm. These EuroRack modules use jack connectors for patching. In the DIY modular synth scene it is a common practice to use banana jack connectors instead of mini jack for patching. Those have the possibility of stacking the connectors on top of each other and split the signal without having to use a multiplier module. In this project the modules are EuroRack size, but use banana plugs.

5.3 Building

To get started with building electronic equipment, one has to obtain some tools first. This includes a soldering iron - best with adjustable temperature, a role of quality soldering tin, a desoldering pump and pliers for cutting and bending wire.

Soldering is a process of mounting electronic parts onto a circuit board by heating up board and component and then melting the soldering tin into the joint. A good temperature for the soldering iron is between 300ř and 350ř celsius. The iron should not be pressed onto the joint for too long, because there is a risk of destroying the component if it is sensitive to heat.

5.4 Power Supply and Case

For the power unit a universal power supply circuit was chosen from Robert Sontheimers audio circuit technology book (Sontheimer, 2004, p. 74) and mounted onto a perfboard. Instead of the 7815 and 7915 voltage regulator ICs the 7812 and 7912 were used in order to get a ś12 volt power supply with a center tap for the ground. The modules can be connected to the four male 16-pin flat ribbon connectors, that were added to make the power supply compliant to the Euro-Rack standard. Another possibility would be to make a flying bus board by attaching those connectors to a flat ribbon cable that lies in the case. Or even just fix female connectors to the cable and plug them directly into the modules. Additionally it is planned to add an IEC socket and a power switch for comfortable on and off switching and more steady starting current.

The case is a simple rack constructed from a few pieces of wood that are held together by 19 inch rails equipped with thread rails to fasten the modules.

5.5 Front Panels

The panels for all modules were made from pre-cut aluminum plates with a white varnish. The labels for knobs and banana sockets are printed on the plates with a method, that is similar to homemade circuit board etching[1]. A mirror-inverted label template (see figure 5.1) is printed onto a piece of high gloss paper for inkjet printers - but with a laser printer. It is cut and placed face down onto the upper side of the panel. By thoroughly pressing down a hot flat iron (for ironing clothes) onto the panel for a few minutes, the toner cartridge particles move to the panel. The paper residues need to be removed by placing the panel in some water and rubbing them off with a sponge. Afterwards the panel is sealed with transparent lacquer. Once the panel is dried, the holes for the knobs, switches, etc. can be prepunched and drilled. Lastly all boreholes are deburred.

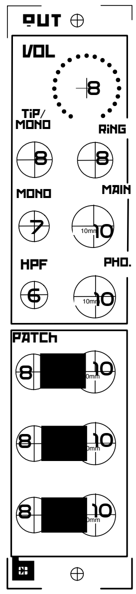

Figure 5.1: Output module template

5.6 BF-22 Filter

This module (shown in figure 5.2) is an extended copy of the filter from the legendary Korg MS-20 and is based upon the principle of the Sallen-Key filter (see chapter 4.2.2). It combines two linkable filter stages in one module. Each stage features cutoff and resonance knobs and the corresponding CV inputs, whereas the cutoff frequency input can be attenuated and inverted with one knob representing modulation depth (labeled: X-1 ... 0 ... X1). The HP/LP switch determines, if the filter is used in high pass or low pass mode.

When turning resonance up, at one point the filter begins to self-resonate at the given cutoff frequency, which means that the filter can also be utilized as an oscillator.

Figure 5.2: BF-22 filter module

[1]For etching the template would be transferred onto a copper board and put into an acid bath.

Therefore a volt per octave input for the cutoff control voltage was added, to be able to control the oscillating frequency in a musical context.

A look at the oscilloscope shows a sine like waveform. Turning the resonance to the maximum, the filter goes into distortion and the wave becomes more square causing the sound to get more rough. The amount of distortion is visually represented by a red LED.

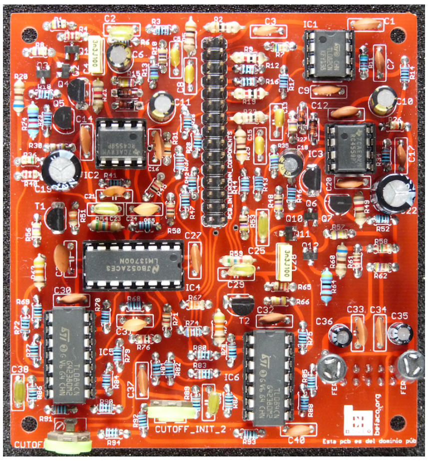

Figure 5.3: BF-22 main circuit board

Below the front panel sits an interface board where the knobs, switches, LEDs and banana sockets are mounted to. A thirty pin socket connects the interface board to the main circuit board (see figure 5.3), which contains the signal processing circuitry. The socket for the power supply sits on the very bottom.

5.7 Output

The output module is responsible for transforming the *hot*[2] synthesizer signal to a standard line level. It can be used in a dual mono mode, picking up two separate output signals from tip and ring of the 6.35 mm output jack. The mono switch splits the signal to tip and ring. The phones output provides the output signal optimized for headphones.

Additionally the module features a switch which activates a high pass filter. The idea is to be able to protect speaker systems from high energy bass signals. In the lower section 3 banana to jack patch sockets were added.

[2]The term hot signal is used commonly to describe signals with a relatively high voltage, in this case ś5 volts.

5.8 MIDI Input

This module is still at an experimental stage. The idea was to use a digital microprocessor to be able to connect a keyboard to the synthesizer and play it via MIDI. To achieve this the implementation of an Arduino Uno was planned. Arduino is an open-source electronics prototyping platform with a ATMega168 microprocessor at its heart.

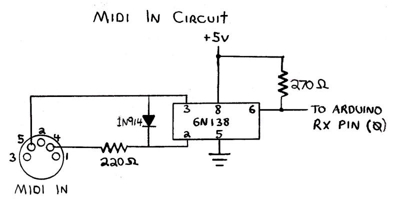

Figure 5.4: Arduino MIDI input circuit

Conveniently the Arduino must be provided with a 9-12 volt power supply, which made the process of connecting it to the synthesizers power supply easy. Only the appropriate plug connection had to be put together.

The MIDI input communication has been established by the usage of a 6N138 optocoupler, that isolates the Arduino circuit from the MIDI device (see figure 5.4). On the software side the Arduino MIDI Library v3.2 has been included, which takes care of interpreting the serial digital data arriving at the Arduino's RX pin.

What needs to be done next is converting the desired MIDI messages into control voltages. The gate voltage would be easy to implement, because it can be represented by a digital on/off state. The realization of the pitch control voltage presents itself as more complex, since the Arduino cannot output analog voltage values at its output pins but only digital states. One way would be to create a high frequency pulse width modulated signal and adding a low pass filter to convert it to a stationary voltage. This may have the disadvantage of getting a slight portamento effect though. The other way would be to convert the possible 128 note values of the note on message into seven binary values. These would be transmitted onto seven respective output pins. A digital to analog converter (DAC) would then generate the analog output voltage.

Regardless of which method is chosen, at the end the voltage should be scaled to conform the 1 volt per octave interface.

Summary

This paper is the result of a research on the topics of history, theory and electronics of analog synthesis. Of course in all of the discussed subjects only the surfaces could be scratched. Even only the field of active filter circuitry can (and does) fill entire books. Accompanying this research a small synthesizer prototype has been set up, making use of the knowledge acquired during the research.

Results

The objective to create a functional prototype can be considered a complete success. The synthesizer that was built is a lot of fun to experiment with, which was one of the initial goals also. Even though it is practically just a self oscillating filter, it offers a lot of possibilities for experimental sound design. Due to the fact that the banana plugs are stackable, the two output signals can be split several times to voltage control resonance and cutoff of each filter stage. Also various types of feedback can be generated.

This project provides a great basis for further sound experimentation. Because of its modular setup, there is a lot of space for further modules, although the requirements for power supply and case size will increase with the number of modules. The next step will be the construction of an oscillator module as soon as time and monetary means allow it.

A Personal Journey

As a trained programmer and web application developer the field of electronic engineering always seemed appealing to me. Hence the assignment for a research paper during the audio engineering course at SAE Institute seemed like a welcome opportunity to dive into the realm of building electronic devices in the context of sound generation and modification.

The process of writing this paper has been an unexpectedly rewarding and inspiring experience, pushing the boundaries of my own musical and technical understanding. Most notably the concepts of free composition - meaning allowing randomness and therefore putting oneself in the position of reacting to a musical system, influencing it in terms of tendencies, rather than controlling it with a predetermined mindset - has been something that really changed my perception of musical creativity. This for me seems much more attainable in the analog world, where electrical components and signal chains can be brought to their tipping points, resulting in an unpredictable outcome. That is where sound exploration begins, which is a totally different experience than knowing what will happen. Virtual digital environments, which I was familiar with on the other hand, generally seem to tend persuade the user to feel in control at all times.

Acknowledgment

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List of Figures

Anwander, Florian: *Synthesizer - So funktioniert elektronische Klangerzeugung*. 6. edition. PPV Medien GmbH, Bergkirchen, Germany, 2011.

Bode, Harald: *History of electronic sound modification*. Journal of the Audio Engineering Society volume 32(10) pp. 730–736, 1984.

Dunn, David: *A History of Electronic Music Pioneers*. Eigenwelt der Apparate-Welt.(Katalog) pp. 21–62, 1992.

Fjellestad, Hans: *Moog*. Plexifilm, Asheville, USA. [DVD], 2004.

Friesecke, Andreas: *Die Audio-Enzyklopädie: Ein Nachschlagewerk für Tontechniker*. KG Saur Verlag Gmbh & Company, München, Germany, 2007.

Henry, Thomas: *The Noise Generator Cookbook*. 2. edition. Magic Smoke Electronics, San Jose, USA, 2009.

Humpert, Hans Ulrich: *Elektronische Musik*. Schott, Wien, Austria, 1987.

Hutchins, Bernie A.: *Musical Engineer's Handbook - Musical Engineering for Electronic Music*. 1. edition. Electronotes, Ithaca, USA, 1975.

Lancaster, Don: *Active-filter cookbook*. 1. edition. H. W. Sams, Indianapolis, USA, 1975.

Manning, Peter: *Electronic and computer music*. Clarendon Press, Oxford, UK, 1985.

Ruschkowski, Andre: *Soundscapes - elektronische Klangerzeugung und Musik*. 1. edition. Lied der Zeit, Berlin, Germany, 1990.

Sontheimer, Robert: *Audio-Schaltungstechnik*. Elektor, Aachen, Germany, 2004.

Declaration of academic honesty

I hereby declare that in the attached submission I have not presented anyone else's work, in whole or in part, as my own using only the admitted resources. Where I have taken advantage of the work of others, I have given full acknowledgement.

Place, Date　　　　　　　　Signature