Prototyping a Modular Analog Synthesizer

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Motivation

The project was inspired by the film *moog* Fjellestad (2004), a documentary about Dr. Robert Moog, electronic instrument pioneer and inventor. Its goal is to convey an understanding of the inner workings of electronic synthesizers and their components. 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, let alone building electronics devices before.

Chapter Overview

The first chapter 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.

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 predetermed mindset - has been something that really changed my perseption 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.

2.1 Early Development Milestones

Around 1900 american Thaddeus Cahill initiated a new era of music by inventing a 200 ton machine known as the Dynamophone or Thelharmonium (Humpert, 1987, p. 19). Working against incredible technical difficulties, he succeeded to create an electrical sound generator, that 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, 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). 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 central figures in the history of electronic music, since he was also one of the primary engineers 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 Baer (2013).

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, mixers and others (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, was selling kits of transistorized Theremins (Dunn, 1992, p. 20). As he states in the movie about him Fjellestad (2004), 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 carreer in electronic music experimentation, and became increasingly dissatisfied with the severe limitations of traditional equipment at the San Francisco Tape Music Center, where they were working. They sought out to hire a competent engineer and met Donald Buchla (Manning, 1985, p. 117-118), Dunn, 1992, p. 22. 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. 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).

In 1966 Bob Moogs 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 adviced Bob Moog while perfecting his system, worked with Benjamin Folkman to produce an album of titles by Johann Sebastian Bach interpreted only with Moog synthesizers. With the title *Switched-on Bach* they demonstrated the performance of the system so convincingly, that they hit the popmusic 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 manufactrurers 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).

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 oscillate in a periodic 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 periodic 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. The pitch of the output signal is defined by the frequency of the wave and must oscillate between 20Hz and 20kHz 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.

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 trompet-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. 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, which generate random non-periodic frequencies. Therefore the signal contains no tonal information.

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 colour 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 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 endede

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 slew rate sets the slope of the filter - meaning how abrubt frequencies are being cut.

Filters can generally be devided into two categories: Low pass and high pass filters (also called high cut and low cut). To get a bandpass filter, low- and high pass are connected in series. When connected parallely, they become a bandstop or bandreject filter. Lastly the allpass 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 dependend upon a manual interaction except for their programming and activation.

3.4 The Modular Approach

A modular synthesizer is an electronic 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 connected together with patchchords. The second essential aspect is the concept of intermodular controllability, with which modules may modulate or regulate the behaviour of other modules.

4.1 General

In the context of audio electronics it is important to be know that sound signals are nothing but currents.

Voltage

Control Voltage Audio Signal

Current

Rotary Knob

intermodular stuff like buffering

4.2 Modules

4.2.1 Oscillator

Natural Resonance

The most basic form of an oscillator circuit producing a sawtooth signal can be realized with just a few parts. A current charges a capacitor at a certain rate. Between the electrodes of the capacitor a voltage potential rises. 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 continiously, wheras the rate of the repetition determines the frequency of the generated signal.

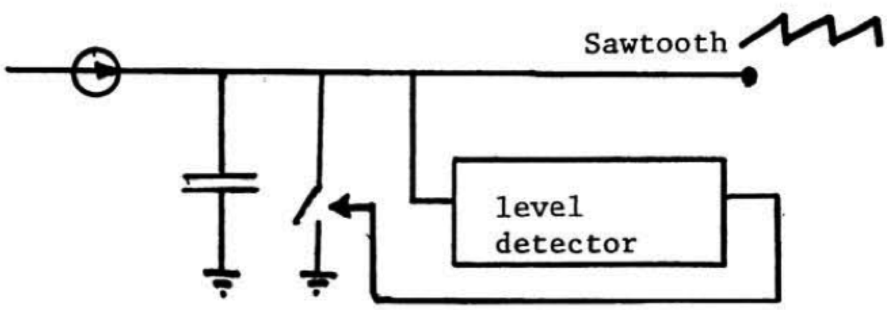

Figure 4.1: Abstract circuit scheme of a basic sawtooth oscillator

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

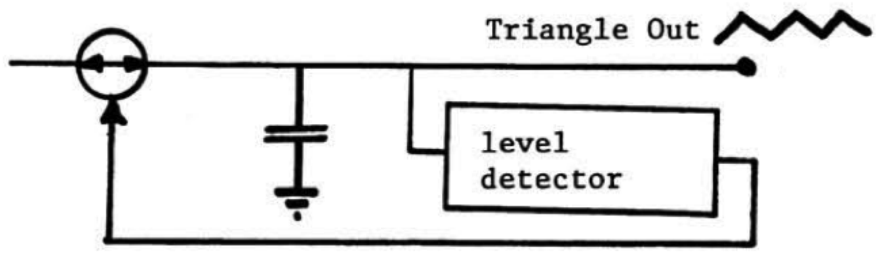

Figure 4.2: Circuit scheme of a simple triangle oscillator

Either of these two waveforms can be waveshaped into to other common basic waveforms. All required waveshaping 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. In order to get a purer sine wave, the harmonics can be filtered out with a VCF 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 VCO 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

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. This is referred to as a resistor-capacitor (RC) element.

4.2.3 Amplifier

4.2.4 Envelope Generator

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 perf boards 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 the author found out about a modular synthesizer building workshop taking place in berlin monthly. It is organized by a spanish collective from barcelona called *befaco* (http://befaco.org/). At the workshop it was possible to acquire various module kits containing all necessary parts and also receive tips and support while assembling them.

Since the budget for this project was limited, 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. A different size format often used in the DIY modular synth scene is the one the serge synthesizers use. They use banana jack connectors instead of mini jack for patching, which have the possibility of stacking banana connectors on top of each other and splitting the signal without having to use a multiplier module. For this project a combination was chosen: The modules are EuroRack size, but using banana plugs.

5.3 Building and Testing

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 an audio circuit technology book (Sontheimer, 2004, p. 74) and mounted onto a perf board. 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 EuroRack 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 to it for more 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 Frontpanels

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. A mirror-inverted label template 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 borholes are deburred.

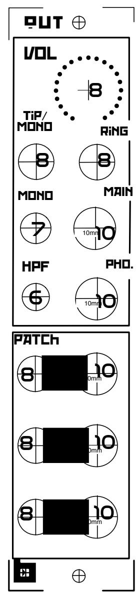

Figure 5.1: Output module template

5.6 BF-22 Filter

This module is an extended copy of the filter from the legendary Korg MS-20 and is based upon the principle of the sallen and key filter. It combines two linkable filter stages in one module. Each stage features cutoff and frequency knobs, as well as several voltage control inputs for cutoff frequency and resonance, whereas the cutoff frequency input can be attenuated and inverted with one knob representing modulation depth (labeled: Œ-1 ... 0 ... Œ1). 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 used as an oscillator. 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 with few overtones. 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.

5.7 Midi Input

Note Source

5.8 Output

describe the journey, discribe the difference and natururality of analog sound as opposed to the digital, which i only knew before.

tweaking knobs to borders where the outcome is on a threshold resulting in unpredictable patterns.

Thanks to Eddi, Derek, Befaco, Richard, David

List of figures

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

Baer, Christopher T.: *From 'Home Sweet Home' to 'Karn Evil 9' : Hagley Documents the Origins of the Synthesizer > What Are DVAG Folks Up To?* fetched 30th of March 2013. URL `http://www.dvarchivists.org/wire/items/view/507`.

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*. ders.(Hrsg.), Eigenwelt der Apparate-Welt.(Katalog), Linz pp. 21–62, 1992.

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

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

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

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

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

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

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My signature below constitutes my pledge that all of the writing is my own work, with the exception of those portions which are properly documented.