Prototyping a Modular Analog Synthesizer

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March 24, 2013

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Course: AED412
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Berlin, Germany 2012

Table of Contents

About this paper

Description

This paper describes an attempt to design and assemble a basic monophonic synthesizer prototype consisting of some standard modules that are to be found in virtually every classical synthesizer device, such as an oscillator, an envelope, and a filter.

The first sections represent the research on the history and theoretical background of analog synthesizers in general and modular systems in particular. These findings are applied to building an experimental device. First, different circuit concepts will be introduced for each module, so that the most suitable ones can be identified, whereby comprehensibility and prices of electronic components play a significant role in the choice of a circuit design. The process of building the prototype includes working with an oscilloscope to examine and verify the shape of various waveforms before and after modulation.

To make it playable with a keyboard, a MIDI input module is added. It features an Arduino microprocessor to convert digital MIDI messages into control voltage outputs that other modules can connect to. It is the only digital component of the synthesizer, while tone generation and processing are analog.

Motivation and Goal

The project was inspired by the film *moog*, a documentary about Dr. Robert Moog, electronic instrument pioneer and inventor. Its goal is to attain a better understanding of the working of electronic components and circuits as well as their influence on audio signals. Another goal is to create a functional synthesizer that is fun to play and experiment with and therefore obtain some practical experience in the field of artificial sound generation.

1 Introduction

concepts from chapter two are applicable to digital and

2 Historic Evolution of the Synthesizer

2.1 Early Development Milestones

Around 1900 american Thadedeus Cahill initiated a new era of music by inventing a 200 ton machine known as the Dynamophone or Thelharmonium [Humpert, 1987, p. 19]. It was 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 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]

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].

Sakbutt (1948) Hugh LeCaine

Melochord (1949) and warbo formant organ (1937) H. Bode

2.2 Influencial Electronic Music Projects of the Fifties

Elektronische Musik Cologne (1952) Eimert (Stockhausen)
Poem Electronique (1958) World Fair Brussels

2.3 The First Synthesizers

RCA Synthesizer (1955)
First modular synthesizers (1945)
Moog Synthesizer (1964)

2.4 The Digital Age

Text about it.

3 Theory of subtractive synthesis

3.1 Sound 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 a vibration. 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 oscillates 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 of 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 slightly 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, p. 9]. 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 Analog synthesis

4.1 General

Voltage
Control Voltage Audio Signal
Current
Rotary Knob

4.2 Modules

intermodular stuff like buffering

4.2.1 Oscillator

4.2.2 Filter

4.2.3 Amplifier

4.2.4 Envelope Generator

4.2.5 Midi Input / Note Source

4.2.6 Output

5 Building a modular synthesizer

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 blocks. Some of these concepts will be covered later. 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 savings using this strategy are not to be underestimated.

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 patchchord 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 do-it-yourself 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.

For tuned modules it is important to consider whether they use a volts per octave or volts per hertz characteristic.

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.

Oscilloscope, Multimeter, Tracking faults, measuring

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 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. 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 volts 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.

6 Conclusion

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

Thanks to Eddi, Derek, Befaco, Richard, David

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Declaration of academic honesty