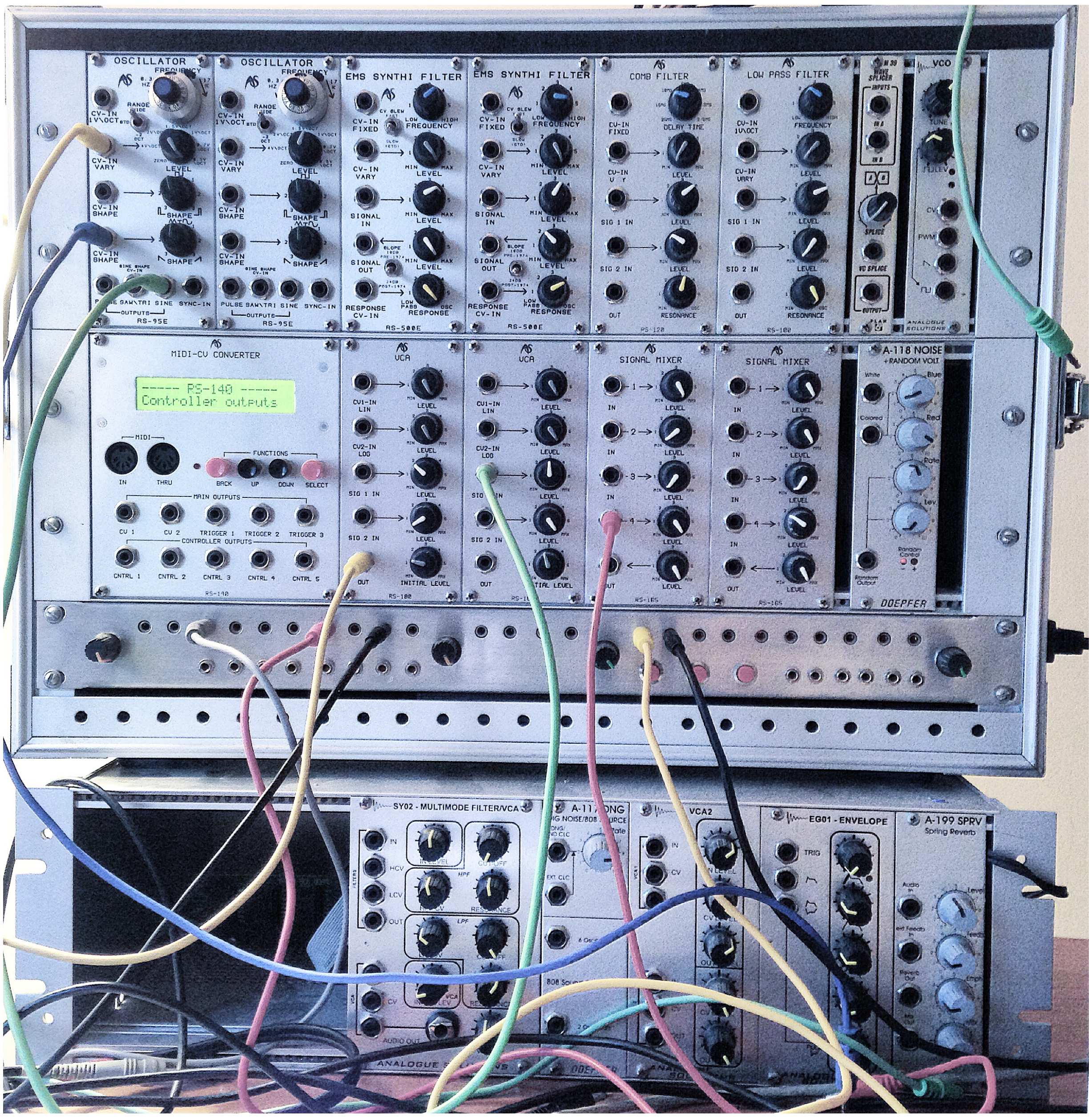
ANALOG MODULAR RESYNTHESIS OF SPECTRAL DATA USING ATS & SUPERCOLLIDER



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1 Introduction

Multi-Channel digital audio interfaces outputting control voltage have been used to control modular synthesizer parameters. Using a computer assisted calibration process for voltage-controlled oscillators (VCO), filters (VCF), and amplifiers (VCA), analog modular components can be controlled for effective resynthesis of complex sounds as specified by the ATS spectral model. A process and user interface has been developed to aid in fast and accurate calibration for modular resynthesis. Additionally, a set of classes have been developed to facilitate efficient resynthesis of ATS sound analysis. Experimental results demonstrate the proposed method reproduces sinusoidal partial data virtually indistinguishable from digital resynthesis. While residual noise band resynthesis bore a strong resemblance to the digital resynthesis, the noise components have some added modular artifacts. Such a process also allows for the inclusion of modular-specific timbres.

1.1 Background

ATS is a spectral modeling library that analyzes and represents sound as a sum of time varying partials and critical band noise. ATS Spectral data is accesses by way of an API. In this implementation, module calibration was developed in SuperCollider, a realtime synthesis environment. This proposed process provides the means to synthesize ATS sound analysis in the analog modular domain where oscillators resynthesize partial data and filtered noise reproduces critical band residual components.

1.2 Process Diagram

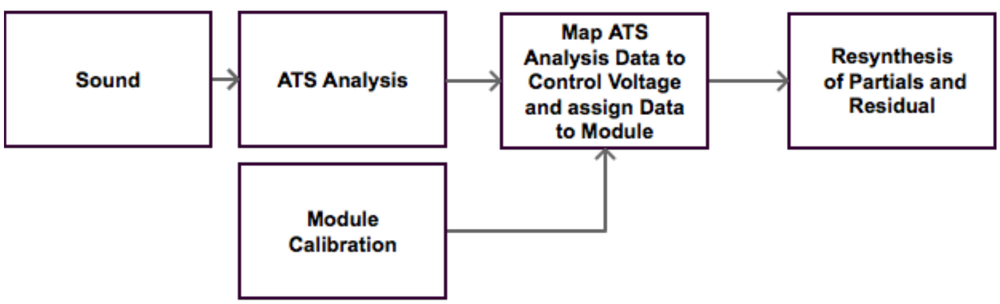


Figure 1: The sequence of steps to calibrate and resynthesize ATS spectral data

1.3 Component Architecture

The proposed method for partial resynthesis has a component architecture consisting of a computer/DAW with DC-coupled outputs sending control voltage to and receiving audio from *n* VCO/VCA pairs. Each sound generating/attenuating module pair represents a single partial as specified by the ATS spectral model.

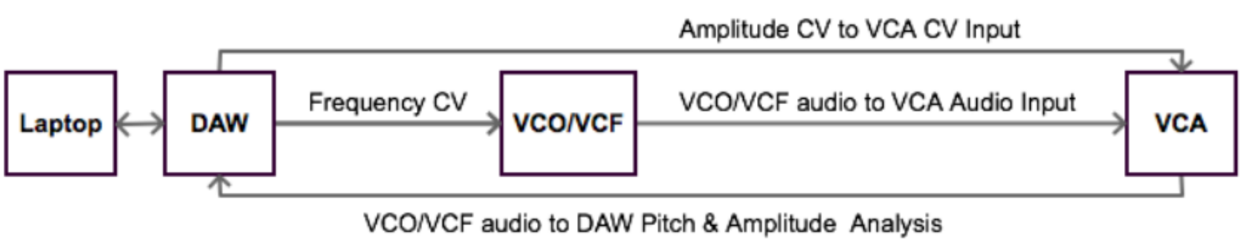


Figure 2: The component architecture for calibrating and synthesizing a single partial

2.0 Calibration

As each modular component has parameter values, ranges and scales independent of one another, there is a need to calibrate each module to the computer/DAW outputting CV. Component calibration is a sequential process consisting of the following four steps:

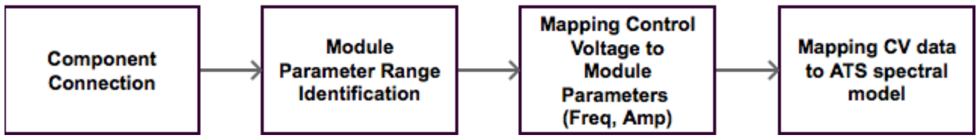


Figure 3

2.1 Calibration Constraints

For accurate resynthesis, all process steps must accommodate module stability constraints:

- The calibration process should be brief to ensure parameters do not "drift" due to temperature changes or other uncertainties.
- The maximum number of module calibrations should occur concurrently and/or asynchronously. Asynchronous calibration will both shorten the process by allowing other modules to have their ranges tuned while other modules are already tracking CV and module parameters.
- It was found that maintaining the same DAW/Module routing for calibration and resynthesis provided the most accurate results. Therefore, any calibration software should be capable of dynamic input and output routing.

2.2 Component Connection

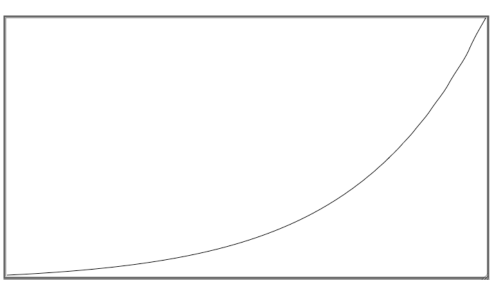
Once components are connected, they should not be re-patched during or after the calibration process as merely unplugging a jack from a module can cause parameter fluctuations such as changing the pitch or timbre. Some modules may require a "warm-up time" to allow for the parameters to stabilize.

2.3 Module Parameter Range Identification

Identifying each module's range will both verify proper component connection and define the range of spectral data the module can resynthesize. To identify the desired parameter range, the modules are tuned to the minimum and maximum DAW control voltage output. System range min and max are -1, and 1, respectively. Module range may also be constrained due to pitch/amplitude detection limits of the analysis software.

2.4 Control Voltage Mapping

To track the relationship of control voltage to a module parameter, output incrementing CV from the DAW into the module while simultaneously mapping the corresponding pitch or amplitude analysis of the module's output. When testing the calibration curve accuracy, frequency information is accurate within 0.001 hertz.



Min Freq:226.75hz ***Max Freq: 9845.32hz.***

Figure 4: Plot of control voltage and frequency curve for a Voltage controlled Oscillator. Note the slight warping on the scale in the upper frequency range.

2.5 Calibration User Interface

To expedite the synthesis process, a user interface has been developed to simultaneously and asynchronously calibrate *n* modules of both pitched sound generators and VCAs. Outgoing CV, module pitch and amplitude values are displayed to aid in range identification and to observe the calibration progress. Additionally, a button toggling between the system min and max is displayed to identify module range. Module input and CV output channels are made variable to minimize re-patching. Once the calibration is complete, a file is generated associating control voltage to frequency and/or amplitude information that can be used later in the resynthesis process.

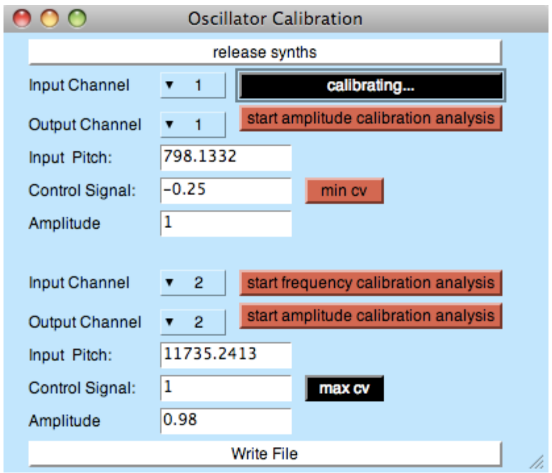


Figure 3: Screenshot of the interface for two modules during calibration the process. The first module is undergoing CV to Pitch mapping while the second module is having the maximum pitch range identified before calibration begins.

2.6 Mapping Control Voltage to Module Parameters.

Once all modules have been calibrated and the corresponding pairs of CV and parameters are stored into tables and written to a file, ATS partial data is indexed into the calibration tables to find an interpolated CV value corresponding to the CV/Parameter pairs using the partial. Eventually all of the partial and noise data is converted into corresponding Control Voltage values.

3.0 Resynthesis

Accurate resynthesis of spectral data by way of unstable analog synthesizer modules necessitates a fast and efficient process. Due to the fact that there are likely to be more partials than modular partial voices, several optimizations are introduced for minimization resynthesis time.

3.1 User Optimizations

Sorting partials and noise bands by loudness, batch resynthesis and adding audio time-stamps resolves the primary causes of increased resynthesis times and decreased accuracy.

3.2 Time-Stamp

Time-stamps identifying the onset of partial and noise information is added to the beginning of the control voltage signal output to account for partials that begin inaudibly or have been transformed to begin at different start time. An audio onset impulse enables fast alignment of partial recordings in an audio editor.

3.3 Partial Ordering

All partials are sorted by their relative loudness to enable faster editing as quiet partials may be more likely to be discarded after the resynthesis process during the mixing stage.

3.4 Batch Resynthesis:

Given that modular resysnthesis systems can synthesize a limited number of partials simultaneously, all of the spectral model's partials are assigned a module and a recording pass to minimize the number of 'overdubs'. Minimizing the number of overdubs will also reduce noise floor. It will be required to manually align partials by time stamps in a multi track audio editor

3.5 Partial Synthesis

ATS translates spectral peaks into sinusoidal trajectories consisting of time-varying frequency and amplitude data for N partials. The modular synthesizer architecture for modeling an ATS partial is a supercollider controlled VCO attenuated by a VCA. As there are often non-linearities in many modular VCO's, sinusoidal waveforms are most closely created by a VCF whose resonance oscillates at a specified frequency.

3.6 Noise Band Synthesis

The ATS spectral modeling tool analyzes the noise energy for each critical band of hearing. To model a residual component, white noise is passed through a bandpass filter or cascading lowpass and highpass filters whose output is attenuated by a VCA. Each filter frequency corresponds to the frequency boundaries of each critical band or in the case of a bandpass filter, the band's center frequency. Noise bands are synthesized sequentially, ordered by loudness with onset time-stamps for alignment in a digital audio interface.

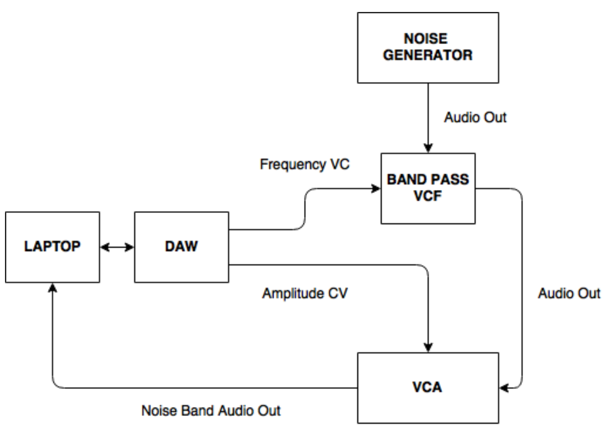


Figure 5: flowchart for single noise band with a band-pass filter architecture .

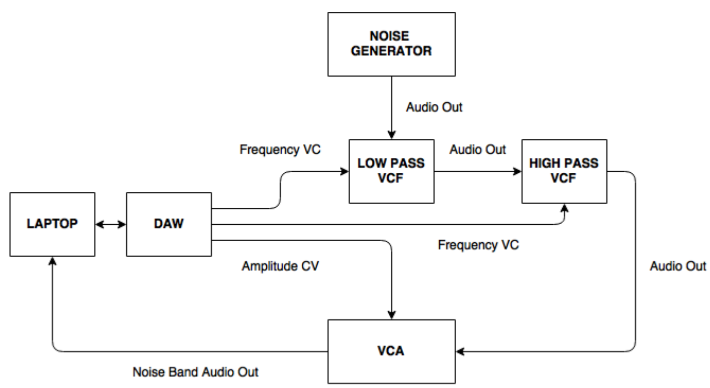


Figure 6: flowchart for single noise band with cascading highs and lowpass filter.

5.0 Results and Discussion

Observed results confirm that the proposed method can accurately resynthesize complex spectra. Analog resynthesis using sine waves is nearly identical to a digital resynthesis of the same sound, albeit with a higher noise floor and more pronounced attacks for percussive sounds. Furthermore the process allows for easy transformation at any resynthesis stage. Resynthesis of noise bands were less accurate: while the overall spectral shape was identifiable, there were notable artifacts introduced into the timbres by the modules. Bandpass filters added resonances at the center frequencies of critical bands, adding a vocoder-like timbre. For both bandpass and cascading high/lowpass residual synthesis, the filter frequency response rolloff overlapped energy into adjacent bands. It was also noted that as many VCA's are linear, it is preferable to discard the amplitude analysis and use the amplitude information directly to reduce noise introduced by the amplitude analysis.