Algorithms for Guitar-Driven Synthesis: Application to an Augmented Guitar

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ABSTRACT

This work studies the use of signal-driven synthesis algorithms applied to an augmented guitar. A robust sub-octave generator, partially modeled after a classic audio-driven monophonic guitar synthesizer design of the 1970s is presented. The performance of the proposed system is evaluated within the context of an augmented active guitar with an actuated sound box. Results of the evaluation show that the design represents an exciting augmentation for the instrument, as it radically transforms the sound of the electric guitar while remaining responsive to the full range of the guitar playing gestures.

1. INTRODUCTION

During the advent of analog synthesizers in the 1970s, companies like Electronic Music Studios (EMS) and Roland, decided to enter the much bigger electric guitar market by releasing new and exciting instruments that came to be known as "guitar synthesizers". This period saw the development of many iconic, albeit unsuccessful, products like the EMS Synthi Hi-Fli, a guitar multi-effects processor capable of producing distinctive synthesizer sounds [1], and the Roland GR-500, a revolutionary augmented guitar with integrated effects and the capability to control external systems via a frequency-to-voltage (F–V) converter [1]. The Roland GR-500 guitar synthesizer can be heard, for instance, in David Bowie's 1980 hit single "Ashes to Ashes" [2].

Analog guitar synthesizers can be divided into two categories according to their operating principle: those that track the pitch of the input signal via F–V conversion (e.g. using a phase-locked loop (PLL)) and use it to control an external synthesis engine [3], and those that process the incoming signal to produce a "synthesizer effect" [4]. As described by Puckette [5], guitar synthesizers that belong to the former category are known to make many audible mistakes, such as octave jumps, and suffer from high levels of latency due to the time it takes for the circuit to track the pitch of the incoming signal. This latency can be heard as a distinctive *glissando* effect which may not always be desirable. Popular guitar synthesizers that fall within this category include the ARP Avatar [4], and the external signal processor section in the Korg MS-20 monophonic synthesizer [6].

This work deals with the second category of guitar synthesizers, i.e. those based on sound modification. We study the behavior of one of the most common processing techniques in this type of synthesis, sub-octave generation, and present a model for its use in digital guitar-driven synthesis. The proposed algorithm is partially based on the sub-octave circuit found in the Electro Harmonix (EHX) Micro Synth, an analog guitar synthesizer designed by David Cockerell [2] in 1978 [7]. Therefore, this research can be classified as belonging to the field of "virtual analog" modeling [8].

Previous research on audio-driven digital synthesis has investigated the use of techniques such as self-modulating and adaptive FM synthesis [9–11]. The use of PLL structures in the digital domain has also been investigated in the context pitch tracking of electric guitar signals [12]. Additionally, since the proposed algorithms are based on the study of an analog guitar pedal, this study falls in line with a long tradition of virtual analog modeling of guitar effects, e.g. [13–15].

The proposed algorithm was implemented in real-time using Max/MSP and evaluated within the context of an augmented electric guitar. Instrument augmentation is an established research field which seeks to add novel sonic and gestural features to an existing musical instrument via signal processing and sensor technology [16]. Due to the rise in popularity of guitar augmentation research, these instruments provide an ideal platform for the incorporation of classic analog-style synthesis and processing techniques. A selection of augmented guitar research papers includes sensor-controlled audio effect processing [17], signal-driven adaptive audio effects [18], individual string processing [19], and networked guitar extended towards mobile devices and virtual reality [20].

More recently, research attention has been drawn towards the augmentation of acoustic guitars, with integrated audio transducers that enable to diffuse electronic and processed sounds through the instrument's soundboard and sound box. This approach, referred herein as "active acoustic in-

[1] http://www.matrixsynth.com/2008/11/ems-synthi-hi-fli.html

[2] Cockerell was also responsible for the design of several classic EMS synthesizers, including the VCS3, the Synthi 100, and the Hi-Fli.

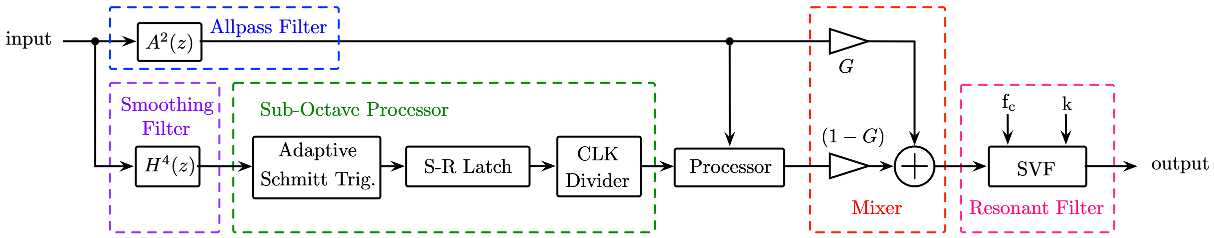

Figure 1. Block diagram of the proposed guitar-driven synthesizer. This design is partially based on the sub-octave generator section of the Electro Harmonix Micro Synth guitar pedal. Gain parameter $G \in [0, 1]$ controls the wet/dry mix.

struments" has been recently gaining momentum, giving rise to several projects, both in the start-up scene and in the Academia, such as IRCAM's *SmartInstruments* [3] project giving rise to the *Hyvibe* guitar [4]. Other prominent augmented guitar start-ups include the *Sensus* guitar [21] and the *ToneWoodAmp* [5] outboard device.

This work is organized as follows. Section 2 details the derivation and implementation of the proposed guitar-driven synthesizer. Section 3 discusses and evaluates the incorporation of the proposed algorithm within the context of an augmented guitar. Finally, Section 4 provides concluding remarks.

2. SUB-OCTAVE GENERATOR

Figure 1 shows a block diagram representation of the digital guitar-driven synthesizer proposed in this study. The proposed design is intended for monophonic input signals only and is built around a sub-octave generator algorithm partially based on the aforementioned EHX Micro Synth [7], which also shares some resemblance to the Boss OC-2 Octave guitar pedal from 1982 [22].

The overall principle behind sub-octave synthesis is to estimate the fundamental frequency of the incoming input signal by generating a rectangular clock signal of the same frequency. The estimated clock is then shifted in frequency by half to produce the desired sub-octave signal. Devices that output this rectangular oscillator for musical use are known as "direct dividers" [23]. On the other hand, "indirect dividers" are those that utilize the estimated sub-octave oscillator to apply some form of signal processing to the input signal. In these cases, the resulting waveform preserves some of the timbral qualities of the original input signal but is perceived to be an octave below in frequency. The system herein presented falls under this category. The following subsections present the details behind the behavior and implementation of each of the blocks in Fig. 1.

2.1 Smoothing and Allpass Filtering Stages

The first step required to estimate the fundamental frequency of the incoming input signal is to remove as many harmonics as possible using a smoothing filter [5]. Since the objective of this work is to apply the proposed algorithm to an augmented guitar featuring a single electromagnetic pickup (cf. Sec. 3), the cut-off frequency of this filter was chosen according to the frequency range of the whole instrument. We propose the use of four identical one-pole lowpass filters arranged in series with a cut-off frequency at 1 kHz. This parameter was chosen empirically based on tests performed on the final implementation. The proposed cascade of filters exhibits a combined spectral roll-off of approx. 24 dB/octave. Comparably, the EHX Micro Synth use a three-pole (18 dB/octave) design with a cut-off at approximately 1.1 kHz [7].

The transfer function of a single one-pole lowpass filter can be written in the Laplace domain as

$$H(s) = \frac{w_c}{s + w_c} \tag{1}$$

where $w_c = 2\pi f_c$ and f_c represent the cut-off frequency in radians/second and Hz, respectively. This continuous-time transfer function can be then discretized using the bilinear transform, which gives us

$$H(z) = \frac{b_0 + b_1 z^{-1}}{1 + a_1 z^{-1}}, \tag{2}$$

where

$$b_0 = b_1 = \frac{w_c T}{2 + w_c T} \approx 0.0317 \tag{3}$$

$$a_1 = \frac{w_c T - 2}{w_c T + 2} \approx -0.9366. \tag{4}$$

Parameter T is the sampling period of the system, i.e. $T = 1/F_s$. Here, we have defined a sample rate $F_s = 96$ kHz which will be used throughout this study. Due to the relatively low cut-off parameter we neglect the warping effects of the bilinear transform.

In applications where access to individual string signals is available, e.g. via hexaphonic pickups, multiple sub-octave processors can be used simultaneously [5]. This approach can be used to adapt the proposed system for polyphonic use. In this scenario the cut-off frequency of each filtering stage should be selected according to the range of each string.

Now, the group delay introduced by the proposed smoothing filter can be compensated in the direct signal path using two first-order allpass filters, each with the z-domain transfer function

$$A(z) = \frac{a_1 + z^{-1}}{1 + a_1 z^{-1}}, \tag{5}$$

[3] http://instrum.ircam.fr/smartinstruments/
[4] https://hyvibe.audio
[5] www.tonewoodamp.com

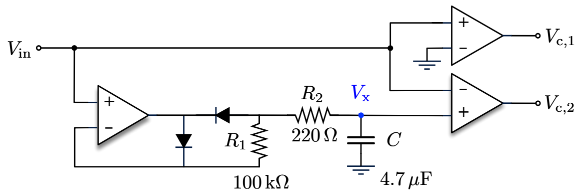

Figure 2. Adaptive Schmitt trigger circuit. Figure adapted from [7] and [24].

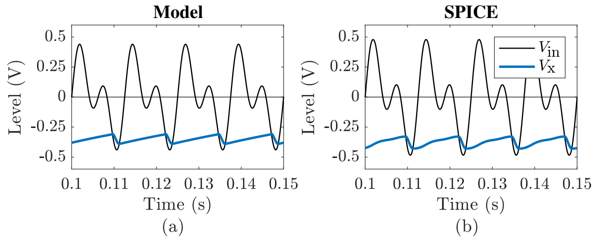

Figure 3. Output of the peak follower used in the adaptive Schmitt trigger for an arbitrary input signal implemented (a) with the proposed model and (b) using SPICE.

where a_1 determines the pole location and remains as defined in (4). This allpass filter will ensure the direct (or "dry") signal path is perfectly aligned with the output of the clock divider stage. This point is important to the proposed system and is revisited throughout this work.

2.2 Adaptive Schmitt Trigger

Once the input signal has been processed by the smoothing filter, we can estimate its fundamental frequency. If we consider an ideal scenario in which the signal being processed is a pure sinusoid, a rectangular clock signal of the same frequency can be extracted trivially using a comparator in the analog domain, or the *signum* function in the digital one. In this scenario the resulting clock signal will be perfectly in-phase with the input waveform. Unfortunately, when dealing with real-world signals, such as those produced by an electric guitar, these approaches can lead to severe mistakes in the estimation process. This is mostly due to the sensitivity of trivial methods to small fluctuations in input signal values.

Some analog devices avoid this issue by using a structure known as a "Schmitt trigger" [25]. Figure 2 shows the circuit diagram of the Schmitt trigger modeled in this study. This circuit is partially based on the design found in the EHX Micro Synth and Boss OC-2 guitar pedals [7,22], where a hysteresis loop is created by comparing the input signal against two peak followers. In Fig. 2, we have replaced one of the peak followers with a trivial comparator in order to avoid introducing a phase mismatch between the input waveform and the estimated clock signal. This same idea is used in the "U-Boat" guitar pedal, a DIY suboctave processor designed by Merlin Blencowe [24]. Since the hysteresis loop in this type of circuits changes depending on the input level, this structure is called an "adaptive Schmitt trigger".

Rather than recurring to detailed circuit analysis tools, in this study we propose a behavioral model of the circuit. First, we derive an algorithm for the output of the negative peak follower, labeled V_{x} in Fig. 2. This circuit is essentially an RC filter in which the value of the resistance and the driving factor change according to the state of the circuit. The design utilizes switching diodes to change between the modes of operation of the system. In continuous-time, the input–output relationship of an RC filter can be written as

$$\frac{\mathrm{d}V_{\text{x}}}{\mathrm{d}t} = \frac{1}{RC}\left(V_{\text{in}}(t) - V_{\text{x}}(t)\right), \tag{6}$$

where t is time. This expression is equivalent to (1) when $RC = 1/w_{\text{c}}$. Equation (6) can be discretized using the trapezoidal rule which gives us

$$V_{\text{x}}[n] = \frac{\alpha\left(V_{\text{in}}[n] + V_{\text{in}}[n-1]\right) + (1-\alpha)\,V_{\text{x}}[n-1]}{1+\alpha}, \tag{7}$$

where $\alpha = T/(2RC)$. This expression is equivalent to (2).

Listing 1 shows a Matlab implementation of the negative peak follower in Fig. 2. The value of V_x is estimated using (7) directly, with the value of the RC factor and driving term determined by whether the input signal is higher or lower than the previous output value, i.e. the state of the filter.

```
% Declare states and variables
R1 = 10e3;  R2 = 220; C = 4.7e-6;

for n=2:length(Vin)

  if (Vx(n-1) >= Vin(n))
    RC = R2*C;    x(n) = Vin(n);
  else
    RC = R1*C;    x(n) = 0;
  end

  % Implement difference equation (7)
  a = T/(2*RC);
  Vx(n)=(a*(x(n)+x(n-1))+(1-a)*Vx(n-1))/(1+a);
end
```

Listing 1. Matlab implementation of the negative peak follower in Fig. 2.

Figure 3(a) shows the result of implementing the proposed algorithm for an arbitrary periodic input signal. As expected, the algorithm tracks the negative peaks of the incoming signal. The results of simulating the circuit using SPICE, an open-source, general-purpose circuit simulation tool, are shown in Fig. 3(b), which indicate a good approximation between the original circuit and the proposed model.

Having computed the value of the signal at node V_{x}, we proceed to implement the comparator stages shown on the right-hand side of Fig. 2. The value of signals $V_{\text{c},1}$ and $V_{\text{c},2}$ can be written explicitly as

$$V_{\text{c},1} = \operatorname{sgn}(V_{\text{in}}) \quad \text{and} \quad V_{\text{c},2} = \begin{cases} 1 & , V_{\text{in}} < V_{\text{x}} \\ -1 & , \text{otherwise}, \end{cases}$$

where sgn( ) is the aforementioned *signum* function. For simplicity, we have assumed the op-amps exhibit ideal rail-to-rail behavior and have 1-V bipolar power supplies. Figures 4(a) and 4(b) show the result of computing $V_{\text{c},1}$ and

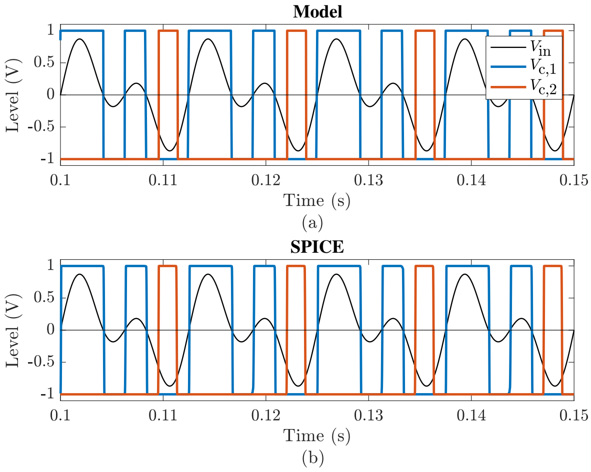

Figure 4. Output of the adaptive Schmitt trigger for an arbitrary input implemented (a) with the proposed model and (b) using SPICE.

$V_{c,2}$ using the proposed model and a SPICE simulation, respectively. As shown in these plots, the adaptive Schmitt trigger outputs positive values for portions of the input signal above zero and below the output of the peak follower.

2.3 S-R Latch

Following the adaptive Schmitt trigger, $V_{c,1}$ and $V_{c,2}$ are fed into an S-R (set-reset) latch whose purpose is to generate a clock signal whose fundamental frequency matches that of the input. To do so, the latch outputs high and low states at rising edge transitions in $V_{c,1}$ and $V_{c,2}$, respectively. Listing 2 shows a simple behavioral emulation of this stage implemented in Matlab. Figures 5(a) and (b) show the estimated clock signal for the same input signal used in the previous examples simulated using the proposed algorithm and in SPICE, respectively. In this example we can observe how the clock signal generated by the system is perfectly aligned with the input signal (after the allpass filter stage) and has the same fundamental frequency.

```
for n=2:length(Vin)

    if VC1(n) > VC1(n-1)
        CLK(n) = 1;
    elseif VC2(n) > VC2(n-1)
        CLK(n) = 0;
    else
        CLK(n) = CLK(n-1);
    end

end
```

Listing 2. Matlab implementation of the S-R latch stage in the EHX Micro Synth pedal.

2.4 Clock Divider

The estimated clock signal is then fed to a clock divider in order to produce the required sub-octave oscillator. In analog circuits, clock dividers are typically implemented using standard D-type flip flops such as the one shown in Fig. 6. Listing 3 shows a proposed Matlab implementation of this

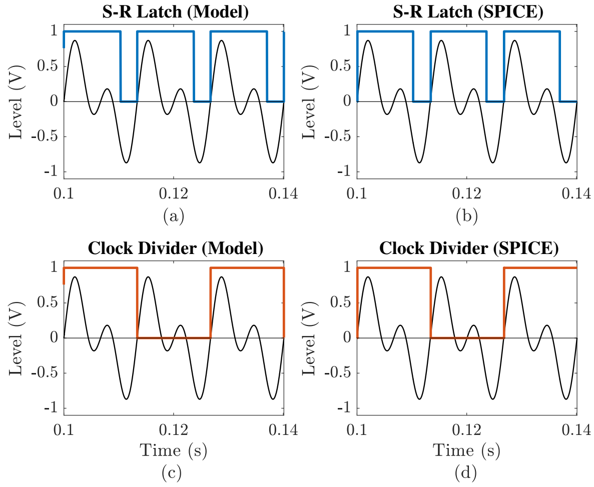

Figure 5. Output of the S-R Latch and Clock Divider stages implemented (a) & (c) using the proposed digital models, and (b) & (d) in SPICE. The black line represents the input signal used in these simulations.

system which can be used to half the frequency of the input clock. The signal `CLK` represents the output of the previous stage and the input to the flip flop. The output signal, `CLKdiv`, can be read from either the Q or $\bar{Q}$ terminals. Figures 5(c) and (d) show the output of the clock divider simulated using the proposed algorithm and SPICE, respectively. For the latter, a model of the CD4013B CMOS D-type flip flop was used. These results show the proposed model effectively emulates the behavior of the circuit.

```
% Initalize variables
Q = 1;    nQ = not(Q);   D = Q;

for n=2:length(CLK)

    % Check for rising/falling clock cycles
    if CLK(n)~=CLK(n-1)

        D = nQ;

        % Compute intermediate states
        X = not(and(D,CLK(n)));
        Y = not(and(not(D),CLK(n)));

        % Iterate until states stabilize
        for m=1:2
            tmp = not(and(X,nQ));
            nQ = not(and(Y,Q));
            Q = tmp;

            if Q~=nQ
                break;
            end
        end
    end

    CLKdiv(n) = Q;
end
```

Listing 3. Matlab implementation of the flip flop in Fig. 6.

2.5 Processor

The output of the clock divider, which represents our estimated sub-octave oscillator, is then fed to a "Processor" block where it is combined with the original input signal. This is the part of the signal processing chain where the

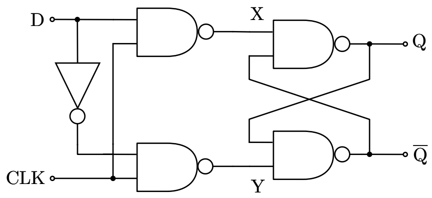

Figure 6. Internal structure of a D-type flip flop.

different analog designs differ. For example, the EHX Micro Synth implements an OTA-based "track & hold" algorithm [7], while the Boss OC-2 inverts every second waveform cycle at the peaks and applies a DC offset to smoothen the signal [22]. This second technique (detailed in [23]), is quite imperfect and can introduce numerous discontinuities in the resulting waveform.

In the proposed guitar-driven synthesis topology, we follow the work of Blencowe on the U-Boat pedal and modulate the input signal directly using the extracted sub-octave clock [24]. For this, we present two alternatives. The first one directly emulates Blencowe's design and consists on converting the clock into a bipolar signal (by scaling it by two and subtracting one) and computing its product with the input signal. This operation is equivalent to ideal ring modulation and its effect is depicted in Figs. 7(c) and 7(d) for the case of a single sinusoidal input with fundamental frequency $f_0 = 500\,\text{Hz}$. The result is a signal with a rich harmonic spectrum will increase the perceived "body" and "presence" of the original signal. As is typical in this type of modulation, the original fundamental frequency is not present at the output. Instead, the output waveform features sidebands at $f_0 \pm f_0/2\,\text{Hz}$ and odd harmonics of the sub-octave frequency.

The second technique presented in the proposed implementation is direct multiplication of the sub-octave signal with the input waveform. In the time domain, this is equivalent to removing every second cycle of the input waveform. A related approach is briefly discussed in [26]. The proposed operation is analogous to audio-rate amplitude modulation and its effect is shown in Figures 7(e) and 7(f) for the same 500-Hz input waveform. As shown in Fig. 7(f), this approach preserves the fundamental frequency of the input. In Section 3 we evaluate the sound produced by the proposed system; however, a detailed study of the frequency-domain behavior of the proposed techniques is left as future work.

Now, since in both processing scenarios the estimated sub-octave clock signal is perfectly aligned with the zero-crossings of the input signal, the modulation process will not introduce any step-like discontinuities in the output waveforms. This is thanks to both the design of the Schmitt trigger and the use of the allpass filtering stage. The main advantage of this feature is that the sounds will not be perceived as being too harsh and will preserve many of the timbral qualities of the original input signal. A second advantage is that the level of aliasing components introduced by the proposed techniques will be significantly lower than if discontinuities were introduced. A formal evaluation on

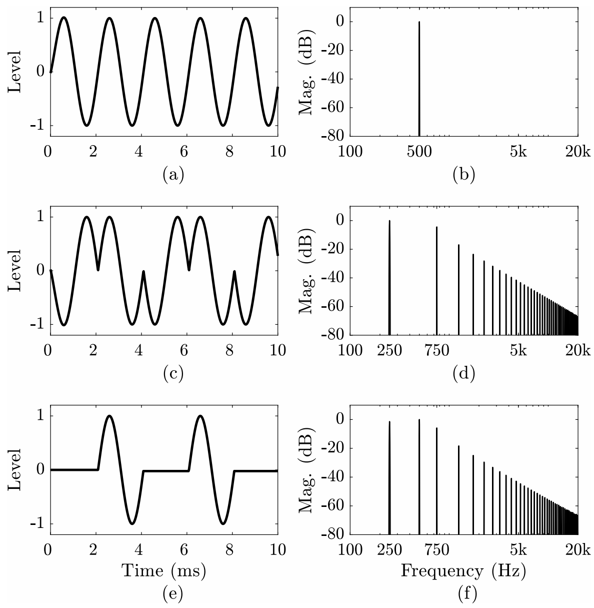

Figure 7. Waveform and normalized magnitude spectrum of (a)–(b) a 500-Hz sinusoidal signal processed by the proposed methods by (c)–(d) inverting and (e)–(f) removing every second cycle.

the audibility of aliasing in the proposed system is outside the scope of this work. Nevertheless, during the evaluation of the system no issues related to aliasing were identified. This can be attributed to the relatively low frequency of the input waveform (assumed to be below 1 kHz) and the choice of sampling rate ($F_\text{s} = 96\,\text{kHz}$).

2.6 State Variable Filter

Since the waveforms generated by the Processor block will have a rich harmonic spectrum, they can be treated in the same way as oscillators in a classic subtractive synthesizer. This process involves shaping the spectral content of these signals using a resonant lowpass filter. In the proposed topology we feed the output of the mixing stage, which simply combines the processed and unprocessed signals, to a linear state variable filter (SVF). The SVF is a second-order resonant filter commonly used in analog synthesizers because of it provides simultaneous access to the lowpass, highpass and bandpass outputs [27]. For the real-time implementation of the proposed design we used Max/MSP's native `svf~` block. Filter parameters f_c and k, which represent the cut-off frequency and resonance, respectively, were left as user-adjustable controls.

2.7 Real-time Implementation

A Max/MSP implementation of the proposed system can be found in the accompanying website: `http://research.spa.aalto.fi/publications/papers/smc18-guitar-synth/`. All algorithms, with the exception of the SVF block, were implemented using `gen~`.

3. APPLICATION TO AN AUGMENTED GUITAR

The motivation to apply the algorithms presented in this study to an augmented guitar stems from the second author's previous research on "active acoustic instruments" [28]. This research documents the design of an electric guitar with an additional actuated resonant body, resulting in a guitar which plays and sounds like a regular electric guitar, but being amplified through the instrument's own body.

3.1 An Electric Guitar with an Actuated Sound Box

The rationale for the design was to create a guitar that would possess the sonic versatility of the electric guitar and its related audio effects, combined with the integrated feel of an acoustic instrument. This design enables the instrument to be the sole sound source, bypassing external amplifier-loudspeaker modules and thus avoiding possible perceptual dichotomy arising from the spatial disjunction of the instrument and its sonic actuator. The present guitar also provides direct audiotactile feedback directly into the upper body of the player, providing multimodal connection between the player and the instrument. To the contrary, the acoustic feedback occurring in the pickup–actuator signal loop has been greatly diminished by the physical separation of the sound box from the strings and the pickup. The sound box is attached to the back of the electric guitar by points of isolating silicon glue. With this setup, acoustic feedback does not constitute a significant hindrance to the system.

The underlying design criteria for this actuated electric guitar prototype involved three main targets. #1 Playability and responsiveness: the augmented instrument should not alter the playability of the original instrument, and the augmentation should present a tight integration into the set of playing gestures of the instrument. The targeted responsiveness of the signal-driven synthesizer should allow for the totality of the right-hand plucking and left-hand control gestures to immediately translate into the synthesized sound, with minimal latency. Synthesizer parameter control should be integrated into the instrument's playing environment. #2 Sound quality: the integrated transducer-driven sound box amplifier sound quality should be comparable to a regular electric guitar amplifier. #3 Sonic originality: the guitar design and audio-driven synthesizer should allow for distinct and novel sonic possibilities on the guitar.

The prototype guitar used in this study is shown in Fig. 8. It was made jointly with the luthier Juhana Nyrhinen, combining a custom-made solid-body guitar with a Wilkinson mini P90 humbucker pickup and an added sound box extracted from an acoustic steel-string guitar. Two audio transducers were attached inside the sound box, following the placement guidelines learned from the previous "Active Acoustic Guitar" project presented in detail in [28]. In this new prototype, the processing is carried out on a laptop in order to facilitate the software development. However, an integrated processing environment is being currently studied using the Bela platform [6].

Figure 8. Electric guitar with an actuated body built by luthier Juhana Nyrhinen. Audio transducers have been mounted into the sound box, providing a guitar with an integrated amplifier/speaker.

3.2 Signal-Driven Synthesis on the Augmented Guitar

The project of implementing signal-driven synthesis on an augmented guitar stems from an aesthetic concern for providing sonic alterities for the guitar. At present, guitar processing has coagulated into a rather well established repertoire of effects, presenting limited possibilities for genuine sonic novelty. Signal-driven synthesis is regarded here as a non-established domain of sonic research (regardless of its long history), enabling to pursue an aesthetic alterity for the sound of the guitar. The fact that in the present guitar all sounds radiate from the instrument itself contributes to reinforce the impression of sonic originality. Upon seeing the familiar looking guitar object, one would expect to hear a familiar palette of sounds emanating from it. Instead, the signal-driven synthesis actuated through the guitar's sound box provides a radical shift towards an electronic-sounding guitar.

Currently, the most widespread digital guitar synthesizer solutions rely on the MIDI standard, converting audio from a hexaphonic pickup into MIDI messages driving a synthesizer. These can be seen as modern-day versions of those early guitar synthesizers which attempted (and in many occasions failed) to do the exact same thing via F–V conversion. The successive generations of MIDI guitar systems have been balancing between the promise of a tremendous extension to the sound palette of the guitar and the reality of the system latency and tracking problems, leaving the musician with a sense of frustration. The translation of the string audio to discrete MIDI messages introduces a discontinuity in the gesture-sound continuum. Information, sensitivity, and expression are lost. As a result, the player feels disconnected from the instrument.

From the perspective of the player, audio-driven synthesis is an ideal technique to extend the guitar towards an electronic soundscape. The nuances of right-hand plucking and left-hand control gestures are directly feeding the synthesis algorithms and find a direct response in the sonic output.

[6] https://bela.io/

3.3 Evaluation From a Player's Perspective

The outcome of signal-driven synthesis applied to an augmented guitar can—and should—be evaluated in terms of its instrumental quality, on the grounds of the project-specific design criteria enumerated in Section 3.1. As we are dealing with a first-round development prototype, the evaluation herein is limited to a subjective testing of the instrument within the project team and pitting it against the initial design motivations. In order to provide the reader with an impression of the sonic outcome, demo videos can be found in this paper's accompanying website.

The result regarding the design criterion #1 "playability and responsiveness" presents a high level of responsiveness on the signal-driven synthesizer to the full range of guitar playing gestures. The player can directly and seamlessly control the synthesized sound onset, timbre and duration via the strings. There is no felt gap in the gesture-sound continuum, i.e. there is no perceivable latency. The integrated feel is heightened by the audiotactile feedback from the sound box to the player's torso. Synthesizer parameters, such as the filter cut-off, resonance and dry/wet mix, need external controllers, either embedded sensors on the guitar, traditional expression pedal-type interfaces or, furthermore, audio-driven algorithms such as envelope followers or trigger detectors.

The design criterion #2 "sound quality" presents mixed results. The sound radiation from the guitarâĂŹs sound box has an integrated quality compared to an external amplifier. However, the sound box acts as a physical filter on the signal, with heavy resonant modes and calls for case-specific equalization. In our prototype, there is a general imbalance in the frequency response. The frequency region around 100 Hz is especially problematic, getting easily out of hand and forming a feedback loop. Multiband compression is being studied in order to balance the instrument's frequency response.

The goal of design criterion #3 "sonic originality" is partly attained. With specific filter settings, the signal-driven synthesizer provides original, "un-guitar-like" tones, which radically transform the electric guitar's sound and provide an engaging area for sonic exploration. With other settings, the synthesizer output resembles traditional fuzz-like sounds, or tones familiar from guitar synthesizer pedals. This is due to the similarities between sub-octave processing using clock signals and fuzz distortion processing. Both techniques imprint a square-wave-like quality on the guitar signal which consists mainly of odd harmonic elements.

Overall, the actuated electric guitar with a signal-driven synthesizer holds the advantages of an immediate synthesizer response to the totality of playing gestures, with a situated output from the instrument itself involving multimodal feedback. The main limitations of the integrated system are related to the sound quality of the actuator-sound-box system. Additional work points towards a further exploration of the potential of the system for sonic originality. This could include exploring more processing applications, either by studying classic analog designs or by proposing novel techniques, and focusing on the control aspect of the signal processing tools.

4. CONCLUSIONS

In this work we discussed the concept of guitar-driven synthesis and proposed a virtual analog sub-octave generator based on the circuits used in the EHX Micro Synth and Merlin Blencowe's U-Boat guitar pedals [7]. Two techniques for indirect division were presented: inversion and muting of alternate cycles of the input signal. The proposed system was implemented in Max/MSP and evaluated within the context of an augmented electric guitar with an actuated sound box providing sound radiation directly from the body of the instrument. The presented algorithm is considerably robust and does not exhibit major issues, such as jumps between octaves or perceivable latency. Overall, audio-driven synthesis provides a direct response to the ensemble of guitar playing gestures. Combined with the multimodal feedback of the guitar's actuated sound box, the system presents a responsive and engaging augmentation for the guitar. Future work includes an optimization of the sound quality of the actuated guitar, as well as further exploration of the sonic potential enabled by the algorithms presented in this article.

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