

A Digital Model for the Prologue Voltage Control Filter

VICTOR LAZZARINI, AND JOSEPH TIMONEY, *AES Member*

(victor.lazzarini@mu.ie)

(Joseph.Timoney@mu.ie)

Maynooth University, Maynooth, Ireland

The Prologue polyphonic synthesizer features a voltage control filter (VCF) employing an original topology. This paper examines its circuit and extracts a flowchart, which yields a set of analog filter equations. From these, the authors were able to develop a transfer function to describe its ideal linear time-invariant form. Then a linear digital version of the filter to validate the analysis was put forward. Noting that the filter operation within the synthesizer is fairly nonlinear, modifications yielding a nonlinear model of the VCF are proposed. These were compared with the outputs of the original analog filter, and it was found that they approximate them consistently.

0 INTRODUCTION

Voltage control filters (VCFs) are standard components in analog synthesizers. These were introduced originally by Robert Moog as part of his modular synthesis systems [1]. The main characteristic of these filters is the replacement of resistors by variable gain components, originally transistor pairs, then operational transconductance amplifiers (OTAs), and, less commonly, diodes and vactrols. The use of these components allow the filter cutoff frequencies to be made variable but also to make them controllable via an external voltage. This is an extremely useful property that allows programming and automation to define time-varying characteristics of the filter.

The most common filter design found in VCFs is the four-pole *ladder* type [2–7], which includes the original Moog filter as well as its OTA and diode [8] variations. The two-pole state-variable design has been also widely used [9]. These filter topologies have characteristics that make them more readily transformed from fixed or manual operation into voltage-controlled devices. In the case of the state-variable filter, for example, it is fairly obvious where resistors can be replaced by variable-gain elements.

Other filter designs, such as the Sallen–Key configuration, have been made voltage-controlled (for instance, in the case of the Korg 35 [10] and MS20 filters [11]), but they are more difficult to construct in such a way that a stable and more linear operation can be achieved. However, it is also important to note that nonlinearities exist in most VCFs and that this can be a desirable characteristic for a musical instrument device. The operation of variable gain components is only linear within a certain range of voltages [12],

so driving the filter with higher-level signals often elicits a nonlinear behavior that is appreciated by musicians [13].

VCFs can be implemented in a circuit from discrete components or using dedicated integrated circuits (ICs). This paper examines a recent VCF design equipping the Korg Prologue synthesizer, released in 2018. This is a discrete-component low-pass filter, whose circuit diagram was published in a service manual made available by the manufacturer [14]. This paper is organized as follows. First, the filter design based on an analysis of the circuit is introduced, and a general description of it is provided, including its analog transfer function. The authors then propose a linear digital model of the filter obtained using both a bilinear transformation of the filter transfer function and the filter circuit flowchart. From this, existing nonlinearities in the signal path are also analyzed, and a second version that includes these is proposed. Then, the authors discuss the results and provide some conclusions.

1 THE PROLOGUE VCF

The Prologue is a polyphonic synthesizer released by Korg in 2018, whose circuitry includes eight or 16 identical voice cards containing three oscillators (two voltage-controlled filters and one digital multi-engine) that feed a VCF and finally, a voltage control amplifier (VCA). The low-pass VCF used in the Prologue synthesizer has two control parameters, cutoff frequency and resonance. These can be controlled manually from the front panel or automated by a variety of voltage sources. The synthesizer panel only indicates a numeric reference for both parameters, which then needs to be analyzed and interpreted to yield equivalent

VCF

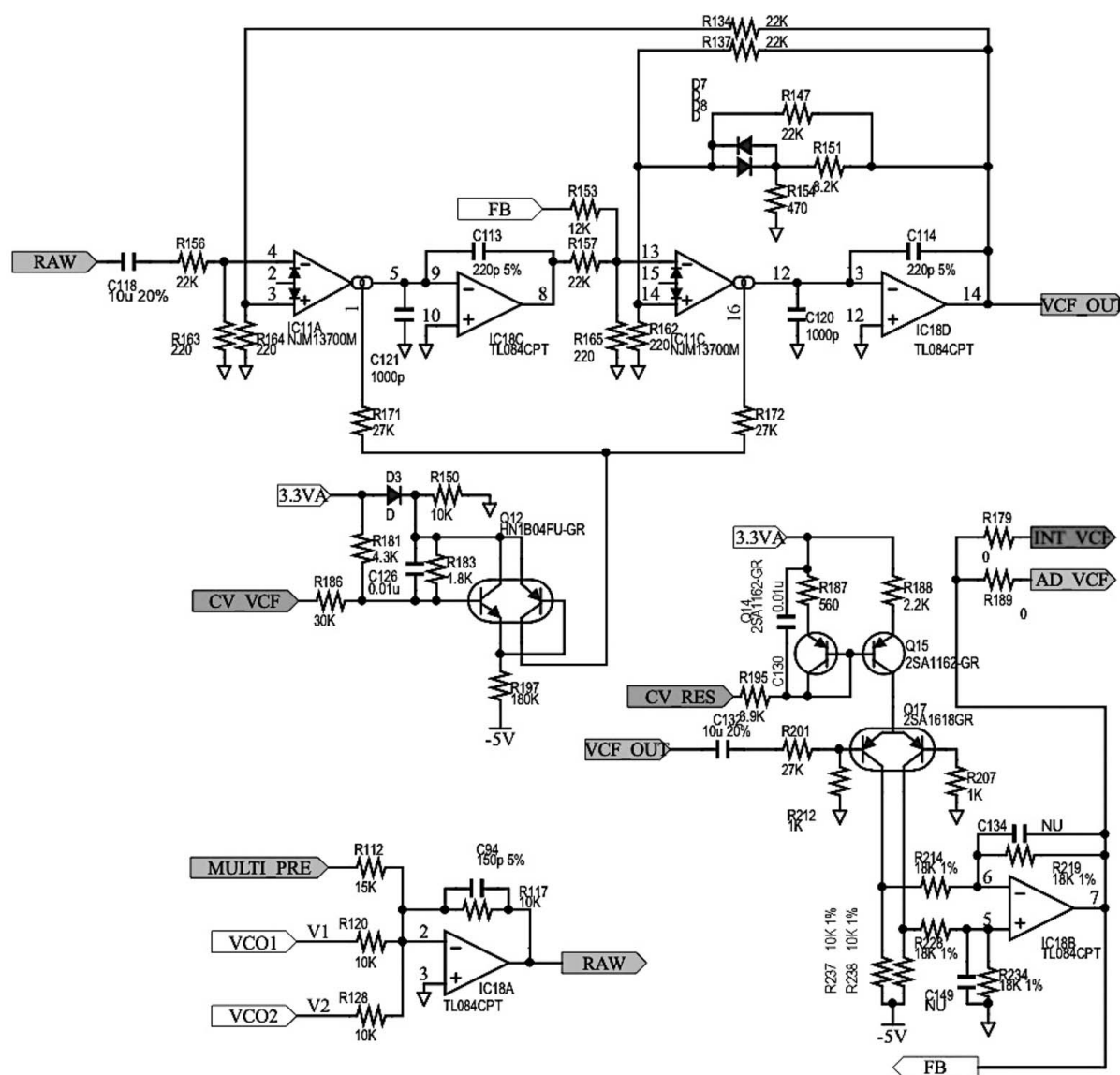


Fig. 1. VCF circuit diagram [14].

lent frequencies in Hz and damping (or Q) values. In this study, the authors will fix the cutoff frequency and vary the resonance from 0 to 100%, providing the equivalent values in Hz and damping/Q applied.

1.1 Filter Circuit

The VCF circuit is shown on Fig. 1, together with its cutoff and resonance control elements. It is based on two 13700 OTAs and two operational amplifiers (opamps). The former are used to apply variable gain as well as summing signals, and the latter are used as undamped integrators. Additional circuitry is included to make the filter voltage-controlled. In the schematic, the filter input is marked as RAW, and its output as VCF OUT. The RAW signal is an inverted mix of three sources (voltage controlled oscillators

1 and 2 and multi-engine). The OTA voltage control is derived from the CV VCF input, and the gain for the signal path marked FB is controlled by the CV RES input.

This circuit operates with two feedback paths: one from the output to the first OTA input and another from the output to the second OTA input. This second feedback route is composed of set-gain and CV-gain paths (marked FB in the schematic). These two are summed together in opposite polarities. In the absence of any signal in the CV feedback path, the signal flows through the set-gain path where it sees half the resistance if compared with the input signal. The presence of two back-to-back diodes in this route should be noted. These conduct if the signal exceeds approximately 0.7 V across this section of the circuit, following a voltage divider. In this case, the gain through this part of the network increases. The control voltage (CV)-gain feedback signal

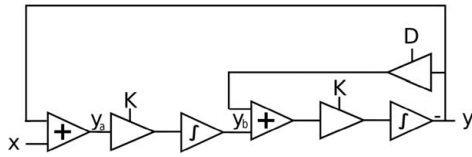


Fig. 2. VCF flowchart.

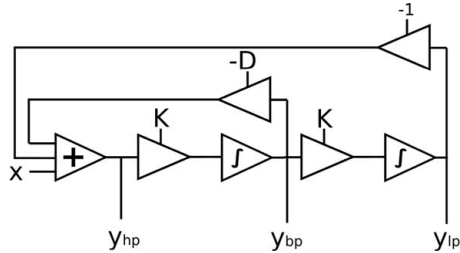


Fig. 3. State-variable filter flowchart.

path has its gain controlled by a differential amplifier. With an increase in CV and a corresponding gain increase in the variable signal path, the total feedback from the output to the second OTA input is reduced proportionally.

1.2 Filter Flowchart and Equations

Grouping all the variable gain elements, signal summing, and integrators, the flowchart given by Fig. 2 can be obtained. This filter can be described by the following tripartite equation,

$$\begin{aligned} y_a(t) &= y(t) + x(t) \\ y_b(t) &= K \int_0^t y_a(\tau) d\tau \\ y(t) &= -K \int_0^t [Dy(\tau) + y_b(\tau)] d\tau, \end{aligned} \quad (1)$$

with intermediate stages y_a and y_b as indicated in the flowchart. Using $1/s$ as the Laplace transform for the integrator and substituting in Eq. (1),

$$\begin{aligned} H_a(s) &= H(s) + 1 \\ H_b(s) &= \frac{K H_a(s)}{s} \\ H(s) &= -\frac{K [D H(s) + H_b(s)]}{s}, \end{aligned} \quad (2)$$

yielding the transfer function $H(s)$ for the VCF,

$$H(s) = \frac{-K^2}{s^2 + sKD + K^2}, \quad (3)$$

which is identical to the low-pass transfer function of an inverting state-variable filter (Fig. 3) [15, 16].

From this analysis, it can be seen that:

1. The filter configuration is that of a second-order (two-pole) voltage-controlled state-variable design.
2. Voltage control applied to the OTAs defines the cut-off frequency of the filter.
3. Voltage control applied to the gain of the feedback path defines the amount of peaking/resonance. Increasing the CV results in higher levels of peaking (or resonance), as the damping is minimized. Thus,

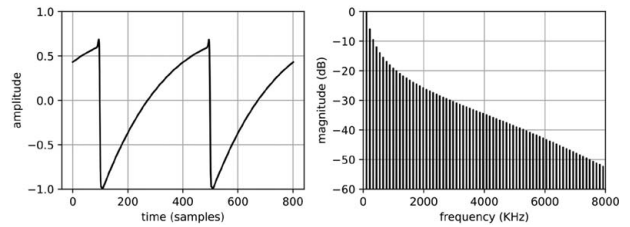


Fig. 4. Prologue digital oscillator sawtooth.

if the two paths combine to a zero sum, there is no damping applied to the circuit, and the poles would lie on the imaginary axis of the s -plane. This may lead the filter to become unstable. In practice, this does not take place because there is overload limiting built into the network.

The minimum amount of damping $D = 1/Q$ is set to 2, normally, with $Q = 0.5$. This places the filter poles on the real line in the s -plane. With $D = 0$, Q is infinite and poles lie on the imaginary line. The feedback VC control thus controls Q from a minimum of 0.5 to a very high value to make D approach 0. In the circuit, this is done in such a way that variable path signal gain goes from 0 to nearly 2, proportionally reducing D to nearly 0. Although many of the new synthesizers have generally tended to employ well-known topologies (such as the four-pole ladder filter derivatives), the Prologue filter represents an original two-pole not yet seen in the literature. As the authors will detail, it is, in fact, based on a variation of a known network, which nevertheless has been implemented in a novel way.

1.3 Output Analysis

Using a 110-Hz sawtooth wave as input, filter outputs for various resonance values are shown in Figs. 5–8. The sawtooth wave was sourced from the Prologue digital oscillator set to 50% level in the filter input mixer, and the cutoff frequency was fixed to 1,760 Hz. All waveforms were captured using the same output level settings, but given the various amounts of resonance, the peaks differed. The authors chose to plot all recorded signals normalized to a peak of 0 dBFS = 1. The Prologue allows the digital oscillator signal to bypass the VCF and be sent straight to the VCA. The pre-VCF digital oscillator output is shown in Fig. 4. This was later further reused in the analyses as a source signal for the digital models.

The zero-resonance behavior is fairly linear. As was expected, according to the circuit, the output is inverted in polarity, and the damping is set to 2, placing the two filter poles at real line in the s -plane. Because the filter is critically damped and $D \geq \sqrt{2}$, there is absolutely no peaking at the cutoff frequency (Fig. 5).

With 50% resonance, it can be seen that there is some amount of peaking, which has been found to be equivalent to $D \approx 0.16$ ($Q \approx 6$). There is also a small amount of nonlinearity, which can be seen as a shoulder in the filter amplitude response at about -40 dB (Fig. 6).

If the resonance is increased, the effect of nonlinearities in the filter output can be seen more clearly. With

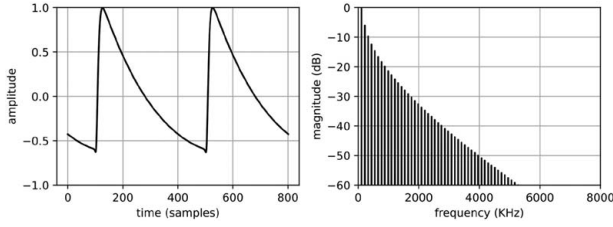


Fig. 5. Prologue VCF output, with zero resonance.

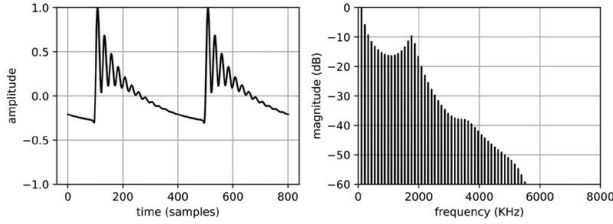


Fig. 6. Prologue VCF output, with 50% resonance.

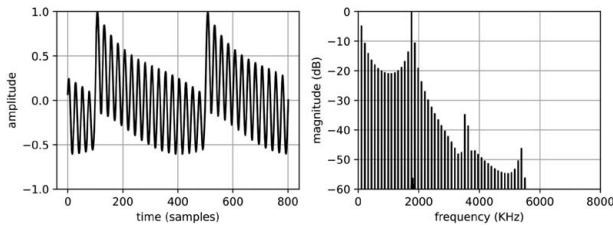


Fig. 7. Prologue VCF output, with 70% resonance.

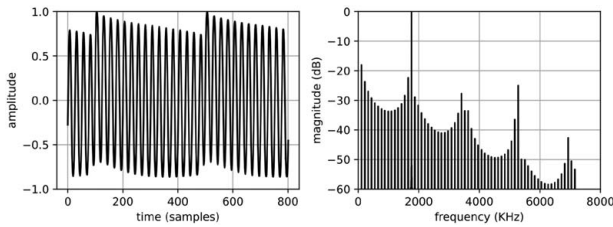


Fig. 8. Prologue VCF output, with 100% resonance.

70% resonance (Fig. 7), the filter amplitude response shows bumps and subsidiary peaks at frequencies beyond the cut-off. It is interesting to note as well how different the waveform is from a linear low-pass state-variable filter output of an equivalent Q (see Fig. 12 for comparison). The resonance follows the shape of the input sawtooth quite well, whereas in the linear filter, there is a widening followed by a squeezing out of the resonance toward the end of the waveform ramp. A similar effect has been observed in nonlinear four-pole filters (of the ladder-derivative design).

If the resonance is increased to the maximum and the filter is beyond its self-resonance point, distinct subsidiary peaks in the filter amplitude response are seen. These indicate that the filter output, although deemed to be close to a sine wave in the absence of input, will exhibit a few extra harmonics due to the effects of nonlinear waveshaping (Fig. 8).

Listing 1. The Korg Prologue VCF implemented as a Csound UDO.

```
opcode PrologueVCF, a, akk
setksmps 1
as1, as2 init 0, 0
as, kK, kD xin
kA = 1 / (1 + kK * kD + kK * kK)
aya = (as2 + as) * kA * kK
ayb = as1 + aya
as1 = ayb + aya
au = -kK * (ayb + kA * (kD + kK) * as2)
alp = as2 + au
as2 = alp + au
xout alp
endop
```

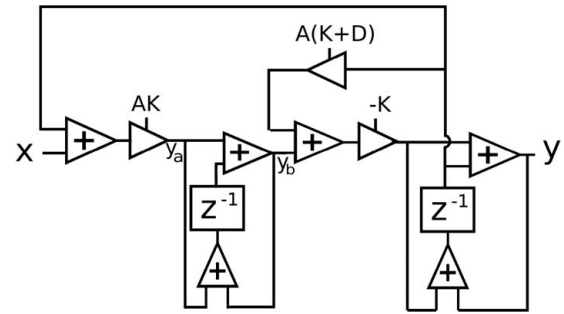


Fig. 9. VCF model flowchart.

2 LINEAR IMPLEMENTATION

A linear implementation of the Prologue VCF as a digital filter has been developed. This is based on the application of the bilinear transformation [17, 18] to the analog transfer function (Eq. (3)), employing the approach described in detail by [15],

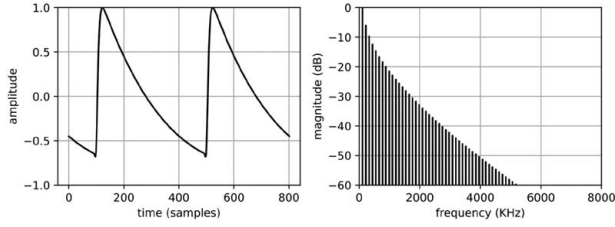
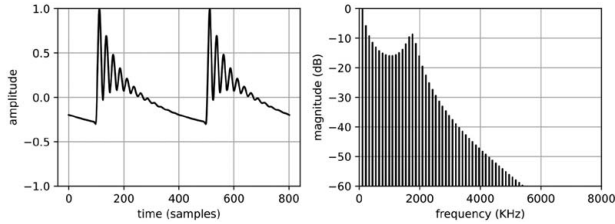
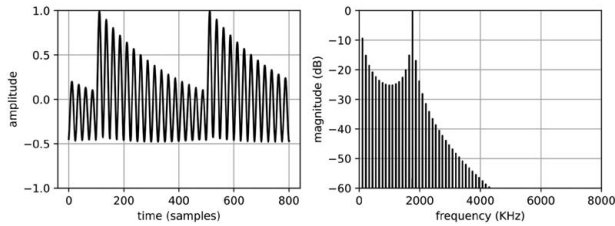
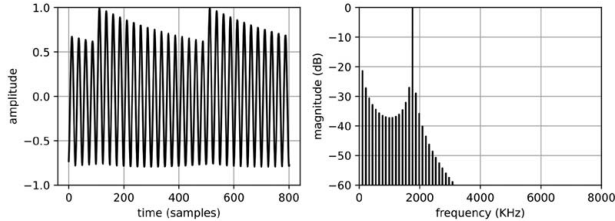
$$H(z) = -\frac{K^2}{1 + KD + K^2} \left[\frac{(1 + z^{-1})^2}{1 - \frac{2(1-K^2)z^{-1} - (1-KD+K^2)z^{-2}}{1+KD+K^2}} \right]. \quad (4)$$

The digital implementation largely follows the analog filter form, implementing the undamped integrators of Fig. 2 as bilinear integration filters and deriving the feedback signals directly from the state variable of each integrator as described in [15]. This yields the following set of digital filter equations,

$$\begin{aligned} y_a(n) &= AK[s_2(n) + x(n)] \\ y_b(n) &= s_1(n) + y_a(n) \\ s_1(n) &= y_b(n) + y_a(n) \\ u(n) &= -K[y_b(n) + A[D + K]s_2(n)] \\ y(n) &= s_2(n) + u(n) \\ s_2(n) &= y(n) + u(n), \end{aligned} \quad (5)$$

with $A = (1 + KD + K^2)^{-1}$. In this equation, s_1 and s_2 represent the states of each one of the two integrators. The Csound user-defined opcode (UDO) implementation of this filter is shown in Listing 1 and its flowchart on Fig. 9.

Setting $K = \tan\left(\frac{\pi f}{f_s}\right)$ [15] allows this filter to be tuned to a given peak frequency. The linear implementation approximates the Prologue VCF fairly well for high values of

Fig. 10. Linear model output, $D = 2$.Fig. 11. Linear model output, $D = 0.16$.Fig. 12. Linear model output, $D = 0.02$.Fig. 13. Linear model output, $D = 0.005$.

D , starting with the critically damped state (resonance set to zero, $D = 2$). This is shown in Fig. 10, where the authors plotted the output of the linear model fed with the sawtooth wave of Fig. 4.

However, for values below $D = \sqrt{2}$, where peaking starts to be observed, the nonlinear behavior of the Prologue VCF starts to deviate significantly from the linear model. This can be seen for three different values of D , which approximate the Prologue resonance settings of 50% ($D = 0.16$, Fig. 11), 70% ($D = 0.02$, Fig. 12), and 100% ($D = 0.005$, Fig. 13).

3 NONLINEAR IMPLEMENTATION

Recognizing the nonlinear behavior of the OTAs and transistors used in this filter allows the authors to put forward an implementation that tries to model it more accurately. Although there may be issues with the underlying bilinear model when it comes to simulations [19], the authors

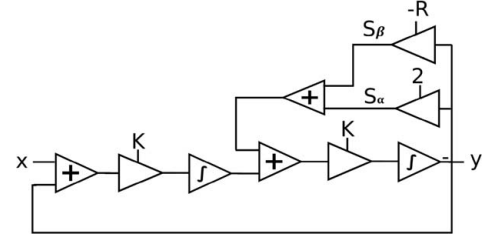


Fig. 14. Updated VCF flowchart.

felt that it was worth attempting. Following the approach taken in [20], this was tackled in a twofold manner:

1. The authors replaced the sections of the code where the K scaling is applied by a hyperbolic tangent waveshaper. This should account for any nonlinearities introduced by the OTAs.
2. In the linear model, all damping feedback was lumped into one single path. Now, attention should be paid to the fact that there are separate fixed and variable gain signal routings. Because the latter is controlled by a differential transistor pair, a nonlinearity should also be included in this path.

3.1 Saturation

In order to have a control over the amount of saturation applied to the signal, a parameter g can be included. This can be used in conjunction with an input level control, and with these, the filter behavior can be fine tuned. In the case of the OTAs, this yields the replacement [20],

$$Kx \rightarrow \frac{K}{g} \tanh(gx). \quad (6)$$

3.2 Feedback

The feedback path in the VCF circuit is slightly more complicated than in the derived flowchart (Fig. 2); thus, the original linear equations do not anticipate two signals being used to control the amount of damping. The approach now is to follow the filter circuit characteristic where the gain of the variable path is proportional to the amount of resonance. With the resonance $R = 0$, then $D = 2$, and as R grows, the amount of signal coming from the fixed path is subtracted by that from the variable path, with D approaching zero as R gets to its maximum,

$$s_v = 2s_\alpha - Rs_\beta, \quad (7)$$

where s_v is the total feedback mix, s_α is the fixed feedback signal, and Rs_β the variable feedback path. This modification is illustrated in Fig. 14. Now, a waveshaper can be inserted here to model the nonlinearity in the circuit,

$$Rs_\beta \rightarrow \frac{R}{g} \tanh(g s_\beta). \quad (8)$$

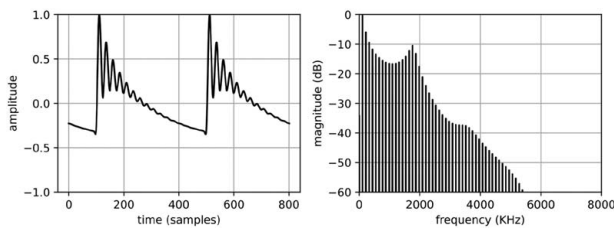
The Csound code implementing this nonlinear filter is shown in Listing 2. Note that its parameters differ slightly from the linear version, as the UDO takes in a resonance control [0, 2] and an extra parameter defining g , whose value actually depends on the input signal levels. For full-scale

Listing 2. Nonlinear implementation.

```

opcode PrologueVCFNL,a,akki
setksmps 1
as1,as2 init 0,0
as,kK,kRes,ig xin
igi = 1/ig
kA = 1/(1+kK*(2-kRes)+kK*kK)
aya = (as2 + as)*kA
au = tanh(aya*igi)*kK*igi
ayb = as1 + au
as1 = ayb + au
afdb = 2*as2 - kRes*tanh(as2*igi)*igi
au = -tanh((ayb +
    kK + 1)*afdb*kA)*igi)*kK*igi
alp = as2 + au
as2 = alp + au
xout alp
endop

```

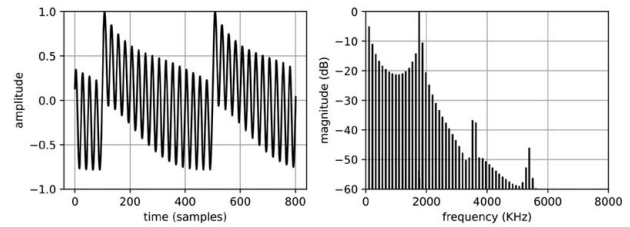
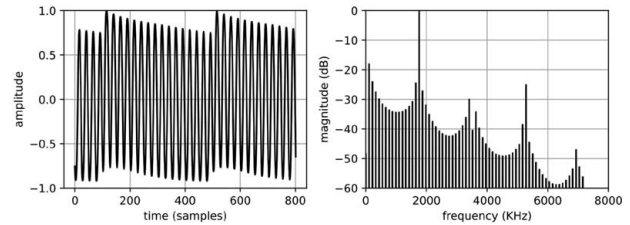
Fig. 15. Nonlinear model output, $R = 1.77$.

input, the authors have found that setting g in the range $[0.1, 0.2]$ approximates the Prologue VCF output fairly well.

In addition to these nonlinear elements in the filter, it is also necessary to account for the nonlinear behavior of the VCA that follows the filter before the output. This has a small but noticeable effect on the spectrum and is important when trying to approximate the filter output spectrum. The VCA also consists of a differential transistor pair, and to include its effects, a $\tanh()$ waveshaper can be placed at the instrument output.

With this implementation, the authors were able to approximate the nonlinear behavior of the Prologue VCF, producing an output that matches both the original waveform and spectrum quite closely. Fig. 15 shows that the 50%-resonance VCF output can be reproduced faithfully, using $R = 1.77$, and $g = 0.2$, applying the nonlinear model to the full-scale sawtooth taken from the Prologue digital oscillator. Comparing with Fig. 6, it can be seen that both the peak and the characteristic shoulder at around -40 dB are well characterized in this filter. Also, when the waveform of the equivalent linear model (Fig. 11) is examined, it can be seen that there are very slight deviations that make the output of this filter a more faithful emulation of the original.

At higher values of resonance, although the original output is fairly well matched, there are a few minor differences in the spectrum and waveform. In Fig. 16, where $R = 1.92$ ($g = 0.2$) is used, the 70%-resonance VCF output is approximated. The differences appear in the shape of the higher part of the spectrum, and although the waveform also exhibits a matching overall shape with the resonance fol-

Fig. 16. Nonlinear model output, $R = 1.92$.Fig. 17. Nonlinear model output, $R = 1.95$.

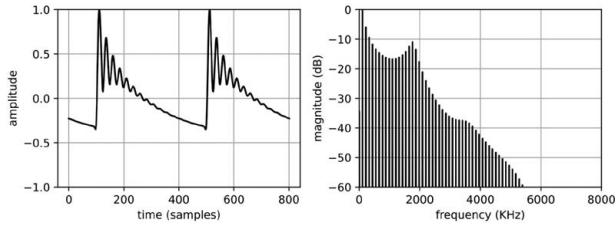
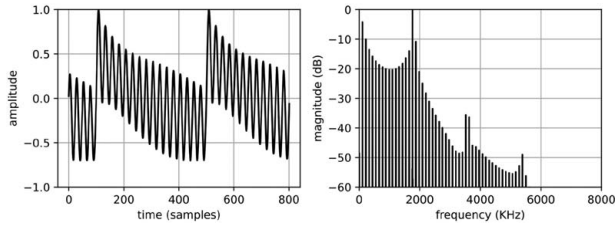
lowing the outline of the sawtooth wave, it is nevertheless slightly different.

Finally, in Fig. 17, the authors attempted to match the 100% resonance signal, with $R = 1.92$. For this, it was necessary to change g to 0.1, because the nonlinearities have the side effect of limiting the minimum damping that is achievable. The results both in terms of waveform and spectrum are again very close to the original. The general shape of the amplitude response was matched fairly well, with small discrepancies in some individual harmonic amplitudes. Because of the aural dominance of the first peak, on listening, these discrepancies do not come through, and the two filter outputs sound identical.

4 FEEDBACK LIMITING

In this first nonlinear model, it is assumed that the VCF has a fixed gain damping feedback path, in addition to the voltage-controlled route, but that is not quite the case. The authors are conscious of the fact that they have not accounted for the aforementioned diodes in the circuit. It was not immediately clear to the authors whether these introduce any significant nonlinearity to the output at normal levels. Because the authors are not driving the filter with high input levels (using a single source at 50% level) in this study, the expectation was that these diodes do not conduct except in exceptional circumstances. These assumptions can now be tested by introducing a fourth nonlinearity in the model.

The first impression is that the diodes only come into play with high gain in the feedback path, which are partly dependent on the VCF input levels. They form a protective measure to avoid the damping to become too small and make the filter behavior more difficult to control. The diode path does this by increasing the signal level (and thus damping) when peaking exceeds a certain amount, working as an expander but having the effect of a high-Q limiting device. This is seen in other filter designs, such as for instance the state-variable low-pass overload limiter

Fig. 18. Revised nonlinear model output, $R = 1.77$.Fig. 19. Revised nonlinear model output, $R = 1.93$.

example shown in the CEM3350 datasheet [21]. A similar approach is implemented in the MS20 filter, where a back-to-back diode circuit is employed [11]. This squashes the resonance peaks when the filter input signal level exceeds a certain threshold.

In the Prologue VCF case, the nonlinearity is modeled using a piecewise function. Examining the circuit, it can be observed that the gain increases by a little over a factor of two as the diodes start conducting. Although this change is smooth, this can nevertheless be approximated with a transfer function consistent of two linear segments,

$$f(x) = \begin{cases} 2x, & |x| < 0.7 \\ 4.6x, & |x| \geq 0.7 \end{cases} \quad (9)$$

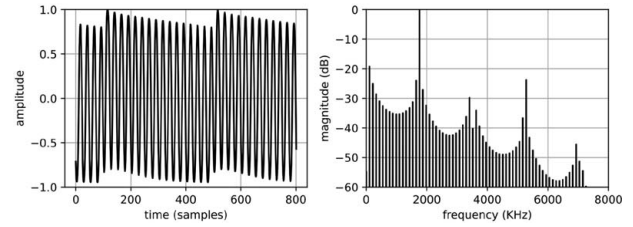
with the nominal threshold set to 0.7. This waveshaper can be inserted in place of s_α , with the appropriate scaling g_α ,

$$s_\alpha \rightarrow \frac{f(g_\alpha s_\alpha)}{g_\alpha}. \quad (10)$$

As expected, in the case of a low amount of resonance (50%), this change does not affect the output of the digital model in a significant way. It can be seen that a similar spectrum and waveform is obtained, as in the case of a fixed gain (Fig. 18).

However, as the resonance is increased to the equivalent of 70% in the original filter, an improvement can be observed in the approximation. For this, R has been increased to 1.93, and as can be seen, the waveform is a very close match to the original VCF output, as is the spectrum (Fig. 19). Although the changes are small, they are not insignificant; a comparison can be drawn with Fig. 16. It can be seen that in the revised model output, the shoulder in the region around 4 kHz is a closer match to the one in Fig. 7.

To try to best approximate the 100%-resonance output with this revised model, it is necessary to set $R = 1.973$. This shows the role of the diodes in increasing the damping, because the output of this filter with the same value of R as the first nonlinear model has a reduced amount of

Fig. 20. Revised nonlinear model output, $R = 1.973$.

resonance. As shown in Fig. 20, the approximation is also improved in this case.

Although these changes may, in practical applications, be seen as small, they nevertheless show that the effect of the diodes plays a part in the Prologue VCF and a digital model would not be complete without it. With higher input levels, their overloading effect should be expected to be more prominent. This approximation of the transfer function may also be improved by a smoothing of the sections where the two linear segments are joined together. This may also contribute to an even closer approximation of the filter output.

5 DISCUSSION

The authors began this study motivated by a desire to elucidate how the VCF operated. At first look, because of the presence of two OTAs and integrators in the circuit, it was suspected the filter was a voltage-controlled state-variable design. On closer inspection, it was noted that the damping-control feedback path was not in its usual place as per the flowchart in Fig. 3, which made the authors ponder. They then proceeded to prove that this modification still yielded a state-variable filter, albeit with an inverted output. It is possible to insert the signal at different points in the circuit to obtain different outputs, and this circuit also shows that the feedback paths can be modified without essentially changing the filter. It is possible to extract high-pass and band-pass outputs out of the Prologue VCF and combine them to obtain notch and all-pass signals as well. Thus, it is perhaps a missed opportunity not to have these available in the synthesizer.

After the design of the filter was determined, the next step was to construct a linear digital model with two goals: the first was to validate the analysis, and the second was to have a means of studying the filter in its pure form, without any interference from the limitations of analog components. The authors not only showed that the filter was of the kind given by the analysis of the original filter circuit, but more importantly, also demonstrated that the VCF was fairly nonlinear by comparison. Although a nonresonant filter matched the linear equations very well, the most common musical use case, where some amount of peaking is applied, deviates significantly from the linear implementation.

This study then turned toward characterization of the nonlinearities in the filter circuit. The authors traced the usual suspects in the design and applied their characteristic waveshaping to the filter code. However, there are a number

of difficulties in this process. First, as the authors tried to approximate the filter output, they only had the recordings of the VCF and that of the source bypassing the filter. These have gone through an extra amount of circuitry, including an analog-to-digital conversion (ADC) and back through a digital-to-analog conversion (DAC) inside the synthesizer (the Prologue does not have a full analog path from filter to output). The authors started by assuming that none of this added any further nonlinearity, but it soon found out that this was not the case. From the study of the digital nonlinear model, it was observed experimentally that there was the trace of some nonlinearity in the spectrum that could not have been produced by the filter. Ideally, the authors would have wanted to sample the voltages coming into the filter input and arising at the output. However, because it was not possible to do this, the authors had to proceed as best as possible with the available resources.

Studying the complete circuit from filter to output (and ignoring the ADC/DAC), it was found that the VCA circuit could potentially be overloaded when higher signal levels are pushed through. This situation could arise particularly in the case of filter peaking and high resonance. When the VCA nonlinearities were applied to the filter output, it was found that it was possible to then approximate the original VCF recordings much more closely. It is expected that the bypassed oscillator signal on its own, which also goes through the VCA, was not of sufficient level to bring any significant nonlinear behavior into play. If this were the case, it would possibly have been impracticable to obtain the results reported here, because the authors relied on a clean linear input for the analyses.

One final question that requires some discussion is that of the variation in the amount of drive required to approximate the signal at full resonance. This is a problem that can possibly only be fully solved by a measuring the voltages in the circuit. First, the exact voltage levels coming into the filter are not known, because the circuit diagram does not give a complete picture of the synthesizer operation. Like many modern instruments, the Prologue is partly digital, and most parameters are driven from the central processing unit (CPU) via pulse width modulation (PWM) signals. So the details of control voltages defining mixer output levels are effectively software defined. Without access to these, the authors do not really know much more than the supply rail voltages present shown in the circuit design. Adding to this, the source signal being used is digitally produced, and for the same reason, the authors do not know its actual voltage output level, given a mixer setting of, say, 50%. So it is necessary to rely on an empirical approximation of the filter outputs.

In the digital model, a normalized input (peaking at 0 dBFSdbfs = 1) was used, and then it was determined experimentally how much the input needs to be reduced as it enters the waveshaper. This is done to try to recreate the voltage levels relative to transistor thermal voltage that are taking place in the filter, thus approximating the signals obtained from the original VCF.

The values of the voltage dividers at the input of each nonlinear component were noted, but these do not tell much

beyond the fact that it is necessary to take care of levels at those points. They could be considered relative to each other; at the OTAs, the reduction ratio is about 1/100, and at the transistor pair in the feedback path, it is about 1/28. Because the same nonlinear function is used for both the OTAs and the differential amplifier, it makes sense to ignore these differences and apply a single gain reduction value at each point, therefore effectively replacing the transistor pair by an OTA in this nonlinear model. These differences may have an impact on the exact modeling of the filter, which required the reduction in the single drive gain employed in the case of 100% resonance.

Finally, it should be noted that the issue of oversampling has not yet been discussed, and this may have important implementation considerations. Because nonlinear waveshapers were employed in the proposed digital model, signals circulating through the filter cannot be guaranteed to be bandlimited. This brings forth the possibility of audible aliasing distortion and additionally, implications for filter behavior, particularly at high cutoff frequencies. It is a question that cannot be ignored, although it is somewhat beyond the scope of this paper. A detailed assessment of this issue should inevitably be part of any practical implementations of this filter model.

6 CONCLUSIONS

This paper set out to study the Prologue VCF and provide both a mathematical and a computer code model for it. Using its circuit diagram, a flowchart describing the filter was developed, and then an analog transfer function describing it was derived. Recordings of the synthesizer output using the VCF at various levels of resonance were then analyzed. From the analog transfer function and the filter flowchart, a linear digital model was derived, which validated the earlier analysis. Comparisons with the original filter recording showed that significant nonlinearities were present, and the authors set off to try and model these in order to approximate better the VCF output.

It was demonstrated that if waveshapers are introduced at the points in the circuit where variable gain components are present, the original filter output waveforms and spectra can be approximated fairly well. In the discussion, the authors have attempted to explore the motivations and pitfalls of the process. The final conclusions reinforce the idea that nonlinearities are a significant element of voltage-controlled analog filters, which are generally appreciated by their users in various musical settings. Code and examples are available at github.com/vlazzarini/prologuevcf.

7 REFERENCES

- [1] R. Moog, "Voltage-Controlled Electronic Music Modules," presented at the *16th Convention of the Audio Engineering Society* (1964 Oct.), paper 346.
- [2] T. Stilson and J. Smith, "Analyzing the Moog VCF with Considerations for Digital Implementation," in *Proceedings of the International Computer Music Conference*, pp. 398–401 (Hong Kong) (1996 Aug.).

- [3] F. Fontana, "Preserving the Structure of the Moog VCF in the Digital Domain," in *Proceedings of the International Computer Music Conference*, pp. 291–294 (Copenhagen, Denmark) (2007 Aug.).
- [4] S. D'Angelo and V. Välimäki, "Generalized Moog Ladder Filter: Part I: Linear Analysis and Parameterization," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 22, no. 12, pp. 1825–1832 (2014 Dec.). <https://doi.org/10.1109/TASLP.2014.2352495>.
- [5] S. D'Angelo and V. Välimäki, "Generalized Moog Ladder Filter: Part II: Explicit Nonlinear Model Through a Novel Delay-Free Loop Implementation Method," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 22, no. 12, pp. 1873–1883 (2014 Dec.). <https://doi.org/10.1109/TASLP.2014.2352556>.
- [6] A. Huovilainen, "Non-Linear Digital Implementation of the Moog Ladder Filter," in *Proceedings of the Seventh International Conference on Digital Audio Effects (DAFx)*, pp. 61–64 (2004 Oct.).
- [7] B. Hutchins, "Additional Ideas for Voltage Controlled Filters," *Electronotes*, vol. 10, no. 85, pp. 5–17 (1978 Jan.).
- [8] F. Fontana and M. Civolani, "Modeling of the EMS VCS3 Voltage-Controlled Filter as a Nonlinear Filter Network," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 18, no. 4, pp. 760–772 (2010 May). <https://doi.org/10.1109/TASL.2010.2046287>.
- [9] B. Hutchins, "Revisiting Some VCF Ideas and a Few New Ideas," *Electronotes*, vol. 23, no. 215, pp. 1–23 (2013 May).
- [10] W. Pirkle, "Modeling the Korg35 Lowpass and Highpass Filters," presented at the *135th Convention of the Audio Engineering Society* (2013 Oct.), paper 103.
- [11] T. Stinchcombe, "A Study of the Korg MS10 and MS20 Filters," https://www.timstinchcombe.co.uk/synth/MS20_study.pdf (2006).
- [12] B. Hutchins, "Analog Signal Processing, Chapter 8," *Electronotes*, vol. 20, no. 196, pp. 2–10 (2000 Dec.).
- [13] D. Rossum, "Making Digital Filters Sound Analog," in *Proceedings of the International Computer Music Conference*, pp. 30–32 (San Jose, CA) (1992 Oct.).
- [14] Korg, *Polyphonic Analogue Synthesizer Prologue-8/Prologue-16 Service Manual* (Korg Inc., Tokyo, Japan, 2018).
- [15] V. Lazzarini and J. Timoney, "Improving the Chamberlin Digital State Variable Filter," *J. Audio Eng. Soc.*, vol. 70, no. 6, pp. 446–456 (2022 Jun.). <https://doi.org/10.17743/jaes.2022.0001>.
- [16] B. Hutchins, "Analog Signal Processing, Chapter 6," *Electronotes*, vol. 19, no. 194, pp. 4–30 (2000 Apr.).
- [17] K. Steiglitz, "The Equivalence of Digital and Analog Signal Processing," *Inf. Control*, vol. 8, no. 5, pp. 455–467 (1965 Oct.).
- [18] K. Steiglitz, *A Digital Signal Primer* (Addison-Wesley, Menlo Park, CA, 1996).
- [19] T. Hélie, "Lyapunov Stability Analysis of the Moog Ladder Filter and Dissipativity Aspects in Numerical Solutions," in *Proceedings of the 14th International Conference on Digital Audio Effects (DAFx)*, pp. 1–8 (Paris, France) (2011 Sep.).
- [20] V. Lazzarini and J. Timoney, "Linear and Nonlinear Digital Filters: From the Analog and Beyond," *Comput. Music J.*, vol. 45, no. 2, pp. 67–83 (2021).
- [21] Curtis Electromusic Specialities, "CEM 3350 Dual Controlled State Variable Filter," <http://www.synfo.nl/datasheets/CEM3350.pdf> (1984).

THE AUTHOR



Victor Lazzarini



Joseph Timoney

Victor Lazzarini is a graduate of the Universidade Estadual de Campinas (UNICAMP) in Brazil. He completed his doctorate at the University of Nottingham, UK. His research is focused on musical signal processing and computer music languages. Recent papers include "Parallel Computation of Time-Varying Convolution" (*Journal of New Music Research*, 2020), "Improving the Chamberlin Digital State Variable Filter" (*Journal of the Audio Engineering Society*, 2022), and "Linear and Nonlinear Digital Filters: From the Analog and Beyond" (*Computer Music Journal*, 2022). Prof. Lazzarini has also recently completed a research monograph for Oxford University Press, *Spectral Music Design: A Computational Approach* (2021).

•

Joe Timoney studied Electronic Engineering, completing his PhD in 1998. He joined the Department of Computer Science at NUI Maynooth in the following year. He teaches undergraduate programs in Computer Science and Music Technology. His research interests are based in the area of audio signal processing, with a focus on musical sound synthesis and the digital modeling of analog subtractive synthesis. He has supervised a number of PhD students in the fields of audio analysis and digital audio watermarking. Additionally, he has worked on EI innovation vouchers and a Commercialisation project on Watermarking.