Research Report

Development of a Novel Filter by Cascade Connection of Two State Variable Filters

Ichiro TSUJI

Abstract

This paper describes the signals of a rich sound effect obtained by adding the output signal of a circuit formed by cascade connection of a digital circuit and an all-pass filter, and the output signal of a state variable filter. This state variable filter was formed by cascade connection of two component state variable filters. The outputs of a state variable filter are obtained from three component filters: a low-pass filter, a high-pass filter, and a bandpass filter (Imada & Fukaya 1989). In this case, the output signal of the low-pass filter of the first of the two state variable filters was input into the second state variable filter, the output signal of the high-pass filter of the second state variable filter was input into the first, and the output signal of the bandpass filter of the first state variable filter was set as the output. Furthermore, an analog photocoupler was used as the resistance that controls the center frequency and sensitivity of the first state variable filter, allowing free control of the center frequency and sensitivity (Tsuji 2016). The control signal input into an analog photocoupler was used for calculations in an analog circuit, along with the signal produced by dividing the frequency of the pulse width modulation (PWM) signal of the input signal with a binary counter (Tsuji 2018). This permitted intricate control of the sensitivity and center frequency, with desirable consequences for the resulting sound effects. The second state variable filter included variable resistors, permitting control of the cutoff frequency. This allowed control of the timbre of the resulting sound effects. The digital signals created from the PWM signal by the binary counter were changed by a multiplexer, creating a digital signal output, which was then input into an all-pass filter. This all-pass filter used an analog photocoupler to provide resistance, enabling it to be adjusted to control the phase of the filter system (Tsuji 2016). Control of the phase could also be carried out by control of the input digital signals. The output signal of the state variable filter and the all-pass filter were added, thus modulating the output signal of the state variable filter. This filter design could then be used to create sound effects by the signal analysis of the output signal of the filter. This design was tested using a computer synthesizer, a USB device, a line selector, and the filter already discussed, permitting evaluation of the filter's performance.

1. Introduction

Doepfer Musikelektronik GmbH has developed the A-100 Analog Modular System (Doepfer Musikelektronik GmbH), featuring over 125 types of modules with appropriate module selection, complex sound signals can be synthesized. However, in order to change the timbre of a signal, some manual adjustments are required. When using many modules, it may require multiple operators to adjust numerous parameters quickly. The basic structure of the synthesizer system developed by the author is shown in Figure 1. Although this system is simple compared with the A-100, many parameters can be changed instantly using computer control (Tsuji 2018). This permits rapid, extensive changes in timbre. Using a computer synthesizer, along with five PWM signals that input into five analog filters, this system can carry out control of the cutoff frequency, center frequency, or sensitivity of an analog filter, permitting automatic composition and speaker control. As shown in Figure 1, five ports, numbered 0 through 4, represent the output terminal of a USB device. Five PWMs, numbered 1 through 5, represent the PWM signal output from this device. Filters 1 through 5 represent analog filters that input PWM signals. Additionally, this version of the system differs from the previous version (Tsuji 2018) as follows. A PWM signal (PWM 5) is input from the USB device to the line selector. Another PWM signal (PWM 4) is input to Filter 4 from the USB device directly. Filter 4 is the cascade connection of a biquad circuit and an all-pass filter. Filter 5 uses a special structure developed to allow a richer timbre for the overall system. The basic structure of the developed filter is shown in Figure 2. Initially, the input PWM signal is divided at the binary counter. This signal is then added to the input signal and input into the state variable filter. This creates an octaver-like effect. The output signal of the state variable filter is more deeply modulated by combination of the output signals of an all-pass filter and a multiplexer. Typically, the control signal of a state variable filter is an analog signal, with intricate effects on the timbre of the output signal. However, digital control of a state variable filter is realizable (Kaneko et al. 2015). This adds some complications. In cases where a digital state variable filter uses a DSP board, or special programming, it is necessary to take into consideration processing time, the

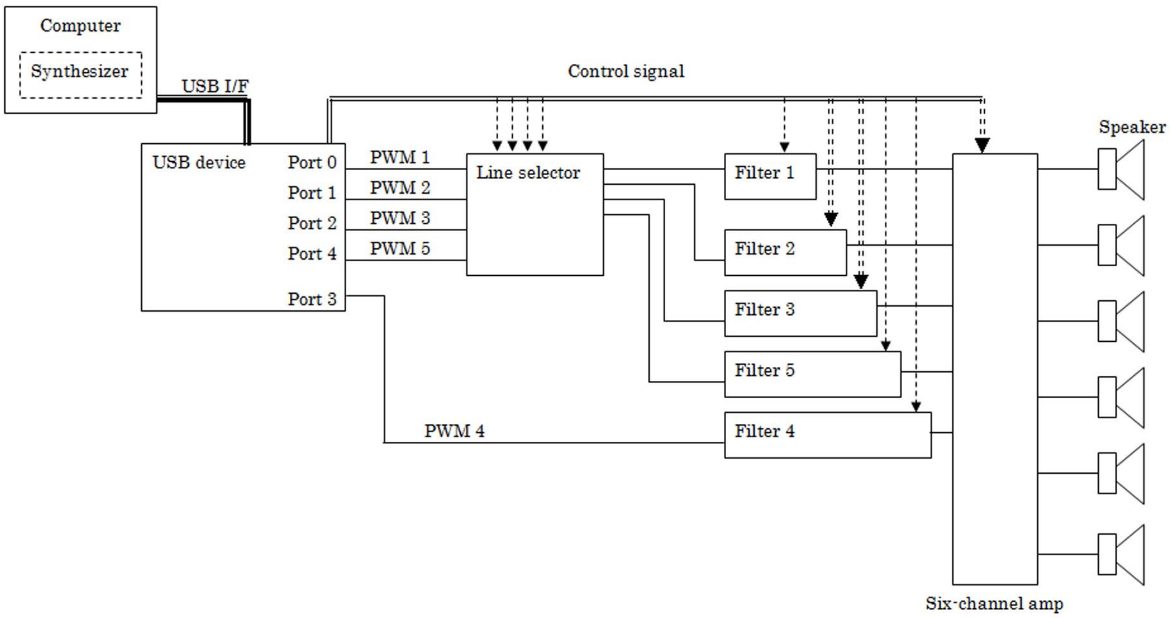


Figure 1. Basic structure of a synthesizer system

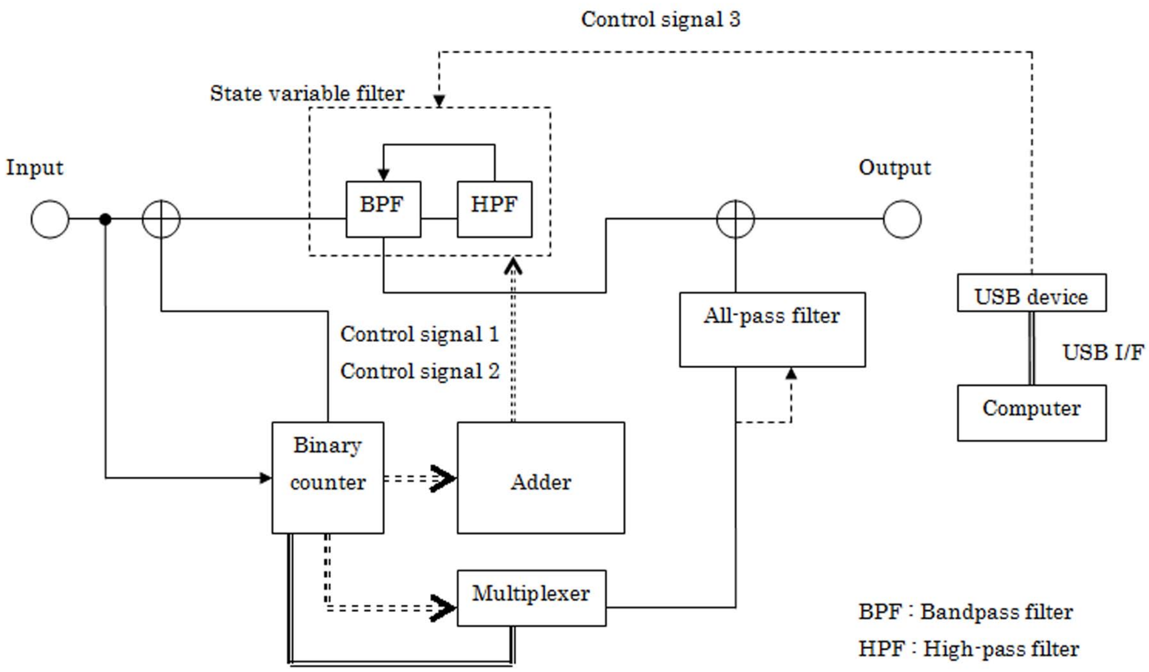


Figure 2. Basic structure of the developed filter

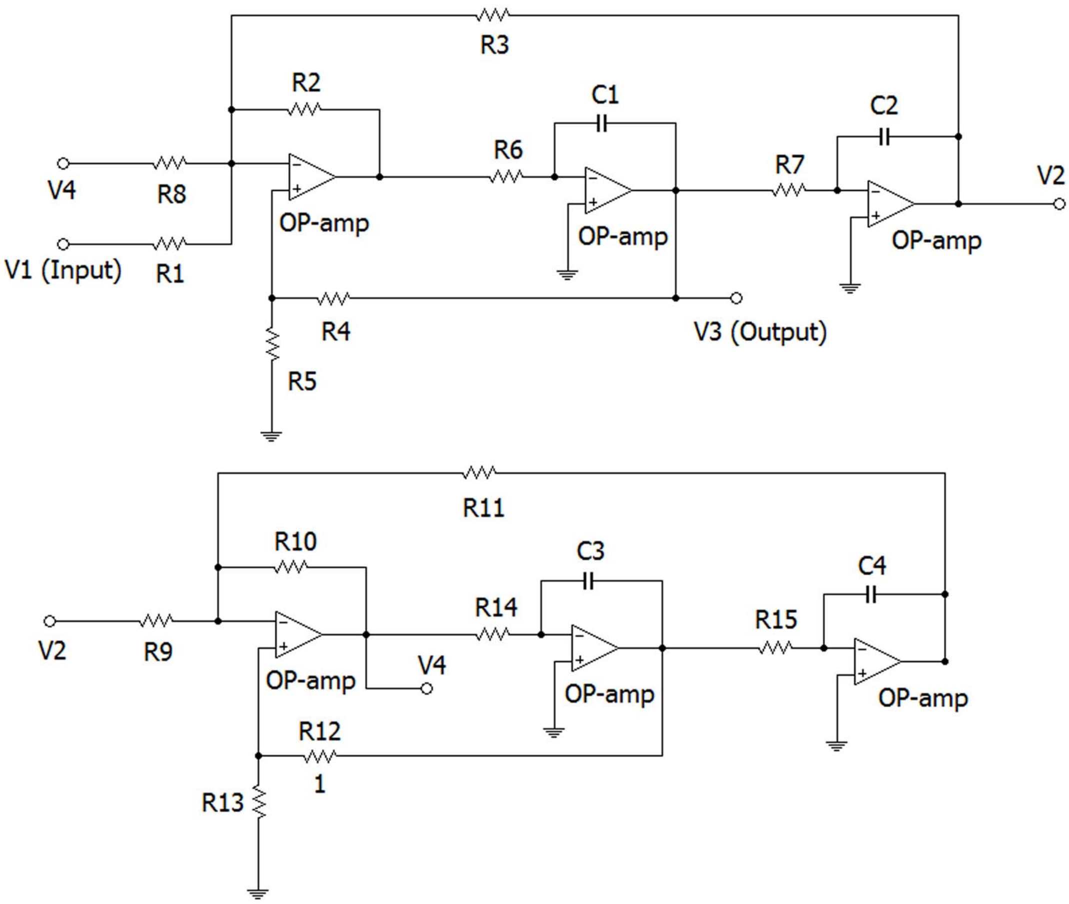


Figure 3. Conventional state variable filter

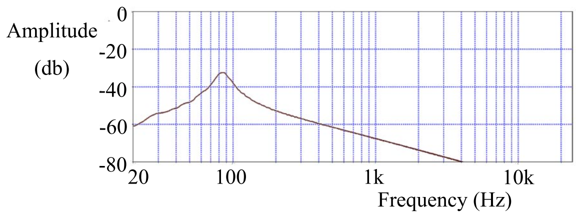


Figure 4. Frequency characteristic of the developed filter

$$
\begin{aligned}
T_{BP} &= \frac{\frac{\omega_1}{Q_1}s}{s^2+\frac{\omega_1}{Q_1}s+\omega_1^2} \\
T_{HP} &= \frac{s^2}{s^2+\frac{\omega_2}{Q_2}s+\omega_2^2} \\
T_1 &= \frac{V_3}{V_1} = \frac{\frac{R2}{C1R1R6}s}{s^2+\frac{R5}{C1R1R3R6R8(R4+R5)}(R1R2R3+R2R3R8+R1R3R8+R1R2R8)s+\frac{R2}{C1C2R6R7}(\frac{T_2}{R8}+\frac{1}{R3})} \\
\omega_1 &= \sqrt{\frac{R2}{C1C2R6R7}(\frac{1}{R3}+\frac{1}{R8})} \\
Q_1 &= \frac{C1R1R3R6R8(R4+R5)}{R5(R1R2R3+R2R3R8+R1R2R8+R1R3R8)}\sqrt{\frac{R2}{C1C2R6R7}(\frac{1}{R3}+\frac{1}{R8})} \\
BW_1 &= \frac{\omega_1}{Q_1} \\
T_2 &= \frac{V_4}{V_2} = \frac{\frac{-R10}{R9}s^2}{s^2+\frac{R13}{C3R9R11R14(R12+R13)}(R10R11+R9R11+R9R10)s+\frac{R10}{C3C4R11R14R15}} \\
\omega_2 &= \sqrt{\frac{R10}{C3C4R11R14R15}} \\
Q_2 &= \frac{C3R9R11R14(R12+R13)}{R13(R9R10+R9R11+R10R11)}\sqrt{\frac{R10}{C3C4R11R14R15}}
\end{aligned} \tag{1}
$$

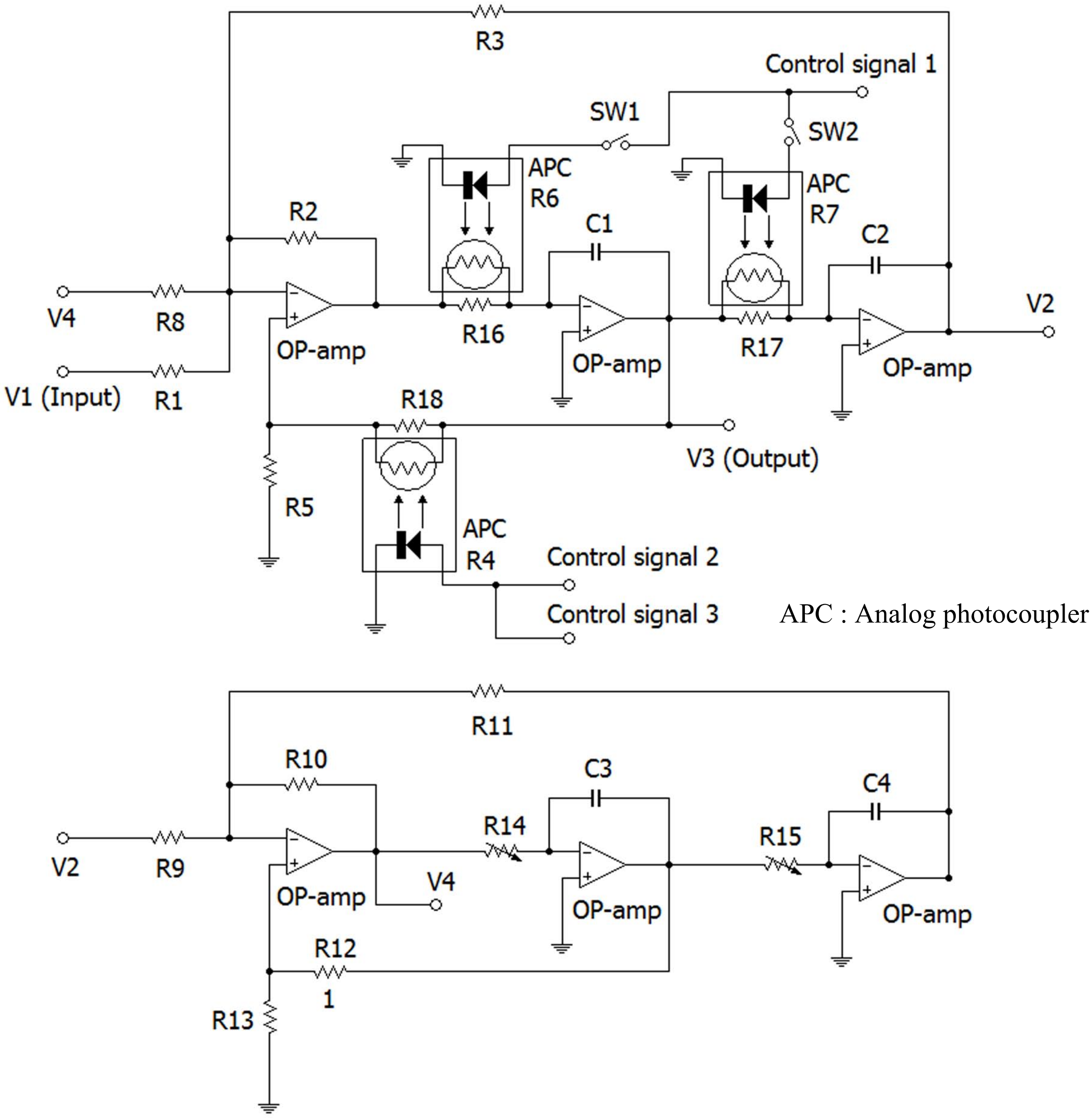


Figure 5. Designed state variable filter

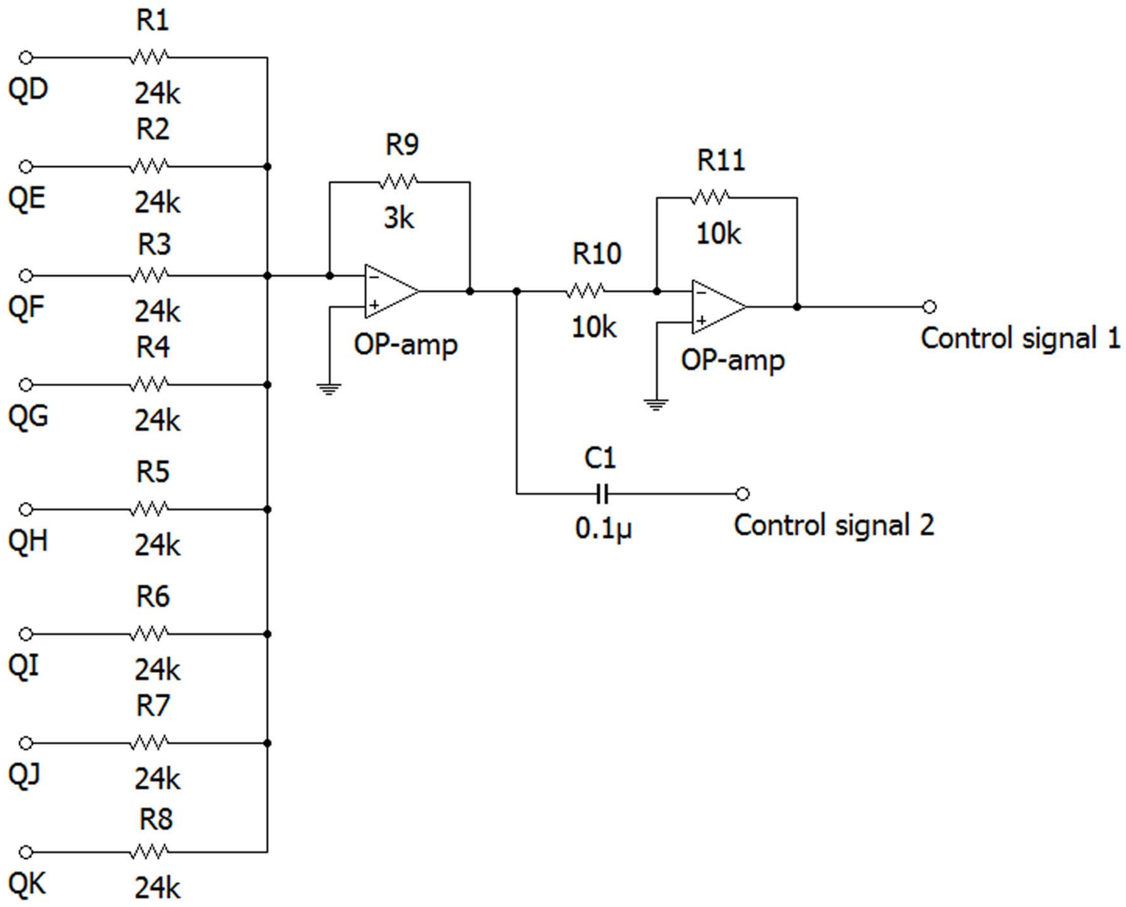
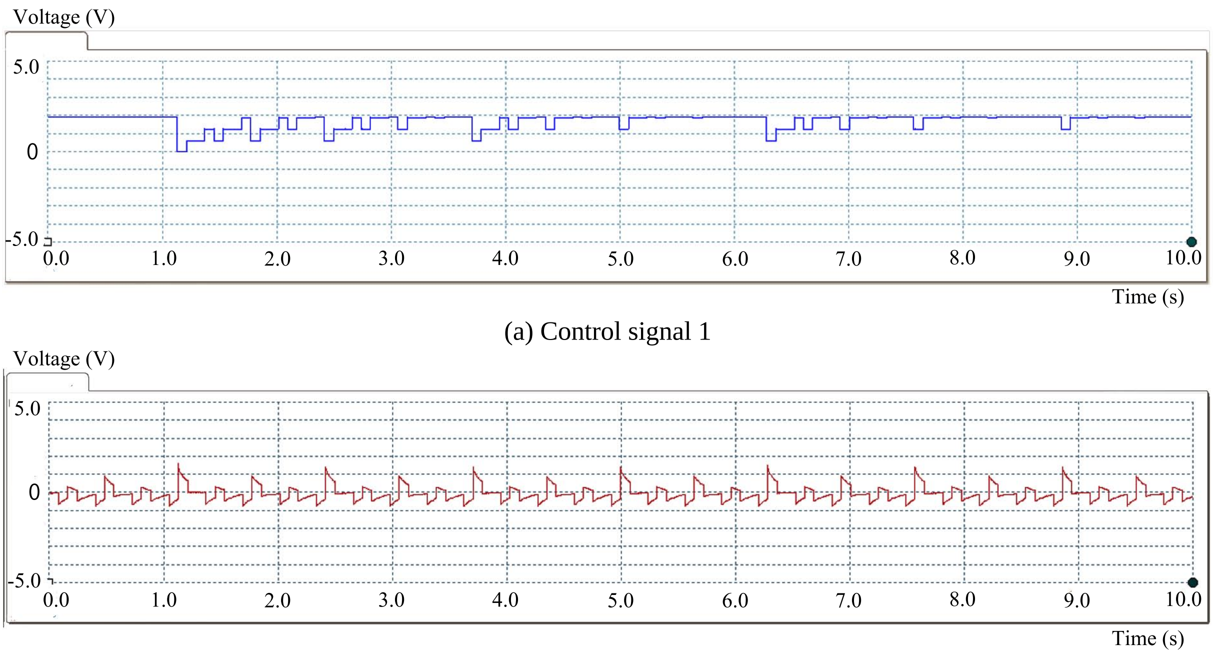


Figure 6. Adder



(a) Control signal 1

(b) Control signal 2

Figure 7. Control signals

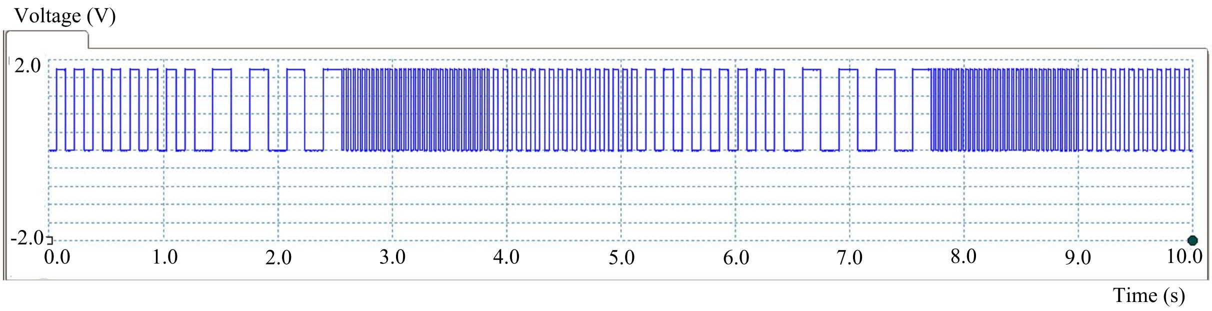


Figure 8. Output signal of multiplexer

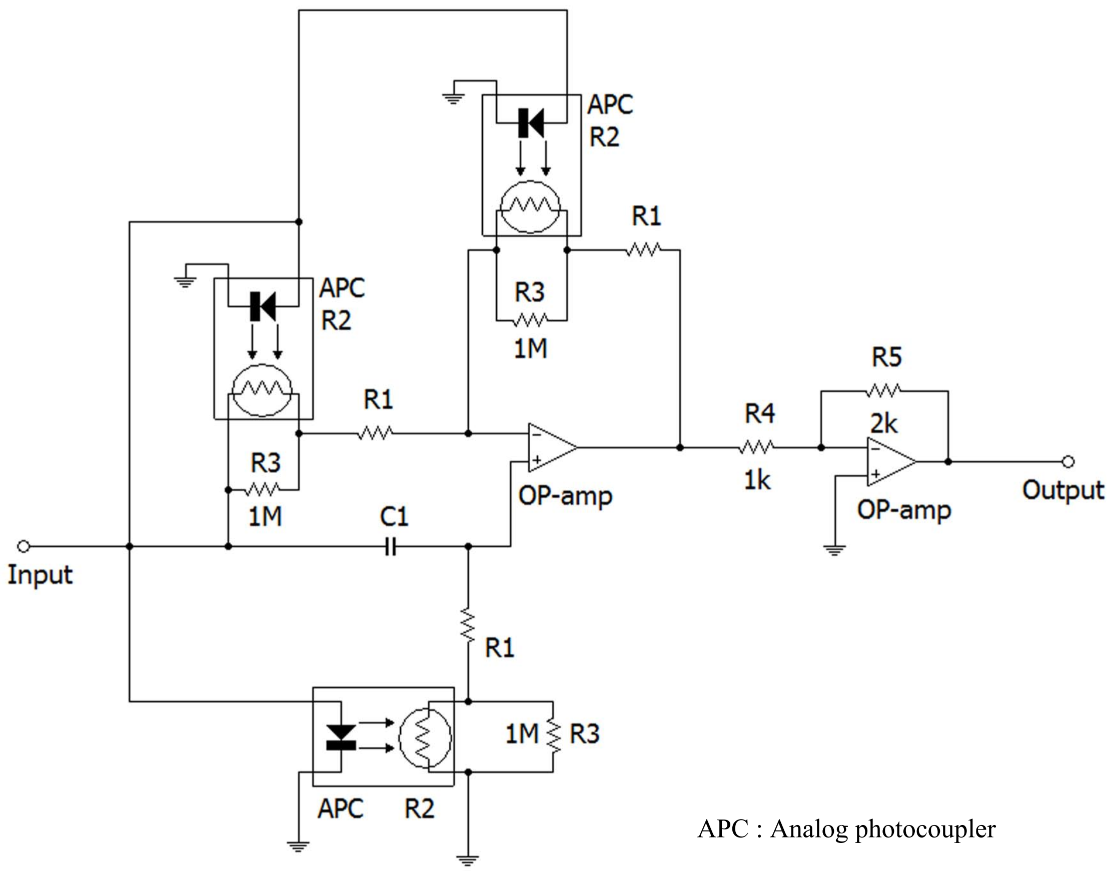


Figure 9. All-pass filter

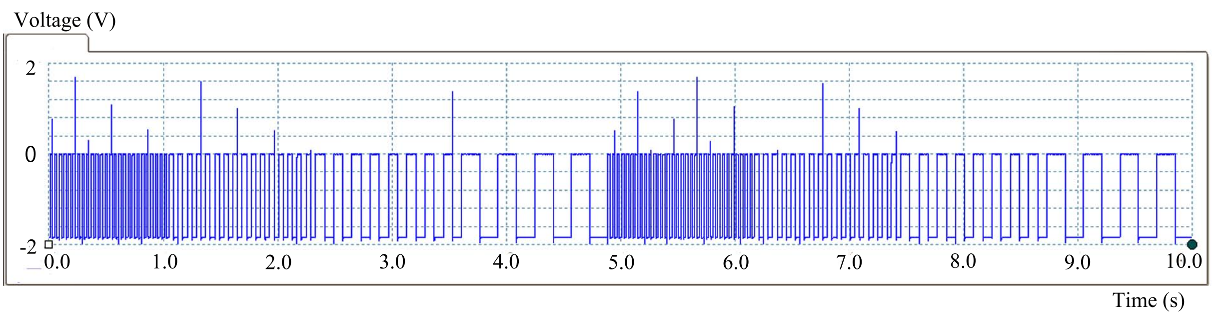


Figure 10. Output signal of all-pass filter

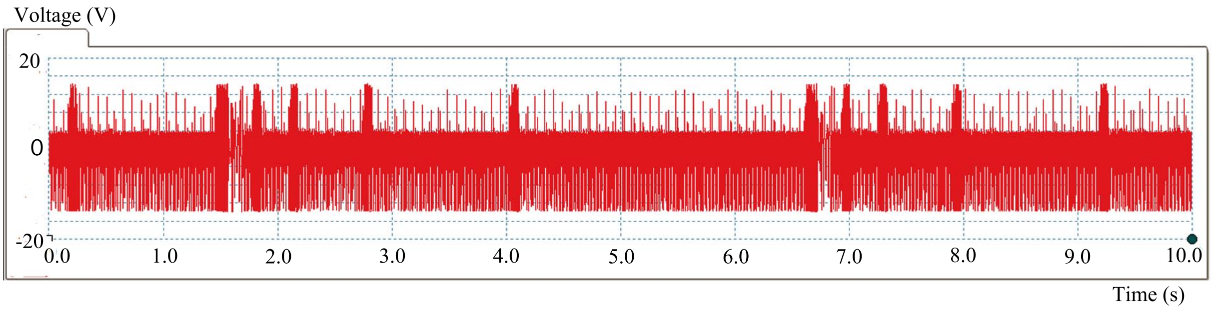
amount of available memory, and so forth. However, if this filter is realizable as plug-in software or an object of pure data (Pure Data), the use of computer signal processing provides greater flexibility than signal processing in hardware.

2. Design of State Variable Filter

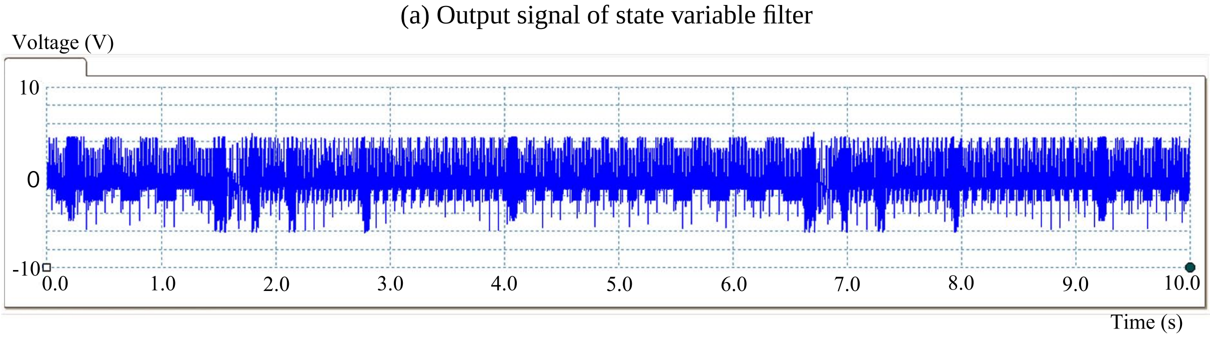
The state variable filter is represented in more detail in Figure 3. For our purposes, the influence of the high-pass filter is ignored; if its transfer function is set to 1, the whole filter is used for the second-order bandpass filter. The transfer function is expressed in T_1 of Formula 1. Here, T_{BP} and T_{HP} represent a general form of the second-order response characteristic. T_2 represents the transfer function of the high-pass filter. ω_1 and ω_2 represent the center and cutoff frequencies, respectively; Q_1 and Q_2 represent the sensitivity, and BW_1 represents the bandwidth. The sensitivity, center frequency and cutoff frequency of this filter were adjusted using the resistances $R4$, $R6$, $R7$, $R14$, and $R15$, before which frequency measurement was carried out. The frequency characteristic was measured using WaveGene and WaveSpectra software from efu (efu). As shown in Figure 4, the frequency characteristic is that of a bandpass filter. The newly designed state variable filter is shown in Figure 5. The resistances $R6$ and $R7$ of Figure 3 were replaced with analog photocouplers, which are in turn labeled $R6$ and $R7$ in Figure 5. The resistances $R16$ and $R17$ were connected in parallel to the analog photocouplers ($R6$ and $R7$, respectively). This added a limit to the center frequency of the lower bound of the designed bandpass filter. This particular filter obtains a constant pass band to the input signal. Control signal 1 is input into the analog photocouplers $R6$ and $R7$, and the center frequency of a bandpass filter is controlled. Similarly, the resistance $R4$ was also replaced with another analog photocoupler, connected in parallel to resistance $R18$, so as to limit the sensitivity. Furthermore, in order to control timbre, resistances $R14$ and $R15$ were replaced with variable resistors. Resistances $R16$, $R17$, and $R18$ may also be suitable for similar replacement. In order to minimize sensitivity of the high-pass filter, the resistance $R12$ was short-circuited as 1Ω.

3. Generation of Control Signal

An SN74HC4040 integrated circuit (IC) from Texas Instruments, Inc. was used for the binary counter, shown in Figure 2, that divides the frequency of an incoming PWM signal (Texas Instruments 2003). The adder adding the binary counter's output signals (QD through QK) is shown in Figure 6. Because the input signal is 5V, the output signal of the adder is also transformed to 5V. The adder is then placed in cascade connection with the inverting amplifier. Control signal 1, which controls the center frequency, shifts the phase to a positive phase. Control signal 2, controlling the sensitivity, uses a capacitor of 0.1μF to function as the high-pass filter. Control signals 1 and 2 are shown in Figure 7. The voltage of a control signal can be decreased by the influence of the light-emitting diode of an analog photocoupler. By transforming a digital signal into an analog signal, Figure 7 shows that signal voltage changes in various ways. Thereby, the center frequency and sensitivity can be controlled intricately. A Toshiba Corp. TC74HC153AP multiplexer IC was used to change signals QB, QC, QD, and QE: four outputs from the binary counter (Toshiba Corp. 2014). The output signals QH and QI from the binary counter were used for address information of the multiplexer. The output signal of the multiplexer is shown in Figure 8. As indicated in Figure 8, the output of the multiplexer is variable, with four different output frequencies shown. An all-pass filter is shown in Figure 9. In this case, the output signal of the multiplexer was used for the input signal of the all-pass filter, and the control signal of an analog photocoupler. As seen in Formula 2, the all-pass filter was designed such that θ_d shifts to $170°$ at a frequency of ˜185.7Hz. In this case, the value of resistance for an analog photocoupler is considered to be negligible. In order to limit phase shifts at low frequency, a 1 MΩ resistance was connected in parallel with the analog photocoupler from figure 9 (Otsuka 2011). This allows adjustment of the timbre of low frequency audio. The output signal of an all-pass filter is shown in Figure 10. Because the phase of the output signal of the all-pass filter was inverted, and its output signal decreased by the influence of an analog photocoupler, an inverting amplifier was placed in cascade connection with the all-pass filter. This shifted the phase to a positive phase and amplified the output signal. The output signal of the state variable filter and the output signal of the all-pass filter were then added, and together formed the output signal of this filter. The output signal of this filter and the output signal of a state variable filter are shown in Figure 11. Figure 11 shows that the output signal of this filter differs from the output signal of a state variable filter. The input signal of this filter is a PWM signal with a frequency of ˜101 Hz and a duty ratio of ˜33.3% as shown in Figures 7, 8, 10, and 11. As seen in Figure 1, a PWM signal is output from a USB device by the computer synthesizer. When a PWM signal is not output from a USB device by the computer synthesizer, then, if the sensitivity of the filter is high, the filter can be oscillated. In this filter, a 5V control signal, labeled as Control Signal 3 in Figure 2, is output from a USB device. This control signal is input into the analog photocoupler $R4$ of Figure 5, which controls the sensitivity of the state variable filter; the sensitivity of the filter is low in this design. For this filter, USB-FSIO30 of Km2net was used as a USB device (Komatsu 2011).



(a) Output signal of state variable filter



(b) Output signal of developed filter

Figure 11. Output signal of filters

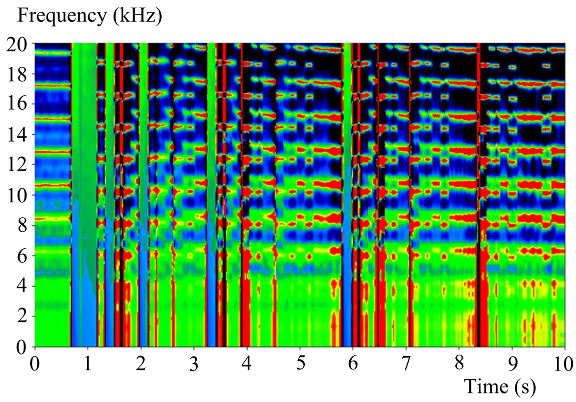


Figure 12. Frequency analysis of an output signal

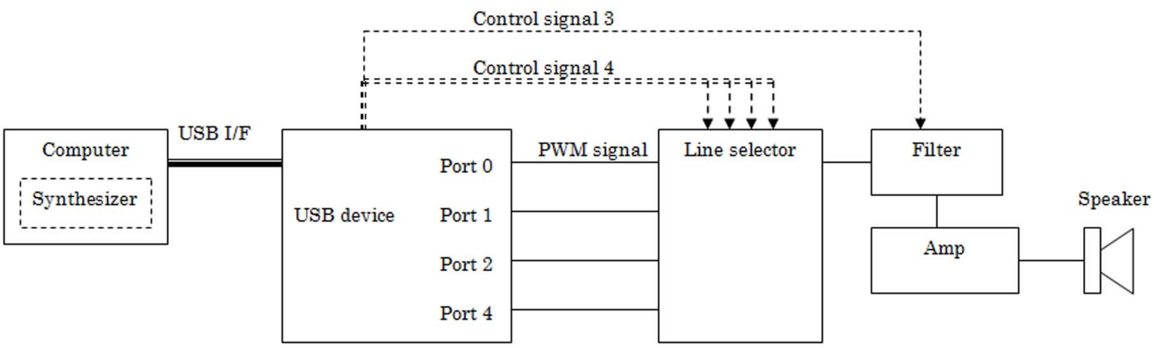


Figure 13. Synthesizer system using the developed filter

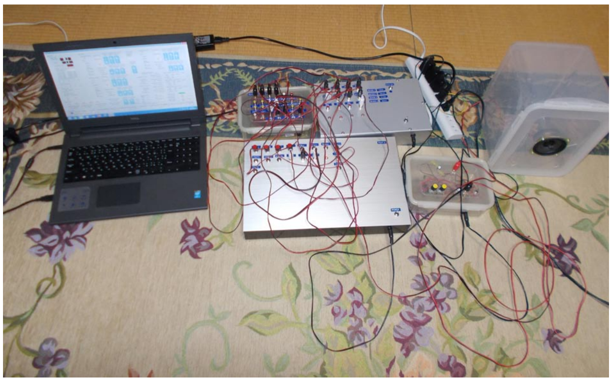


Figure 14. Actual synthesizer system

$$R1C1 = -\frac{tan\frac{1}{2}(180° + \theta_d)}{\omega_d} \quad (2)$$

4. Signal Analysis

Frequency analysis was carried out on the output signal of this filter. As discussed, the input signal into the filter was a PWM signal with a frequency of ˜101Hz and a duty ratio of ˜33.3%. The frequency analysis of the resulting output signal is shown in Figure 12. The output signal was digitized by a sampling frequency of 44.1 kHz, with a 16-bit linear quantization. The system requests a power spectrum by Burg's method, with a 48th order auto regression coefficient (Ishihara et al. 1988). One frame carries out a frequency analysis on 100 ms (4410 samples) of the audio data. Next, the sample numbers of one frame are shifted by 20 ms (882 samples), and the new audio data for an additional 20 ms are input. This operation is repeated, and a frequency analysis is carried out for a total of 10 s. The number of samples of audio data $x(n)$ is set to N, the frame number is set to e, and the number of shifts is set to s; $x(n)$ is expressed by formula 3. From Figure 12, we observe that the power spectrum is high locally for frequency components greater than ˜6 kHz.

$$x(n)\ (n = e \cdot s, e \cdot s + 1, e \cdot s + 2, ..., e \cdot s + (N-1), \\ e = 0, 1, 2, ..., 495, s = 882, N = 4410) \quad (3)$$

5. Actual Performance

The designed synthesizer system, combining the computer synthesizer, the USB device, the line selector, and this filter design discussed above, is shown in Figure 13. An actual photograph of the system is shown in Figure 14. To review, a PWM signal is output from the output terminals, Ports 0, 1, 2, and 4 of the USB device. The four PWM signals are input into a line selector, and a PWM signal of them is connected to this filter by Control signal 4 (Tsuji 2018). The output signals of Ports 0, 1, and 2 are random signals of the PWM signals or the modulated PWM signals by the low frequency oscillator. The output signal of Port 4 is a rhythm signal of the PWM signal. These signals are output from the USB device by the sequencer of the computer synthesizer. The PWM signal that changed the frequency and voltage (duty ratio) periodically was used instead of the modulated PWM signal by the low frequency oscillator. The rhythm signal of the PWM signal of Port 4 is a triple rhythm signal based on twelve steps (Tsuji 2015). Testing has confirmed that the feedback sound of this filter was controlled by the PWM signal, and that the timbre could be changed by changing the center frequency of a bandpass filter that turned off the switches SW1 and SW2 of Figure 5, or by changing the cutoff frequency of the high-pass filter.

6. Conclusion

This study attempted cascade connection of two state variable filters. The output of the low pass filter of the first state variable filter was input into the second , the output of the high-pass filter of the second state variable filter was input into the first filter, and the output signal of the bandpass filter of the first state variable filter was set as the output of the whole filter. The frequency of the PWM signal of an input signal was divided and analog calculation was carried out by using the divided components. Using the result of the analog calculation as a control signal, this filter tried to control the center frequency and sensitivity of the first state variable filter. Furthermore, four signals that divided the frequency of the PWM signal were adjusted, and a control signal was created by controlling the phase of this signal. This control signal was then used to modulate the output signal of a state variable filter by adding it to that output signal. Testing of the resulting filter design has confirmed that feedback sound was controlled by the PWM signal, and rich sound effects were obtained. This leads logically to further possibilities for developing synthesizer systems that use combinations of analog signal processing and digital signal processing.

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7. Author’s Profile

Ichiro TSUJI

Ichiro Tsuji was born in 1966 and graduated from the Department of Electrical Engineering Faculty of Technology of Kokushikan University in 1991. He joined NEC Home Electronics Corp. the same year. At the Development Research Laboratory, he engaged in research of a three-dimensional playback system for a two-channel speaker. He then moved to NEC Corp. and engaged in the research and development of a multimedia-related project. He retired in 1998. He is currently a regular member of the Acoustical Society of Japan.

As for his musical activities, in Tokyo in 1986, he started working on noise/industrial music for his band named “Dissecting Table.” He returned to his hometown of Hiroshima in 1998, and has been pursuing musical activities ever since. His records and compact disks have been released under the independent label of the UPD organization, and under labels in Europe and the United States. Since 2011, the works have been produced by controlling PWM signals output from a USB device on a computer.