Research Report

Investigation of the sound synthesis method using an analog filter

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Abstract

This paper describes the signal of a rich sound effect obtained from the output signal of an analog filter that implemented a cascade connection of a Sallen-Key (Sallen and Key 1995) and biquad circuit (Van Valkenburg 1989). Although a noninverting amplifier is commonly used with Sallen-Key circuits (Jung 2002), here it is replaced with a first-order low-pass filter. The sensitivity is thereby changed, and the timbre of the filter is improved. The resistance that controls the cutoff frequency and center frequency of the Sallen-Key circuit and the biquad circuit uses an analog photo coupler. The control signal output from a USB device is input into an analog photo coupler, and the cutoff and center frequencies are controlled. Because a control signal can be created by the program on a computer, it is possible to make any sound flexibly. Moreover, the output signal of the filter made by the cascade connection of the Sallen-Key circuit and the biquad circuit creates a rich sound effect by conducting signal analysis of an input signal and the output signal. Although the gain of the Sallen-Key circuit enlarges the value, the structure, methodology, and characteristics of the filter described can keep the value of sensitivity low.

1. Introduction

At the 29th the Japanese Society for Sonic Arts (JSSA) regular meeting, my previous paper (Tsuji 2016) described the resistance of the photo register cadmium sulfide cell (CDS), which can be modified to change slightly by changing the brightness of a light-emitting diode (LED) built into the analog photo coupler. By using this characteristic, an analog photo coupler was used for the resistance to control the sensitivity and center frequency of the biquad circuit, and the control signal output from a USB device was input into the analog photo coupler. In this study, an analog photo coupler is used for the resistance controlling the cutoff and center frequencies of the Sallen-Key circuit. The control signal output from a USB device is input into an analog photo coupler, as in the previous study. The Sallen-Key circuit has parallel connections of three filters, and the biquad circuit has parallel connections of two filters and a bypass circuit. By implementing a cascade connection of the Sallen-Key circuit and the biquad circuit, this filter allows the production of a more complicated sound.

2. Computer and analog filter

One computer can easily carry out digital sound synthesis using plug-in software such as Digital Audio Workstation, created by Pure Data (Pure Data). When synthesizing analog sound using hardware, the scale of the hardware becomes large as the sound synthesis becomes more complicated. However, analog sound synthesis makes it possible for an acoustic feature to create a characterful timbre, unlike digital sound synthesis. In this study, the computer operation of the developed filter is taken into consideration, realizing a timbre that cannot be created by human beings. It is thereby possible to create a characterful timbre containing a digital component (computer) and an analog component (analog filter).

3. Design of analog filter

The basic structure of the designed analog filter is shown in Figure 1. From Figure 1, it can be seen that the Sallen-Key circuit consists of filter 1, filter 2, and filter 3, and implements a parallel connection of each filter. The biquad circuit has a parallel connection of a second-order low-pass filter, a second-order bandpass filter, and a bypass circuit. The following sections describe the structure and characteristics of each designed filter. The frequency characteristic was measured using WaveGene and WaveSpectra software by efu (efu). In this study, the voltage of an input signal was divided to filter 1 and filter 2 at $\frac{1}{5}$, was input into filter 1 and filter 2, sent to filter 3 at $\frac{1}{20}$, and finally input into filter 3.

3.1. Structure and characteristics of filter 1

The circuit structure is shown in Figure 2 for the conventional low-pass filter (Sallen and Key 1995) of the Sallen-Key circuit and filter 1. Although the conventional low-pass filter controls the gain using a noninverting amplifier, it is not considered for control of sensitivity. Filter 1 is introduced for designing the second-order low-pass filter. In the course of this design procedure, it can improve the timbre by sensitivity and gain control using the first-order low-pass filter. The inverting amplifier implemented a cascade connection with the first-order low-pass filter so that the gain is positive and the sensitivity can be increased. Furthermore, the analog photo coupler was

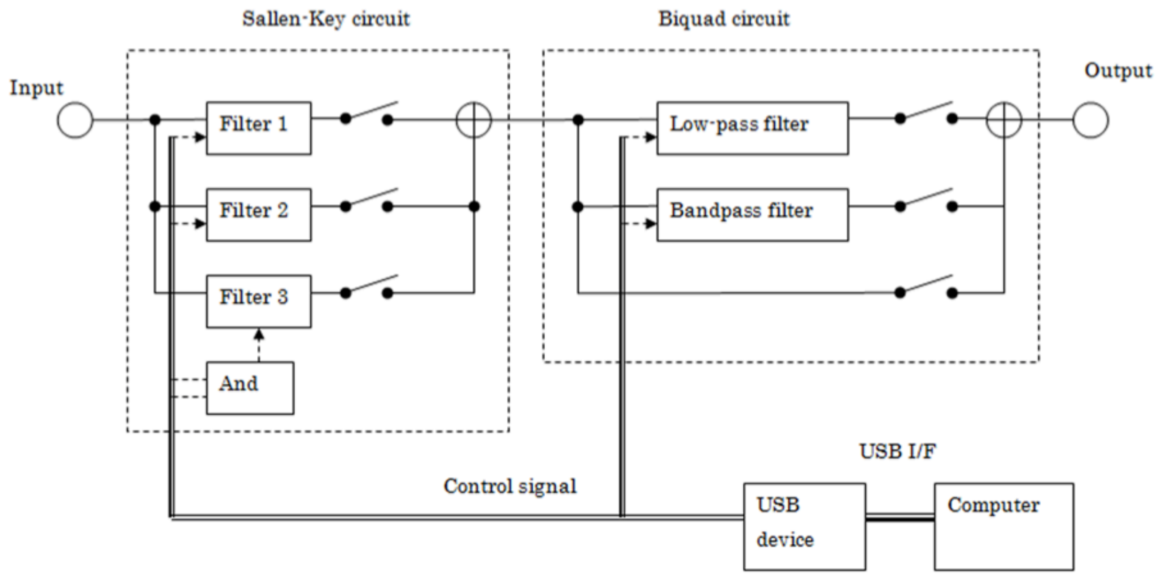


Figure 1. Basic structure of an analog filter

used for the resistance that controls the cutoff frequency of the second-order low-pass filter. A control signal is input into an analog photo coupler from the digital I/O port of a USB device, thereby making the cutoff frequency freely controllable from a computer. The transfer function of filter 1 is expressed by formula 1. T_0 represents the transfer function of the first-order low-pass filter. The cutoff frequency ω_0 of the first-order low-pass filter is expressed by formula 2. The influence of the first-order low-pass filter is ignored, and if T_0 is set to 1, formula 1 is used for the second-order low-pass filter. The cutoff frequency ω_1, sensitivity Q_1, and gain K_1 of the second-order low-pass filter are expressed by formula 3. The formula of Q_1 expresses the oscillation of the second-order low-pass filter, when the value of K_1 is 3. Frequency characteristics are shown in Figure 3, in the cases where the values of K_1 of filter 1 are 3, 4, and 5, and also when the first-order low-pass filter was replaced with an inverting amplifier. When the frequency measurement is carried out, the analog photo coupler is replaced with a resistance of 1 kΩ. In Figure 3(a), the characteristics in red represent the frequency characteristics when the value of K_1 is 3, and those in blue represent the frequency characteristics when the value of K_1 is 4. In Figure 3(c), the characteristics in red represent the frequency characteristics when filter 1 was used as the first-order low-pass filter, while those in blue represent the frequency characteristics when it was used as an inverting amplifier; then the value of K_1 is 4. Filter 1 can guess that the value of Q_1 also increases if the value of K_1 increases by approximately three or more. When the value of K_1 is 5, it begins to oscillate. When the first-order low-pass filter was replaced with an inverting amplifier, the cutoff frequency and the sensitivity of filter 1 was somewhat higher than when filter 1 was used as the first-order low-pass filter. From formula 2, when the value of K_1 is 3, the cutoff frequency ω_0 of the first-order low-pass filter is ˜16076.3 Hz; when the value of K_1 is 4, it is ˜12057.2 Hz; and when the value of K_1 is 5, it is ˜9645.8 Hz. From formula 3, the cutoff frequency ω_1 of the second-order low-pass filter is ˜1591.5 Hz.

$$T_1 = \frac{K_1(\frac{1}{C1R1})^2 T_0}{s^2 + \frac{1}{C1R1}(3 - K_1 T_0)s + (\frac{1}{C1R1})^2}$$

$$T_0 = \frac{\frac{1}{C2R3}}{s + \frac{1}{C2R3}} \quad (1)$$

$$\omega_0 = \frac{1}{C2R3} \quad (2)$$

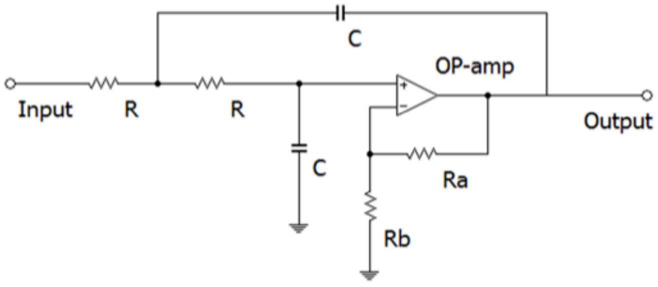
$$\omega_1 = \frac{1}{C1R1}$$

$$Q_1 = \frac{1}{3 - K_1}$$

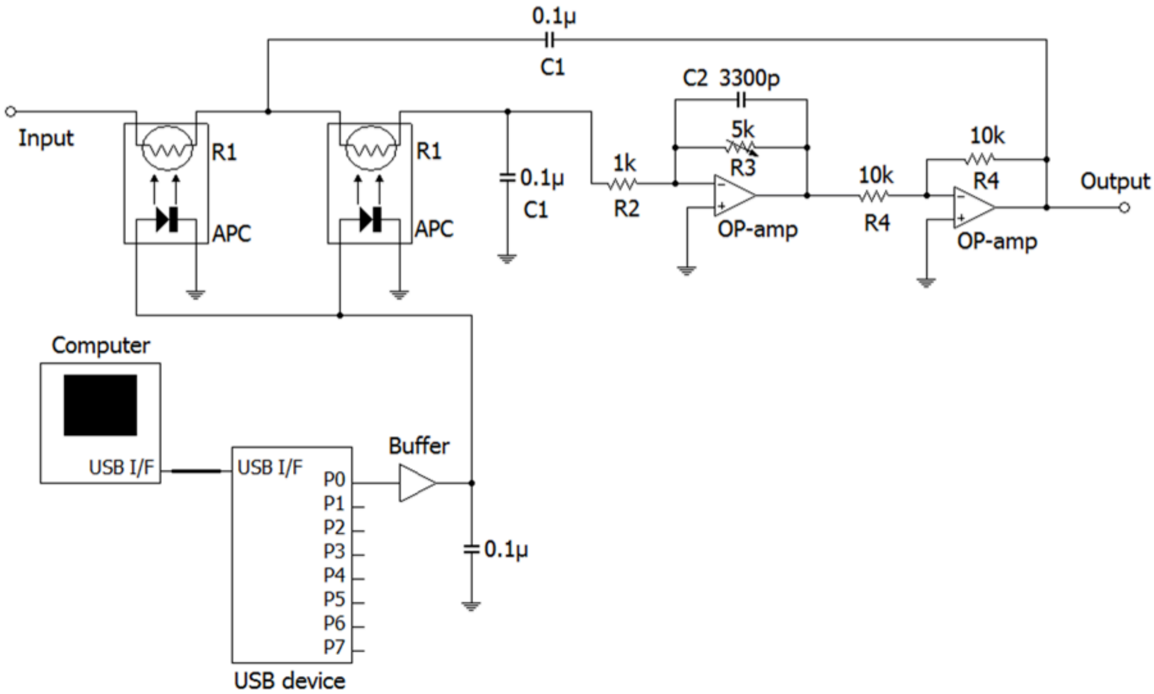
$$K_1 = \frac{R3}{R2} \quad (3)$$

3.2. Structure and characteristics of filter 2

The circuit structure for the conventional high-pass filter (Sallen and Key 1995) of the Sallen-Key circuit and filter



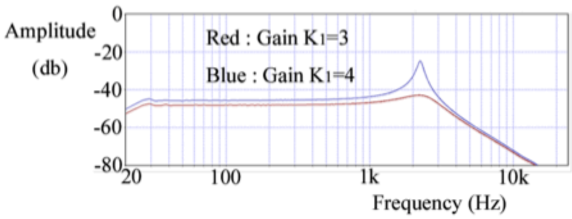
(a) Conventional low-pass filter



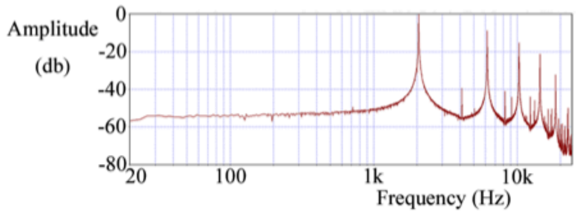
(b) Filter 1

APC : Analog photo coupler

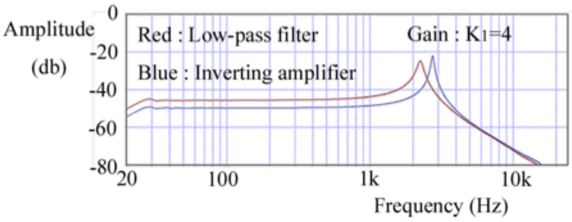
Figure 2. Circuit structure of Filter 1



(a) Gain K_1=3 and K_1 = 4

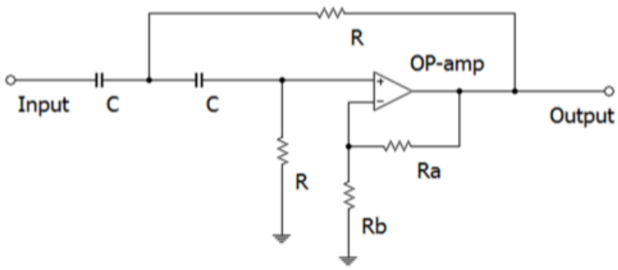


(b) Gain K_1=5

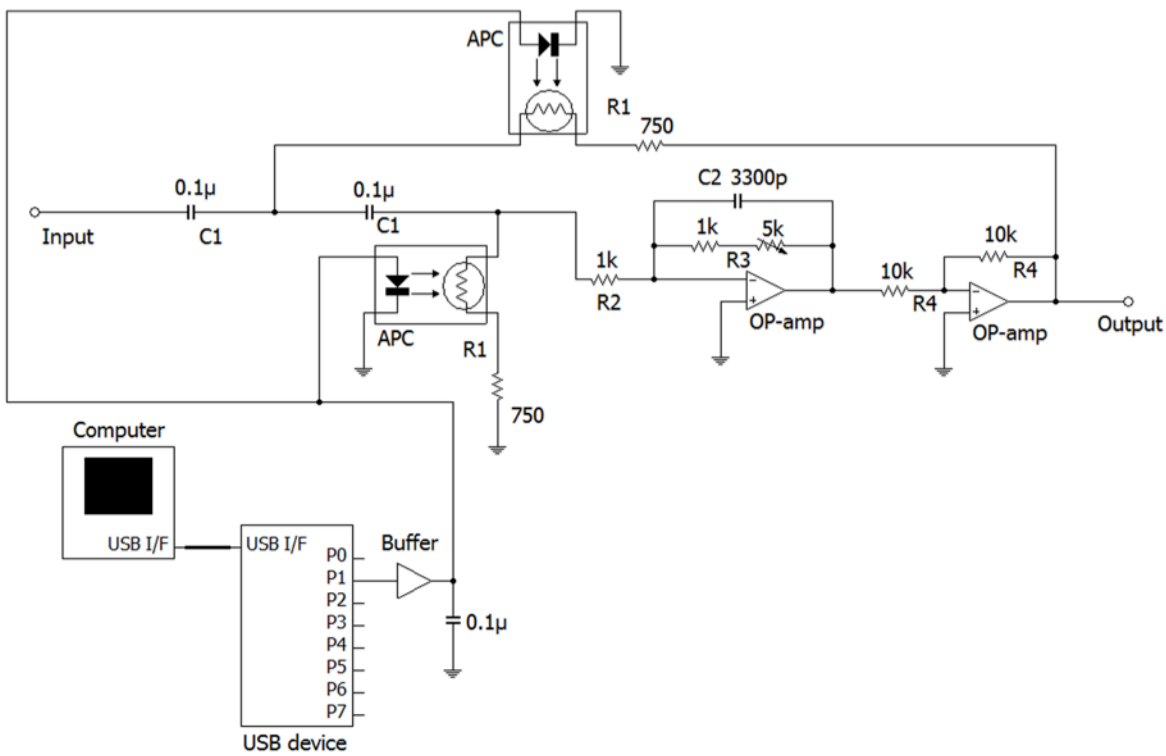


(c) Filter 1 using an inverting amplifier

Figure 3. Frequency characteristics of Filter 1



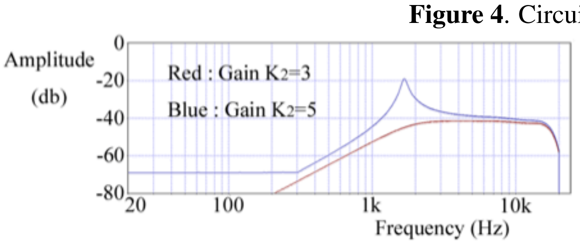
(a) Conventional high-pass filter



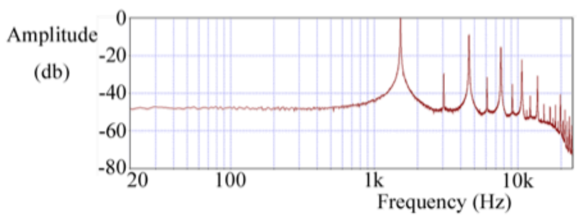
(b) Filter 2

APC : Analog photo coupler

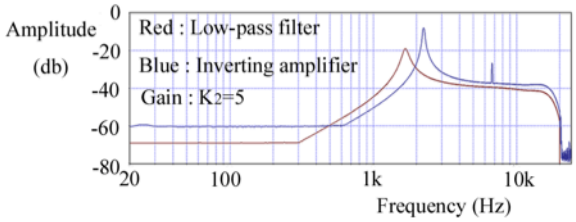
Figure 4. Circuit structure of Filter 2



(a) Gain K_2=3 and K_2=5

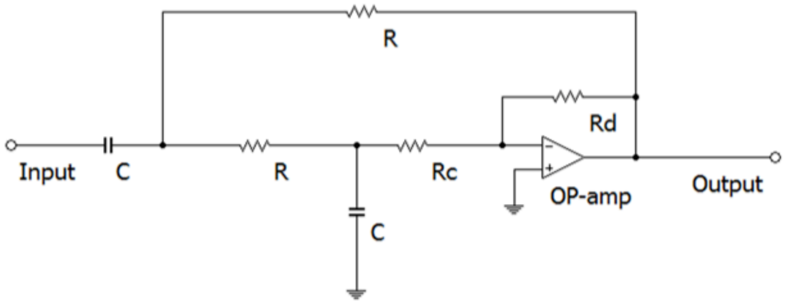


(b) Gain K_2=6

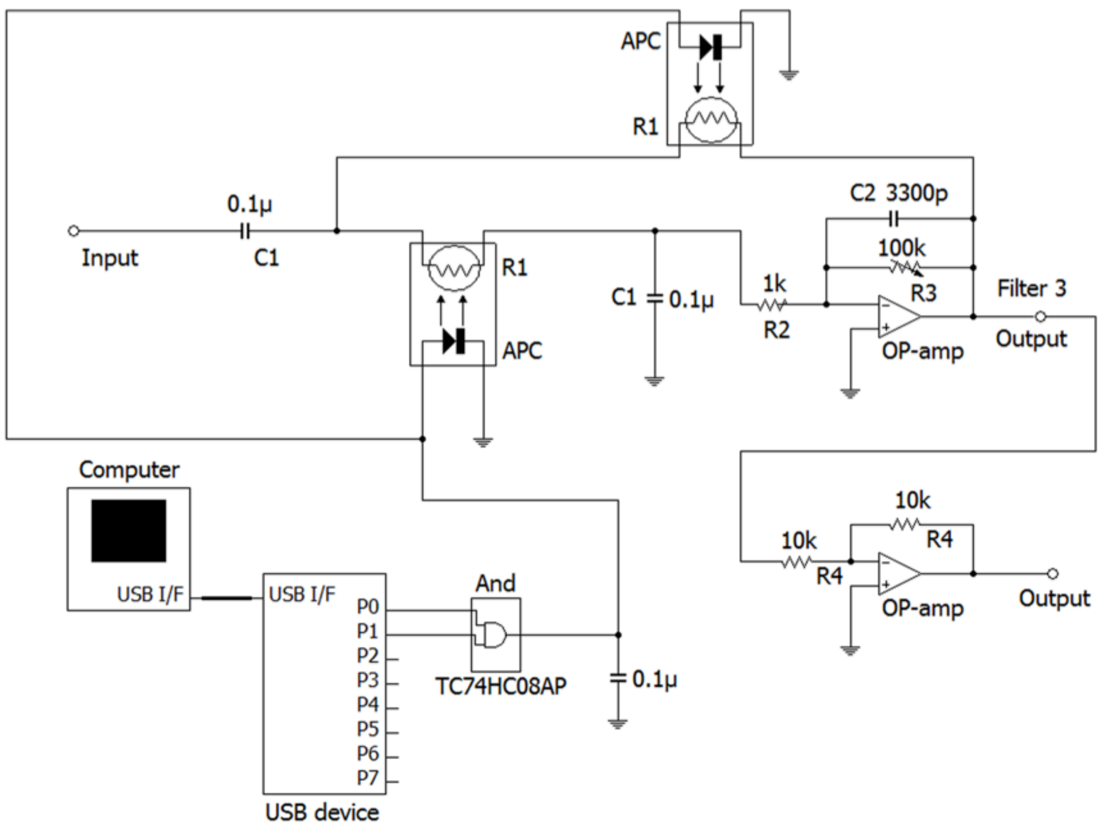


(c) Filter 2 using an inverting amplifier

Figure 5. Frequency characteristics of Filter 2



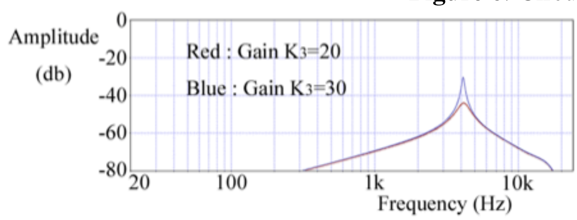
(a) Conventional bandpass filter



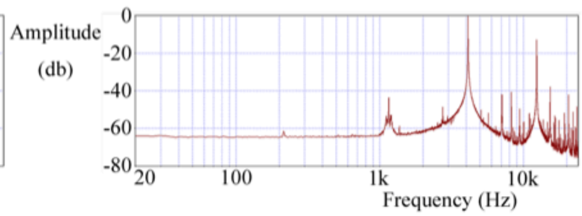
(b) Filter 3

APC : Analog photo coupler

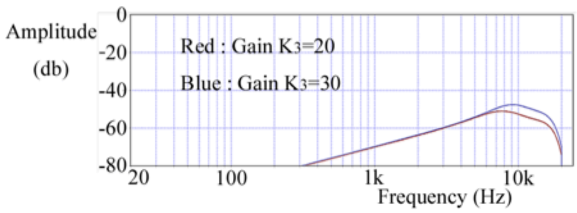
Figure 6. Circuit structure of Filter 3



(a) Gain K3=20 and K3=30

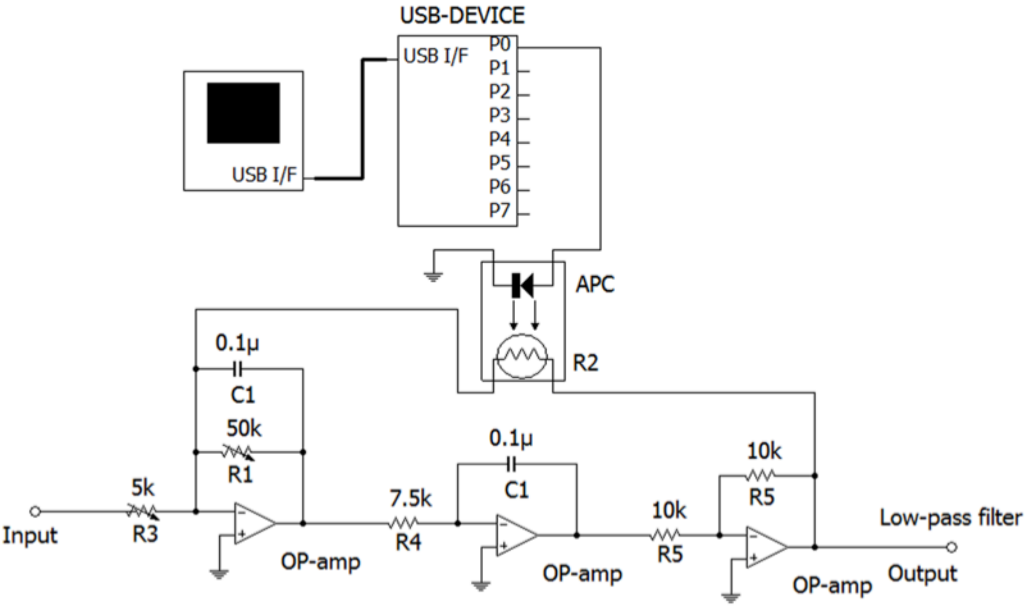


(b) Gain K3=35

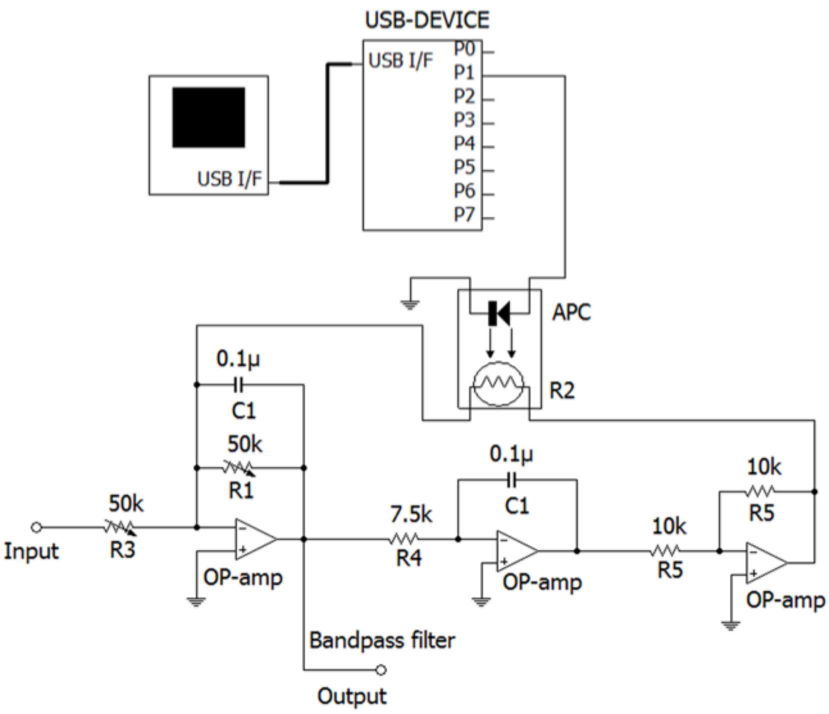


(c) Filter 3 using an inverting amplifier

Figure 7. Frequency characteristics of Filter 3



(a) Low-pass filter that controls the cut-off frequency



(b) Bandpass filter that controls the center frequency

APC : Analog photo coupler

Figure 8. Biquad circuit that controls cut-off frequency and center frequency

2 is shown in Figure 4. Although the conventional high-pass filter controls the gain using a noninverting amplifier, it is not considered for the control of sensitivity. Filter 2 is presented for the design of the second-order high-pass filter. In the process of this design, it can improve the timbre, like filter 1, using a first-order low-pass filter, by controlling the gain and sensitivity. The inverting amplifier implements a cascade connection with the first-order low-pass filter so that the gain is positive and the value of a sensitivity can be increased. Furthermore, the analog photo coupler is used for the resistance, which controls the cutoff frequency of the second-order high-pass filter. A control signal is input into an analog photo coupler from the digital I/O port of a USB device, and the cutoff frequency is freely controllable from a computer. From Figure 4, the resistance $R1$ is set to 750 Ω when cascaded with the value of an analog photo coupler. This adds a limit to the cutoff frequency of the upper bound of the second-order high-pass filter, and this particular filter obtains a constant pass band to the input signal. The transfer function of filter 2 is expressed by formula 4. T_0 represents the transfer function of the first-order low-pass filter. The cutoff frequency ω_0 of the first-order low-pass filter is calculated by the same method as in formula 2. The influence of the first-order low-pass filter is ignored, and if T_0 is set to 1, formula 4 is used for the second-order high-pass filter. The cut-off frequency ω_2, sensitivity Q_2, and gain K_2 of the second-order high-pass filter are expressed by formula 5. From Figure 4, the resistance $R3$ is 1 kΩ, in a cascade connection with a variable resistance of 5 kΩ. The value of K_2 can be set to 6. When the value of K_2 is 3, the formula of Q_2 oscillates with the second-order high-pass filter. The frequency characteristics are shown in Figure 5, in the cases where the values of K_2 of filter 2 are 3, 5, and 6, and also when the first-order low-pass filter was replaced with an inverting amplifier. When the frequency measurement is carried out, the analog photo coupler is replaced with a resistance of 1 kΩ. From Figure 5(a), the characteristics in red represent the frequency characteristics when the value of K_2 is 3, and those in blue represent the frequency characteristics when the value of K_2 is 5. In Figure 5(c), the characteristics in red represent the frequency characteristics when filter 2 was used as the first-order low-pass filter, while those in blue represent the frequency characteristics when it was used an inverting amplifier; then the value of K2 is 5. When the first-order low-pass filter was replaced with an inverting amplifier, filter 2 began to oscillate slightly. Filter 2 can guess that the value of Q_2 also increases, if the value of K_2 increases significantly from 3. When the value of K_2 is 6, it begins to oscillate. From formula 2, when the value of K_2 is 3, the cutoff frequency ω_0 of the first-order low-pass filter is ˜16076.3 Hz; when the value of K_2 is 5, it is ˜9645.8 Hz; and when the value of K_2 is 6, it is ˜8038.1 Hz. From formula 5, the cutoff frequency ω_2 of the second-order high-pass filter is ˜1591.5 Hz.

$$\begin{aligned}
T_2 &= \frac{K_2 s^2 T_0}{s^2 + \frac{1}{C1R1}(3 - K_2 T_0)s + (\frac{1}{C1R1})^2} \\
T_0 &= \frac{\frac{1}{C2R3}}{s + \frac{1}{C2R3}} && (4)
\end{aligned}$$

$$\begin{aligned}
\omega_2 &= \frac{1}{C1R1} \\
Q_2 &= \frac{1}{3 - K_2} \\
K_2 &= \frac{R3}{R2} && (5)
\end{aligned}$$

3.3. Structure and characteristics of filter 3

The circuit structure is shown in Figure 6 for the conventional bandpass filter (Sallen and Key 1995) of the Sallen-Key circuit and filter 3. Although the conventional bandpass filter controls the gain using an inverting amplifier, it is not considered to control sensitivity. Filter 3 is set up for the design of the second-order bandpass filter. During this design process, it can improve timbre, like filter 1, using the first-order low-pass filter by controlling the gain and sensitivity. Moreover, so that the output signal of filter 3 is carried out in phase with the phase of filter 1 and filter 2, the inverting amplifier is cascaded with filter 3. Furthermore, the analog photo coupler is used for the resistance that controls the center frequency of the second-order bandpass filter. A control signal is input into an analog photo coupler from the digital I/O port of a USB device. The center frequency is thereby freely controllable from a computer. The transfer function of filter 3 is expressed by formula 6. T_0 represents the transfer function of the first-order low-pass filter. The cutoff frequency ω_0 of the first-order low-pass filter is calculated by the same method as in formula 2. The influence of the first-order low-pass filter is ignored, and if T_0 is set to 1, formula 6 is used for the second-order bandpass filter. The center frequency ω_3, sensitivity Q_3, and gain K_3 of the second-order bandpass filter are expressed by formula 7. Frequency characteristics are shown in Figure 7, in the cases where the values of K_3 of filter 3 are 20, 30, and 35, and also when the first-order low-pass filter was replaced with an inverting amplifier. When the frequency measurement is carried out, the analog photo coupler is replaced with a resistance of 1 kΩ. From Figure 7(a), the characteristics in red represent the frequency characteristics when the value of K_3 is 20, and those in blue represent the frequency characteristics when the value of K_3 is 30. It turns out that the value of Q_3 also increases gradually if the value of K_3 is increased, and it oscillates when the value of K_3 is 35. From formula 2, when the value of K_3 is 20, the cutoff frequency ω_0 of the first-order low-pass filter is ˜2411.4 Hz; when the value of K_3 is 30, it is ˜1607.6 Hz;

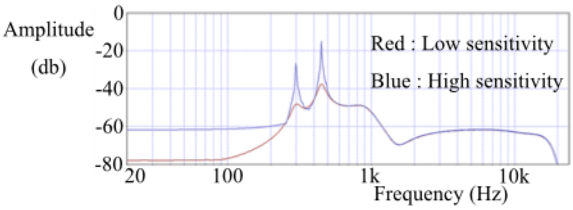


Figure 9. Frequency characteristics of the whole filter

and when the value of K_3 is 35, it is ˜1378 Hz. From formula 7, when the value of K_3 is 20, the center frequency ω_3 of the second-order bandpass filter is ˜7293.4 Hz; when the value of K_3 is 30, it is ˜8861.4 Hz; and when the value of K_3 is 35, it is ˜9549.3 Hz. The center frequency of the measured filter 3 is ˜4136.7 Hz. Because this is considered to be influenced by the first-order low-pass filter, the frequency characteristics when the first-order low-pass filter was replaced with an inverting amplifier are shown in Figure 7(c). When the value of K_3 is 20, the center frequency of filter 3 is ˜7643.6 Hz, and when the value of K_3 is 30, it is ˜9049.8 Hz.

$$
\begin{aligned}
T_3 &= \frac{-K_3 \dfrac{1}{C1R1} s T_0}{s^2 + \dfrac{3}{C1R1}s + \dfrac{1+K_3T_0}{(C1R1)^2}} \\
T_0 &= \frac{\dfrac{1}{C2R3}}{s + \dfrac{1}{C2R3}} \qquad (6)
\end{aligned}
$$

$$
\begin{aligned}
\omega_3 &= \frac{\sqrt{1+K_3}}{C1R1} \\
Q_3 &= \frac{\sqrt{1+K_3}}{3} \\
K_3 &= \frac{R3}{R2} \\
BW_3 &= \frac{\omega_3}{Q_3} \qquad (7)
\end{aligned}
$$

4. Structure of the biquad circuit and characteristics of the whole filter

By the same method as reported at the 29th JSSA regular meeting (Tsuji 2016), the biquad circuit was composed of the second-order low-pass filter, the second-order bandpass filter, and the bypass circuit. The transfer function of the second-order low-pass filter is expressed by formula 8, and the transfer function of the second-order bandpass filter is expressed by formula 9. The circuit of each filter is shown in Figure 8. The analog photo coupler was used for the resistance that controls the cutoff frequency of the second-order low-pass filter and the center frequency of the second-order bandpass filter. A control signal is input into an analog photo coupler from the digital I/O port of a USB device. The cutoff frequency and center frequency are freely controllable from a computer. Next, the transfer function of the filter is expressed as formula 10, which implements a cascade connection of the Sallen-Key circuit and the biquad circuit, and an example of a frequency characteristic is shown in Figure 9. When frequency measurement is carried out, the analog photo coupler was replaced with a variable resistance. The characteristics in red represent the frequency characteristics when the sensitivity of the biquad circuit is low, and those in blue represent the frequency characteristics when the sensitivity of the biquad circuit is high.

$$
\begin{aligned}
T_4 &= \frac{\dfrac{-1}{C1^2R3R4}}{s^2 + \dfrac{1}{C1R1}s + \dfrac{1}{C1^2R2R4}} \\
\omega_4 &= \frac{1}{C1}\sqrt{\frac{1}{R2R4}} \\
Q_4 &= C1R1\omega_4 \\
K_4 &= \frac{R2}{R3} \qquad (8)
\end{aligned}
$$

$$
\begin{aligned}
T_5 &= \frac{\dfrac{-1}{C1R3}s}{s^2 + \dfrac{1}{C1R1}s + \dfrac{1}{C1^2R2R4}} \\
\omega_5 &= \frac{1}{C1}\sqrt{\frac{1}{R2R4}} \\
Q_5 &= C1R1\omega_5 \\
K_5 &= \frac{R1}{R3} \\
BW_5 &= \frac{\omega_5}{Q_5} \qquad (9)
\end{aligned}
$$

$$
T_6 = (T_1 + T_2 + T_3) \times (T_4 + T_5 + 1) \qquad (10)
$$

5. Control method of the analog filter

Two control signals are input into the Sallen-Key circuit and biquad circuit from the digital I/O port of a USB device. USB-FSIO30 (Komatsu 2011) of Km2net was used as the USB device. This USB device was prepared using a dynamic link library (DLL), which can be coded in Microsoft Visual Basic 2010 Express. Thus, a control signal can be easily generated by a computer program, and the cutoff frequency and center frequency of each filter can be controlled. The half-periods hp1 and hp2 of the pulse signal that are used as the two control signals are expressed by formula 11 and formula 12, respectively, and are set

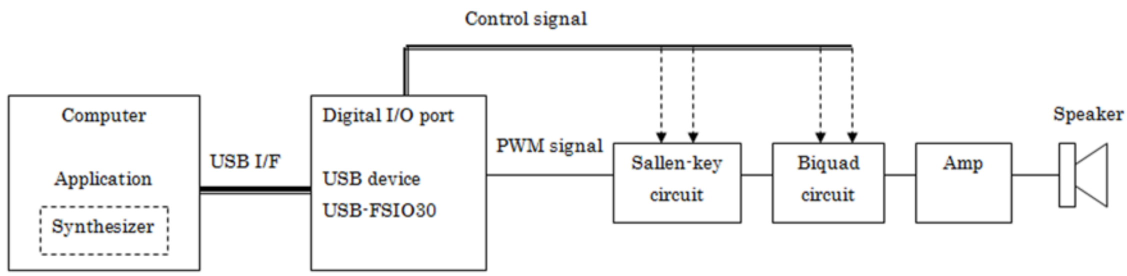


Figure 10. Example structure of a synthesizer and a filter

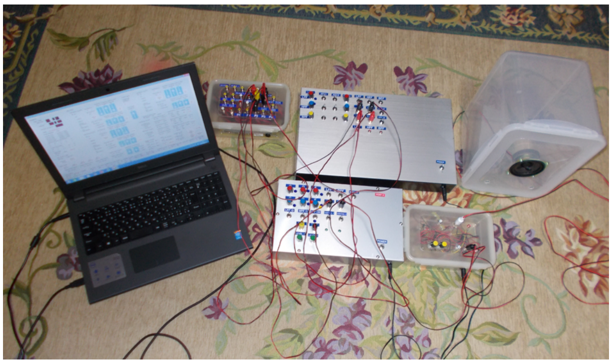


Figure 11. Example of the actual system

to vary between 1 ms to 2000 ms as maximums hp_max1 and hp_max2. The floating-point variable rnd represents a random number between 0 and 1. CInt represents an integer transfer function. These two control signals are asynchronous signals. The half-period in a computer program is not in strict agreement with the half-period of the signal output from a USB device.

$$\begin{aligned} hp1 &= CInt(hp\_max1 \times rnd) \\ &\quad (\; CInt(hp\_max1 \times rnd) \geq 1) \\ hp1 &= 1 \\ &\quad (\; CInt(hp\_max1 \times rnd) = 0) \end{aligned} \tag{11}$$

$$\begin{aligned} hp2 &= CInt(hp\_max2 \times rnd) \\ &\quad (\; CInt(hp\_max2 \times rnd) \geq 1) \\ hp2 &= 1 \\ &\quad (\; CInt(hp\_max2 \times rnd) = 0) \end{aligned} \tag{12}$$

In cases where the control signal was a conventional signal, it was set as a rectangular wave with a duty factor of nearly 50%; in cases where the control signal was a random signal, a pulse-width modulated (PWM) signal, whose duty factor changed in each period, was used. This program is part of a uniquely developed synthesizer (Tsuji 2015). Whenever performance information of the sequencer changes in this synthesizer, a probability function is used, and the type of control signal is determined. From Figure 6(b), the Sallen-Key circuit consists of three filters. Therefore, the logical product was carried out by TC74HC08AP (Toshiba Corp. 2014) of Toshiba Corp. using two control signals. Filter 3 was controlled by the calculated control signal.

6. Signal analysis of the analog filter

The signal analysis was carried out using the input signal and output signal of this filter. In this study, the input signal was an output signal of a synthesizer. The example structure of the synthesizer and this filter is shown in Figure 10, and an actual photograph is shown in Figure 11. A PWM signal is output from a synthesizer. The PWM signal inputs the modulating signal from the low-frequency oscillator into this filter. A PWM signal is input into the Sallen-Key circuit, and sensitivity and gain adjustments are made. Control of the cutoff frequency and center frequency is carried out by the control signal. Further, this output signal is input into the biquad circuit, and an adjustment of sensitivity is carried out. Control of the cutoff frequency and center frequency is carried out by the control signal, similar to that of the Sallen-Key circuit. By carrying out a cascade connection of the Sallen-Key circuit and the biquad circuit, a PWM signal is intricately modulated. The frequency analysis of an input signal and an output signal is shown in Figure 12. The input signal and the output signal were digitized by a sampling frequency of 44.1 kHz, and the linear-quantization was 16-bit. First, these audio data are used, and the system requests a power spectrum by Burg's method of an autoregression coefficient is the 48th order (Ishihara et al. 1988). One frame carries out a frequency analysis by the audio data of 100 ms (4410 samples). Next, the sample

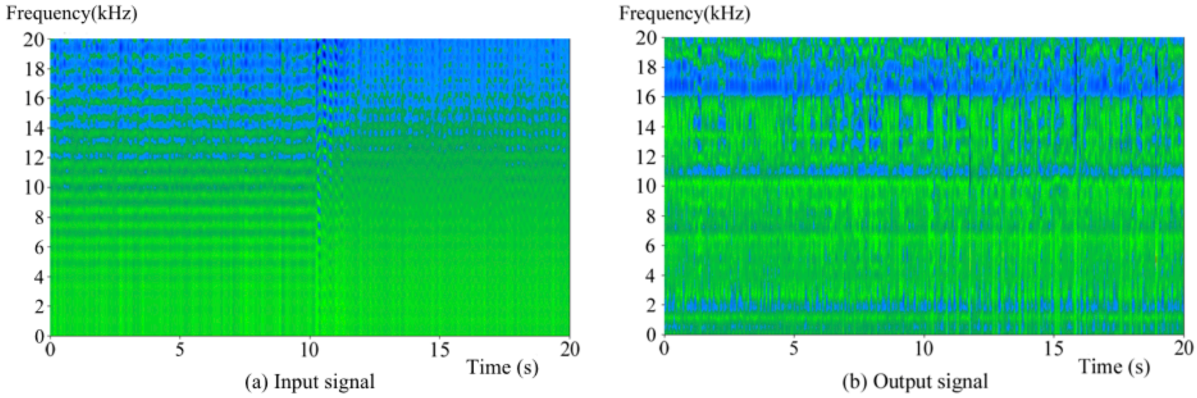


Figure 12. Frequency analysis of an input signal and an output signal

numbers of one frame are shifted by 20 ms (882 samples), and the new audio data of 20 ms is input. This operation is repeated and a frequency analysis is carried for 20 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 13. From formula 14, each required frequency component is divided by the sum of the frequency components of one frame. $Y_e(f)$ expresses a frequency component and f expresses the frequency. $Z_e(f)$ expresses the value that divides each frequency component by the sum of the frequency components of one frame. Thereby, the sum of the frequency components of one frame was set to 1, and the frequency component of the input signal and output signal could be compared. From Figure 12, the input signal decreases uniformly at high frequencies. However, the output signal produced contains the signal components from a low frequency to a high frequency, and the spectrum envelope changes from 12 kHz and 16 kHz.

$$x(n)\ (n = e \cdot s, e \cdot s + 1, e \cdot s + 2, ..., e \cdot s + (N - 1), \\ e = 0, 1, 2, ..., 995, s = 882, N = 4410) \quad (13)$$

$$Z_e(f) = \frac{Y_e(f)}{\sum_{f=0}^{20000} Y_e(f)}$$

$$(e = 0, 1, 2, ..., 995, f = 0, 1, 2, ..., 20000) \quad (14)$$

7. Application of filter 1 and filter 2

Even if the gain of filter 1 and filter 2 increases in value, each filter is converted so that the value of sensitivity can be made small. For this purpose, from formula 3 and formula 5, it turns out that the value of gain should be set to a negative value. The circuit that converts filter 1 and filter 2 is shown in Figure 13. From Figure 13(a), the converted filter 1 needs to connect capacitor $C1$ to the output of the first-order low-pass filter. In this study, the connection of the inverting amplifier and capacitor $C1$ is removed. From Figure 13(b), the converted filter 2 needs to connect resistance $R1$ to the output of the first-order low pass filter. In this study, a connection of the inverting amplifier and resistance R1 is removed. Therefore, it turns out that an alteration of filter 1 and filter 2 is simple. In addition, it can remove the inverting amplifier and capacitor $C2$. The frequency characteristics of the converted filters 1 and 2 are shown in Figure 14. When the frequency measurement is carried out, the analog photo coupler is replaced with a resistance of 1 kΩ. The voltage of an input signal is carried out at a partial pressure of $\frac{1}{5}$, and it is input into each filter. The gain of converted filter 1 is –5 and the gain of converted filter 2 is –6. From Figure 14, the frequency characteristics of each converted filter differ with filter 1 and filter 2, and the timbres of each converted filter differ with filter 1 and filter 2.

8. Conclusion

This study attempted to control a filter that implemented the cascade connection of a Sallen-Key circuit and a biquad circuit using a USB device. The Sallen-Key circuit is changed into an oscillation state, and the sensitivity is again adjusted in the biquad circuit. Furthermore, it seems to be an effective way of obtaining a special sound effect, because the cutoff frequency and the center frequency of each filter are controllable by a computer. This filter was developed as a part of a uniquely developed synthesizer. In the future, I would like to reexamine the special sound effect using an analog filter.

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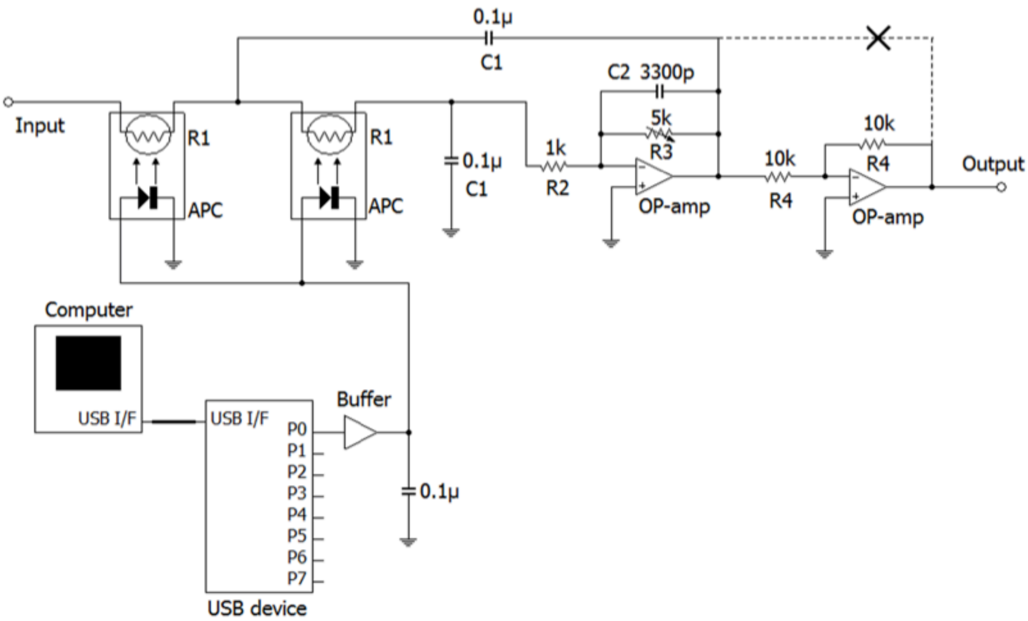
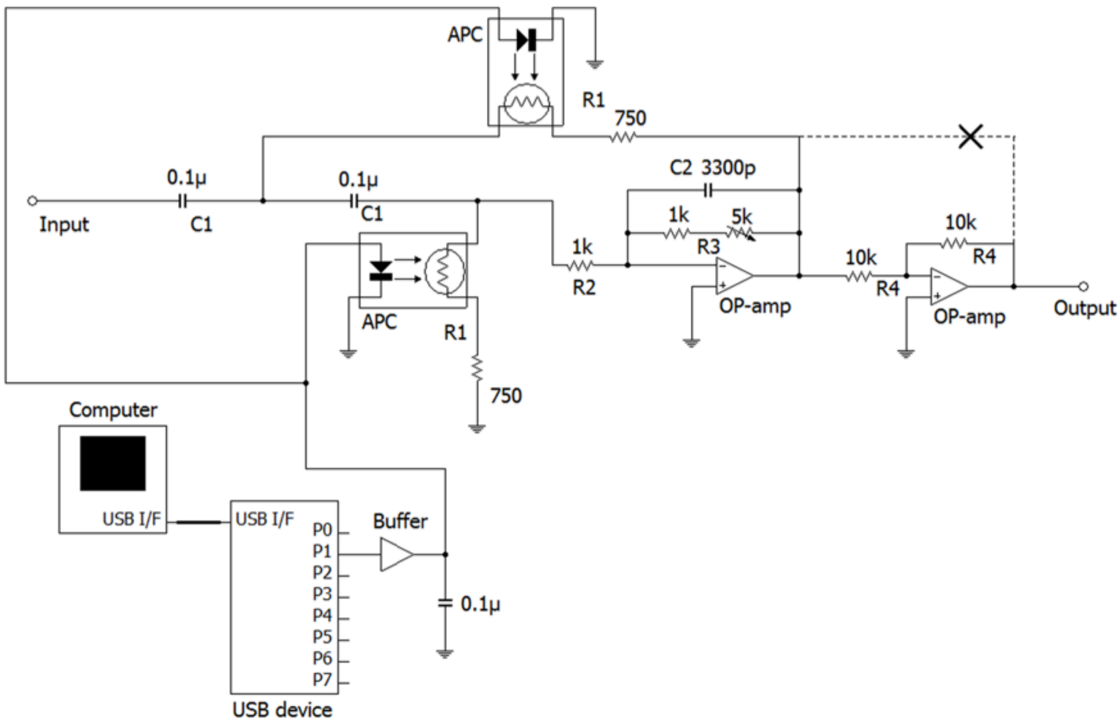


(a) Converted filter 1



(b) Converted filter 2

APC : Analog photo coupler

Figure 13. Circuit structure of the converted filter

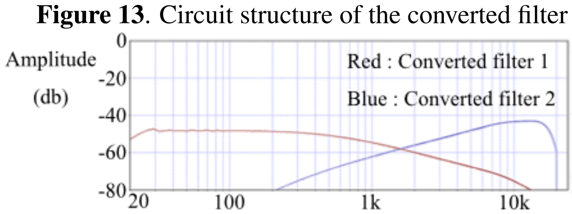


Figure 14. Frequency characteristics of the converted filter

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9. 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 development of the received adapter unit for FM multiplex broadcasts, research on estimation of sound-source localization, and other works. He retired in 1998. He is currently a regular member of the Acoustical Society of Japan.