



**Attractor Editing for Music Expression in Sophia Sound  
---Including Mapping Modulation Method in Cellular Flow ---**

**Mamoru Tanaka<sup>1</sup>, Masaki Bandai<sup>1</sup> and Yoshifumi Nishio<sup>2</sup>**  
**Sophia University<sup>1</sup>                      Tokushima University<sup>2</sup>**

The expression technology including AI with machine learning has been attractive in order to realize better timbre expression of the three elements of sound. A basic technique is **pitch shift processing**, if PCM technique recording sound of expensive live musical instruments is not used. The pitch shift processing is a process of shifting the frequency set of the periodic function  $F(x)$  to the frequency direction in accordance with the change of each pitch frequency  $f_n$  of the note sequence  $\{N_n\}$ . As a direct method, there is a frequency domain analysis in which a frequency set of the function  $F(x)$  is got by Fourier transform, the frequency set is shifted, and an inverse Fourier transform is performed to obtain a time domain function. There is also a time domain analysis in which the main period  $T$  of the function  $F(x)$  is changed so as to match the period corresponding to each pitch frequency  $f_n$  of the note sequence  $\{N_n\}$ . Also, the FM sound source technique uses a method in which the pitch frequency  $f_n$  is equivalent to the carrier frequency  $f$  for the modulated frequency  $g$  with factor  $m$  as  $\sin(2\pi ft + m * \sin(2\pi gt))$ .

In general, the function  $F(x)$  has a problem that the period  $T$  is not expressed explicitly. We introduce the **mapping modulation method** as a new time domain analysis method. The mapping modulation method is generally used as a nonlinear mapping function

$$y_n = F(G(t_n)).$$

Here, for example, let

$$\{t_n \mid n = 0, 1, 2, \dots, t_{max-1}\}$$

be a time sequence of one note, then, the nonlinear mapping method is expressed in a discrete form as

$$x_n = G(t_n) = \frac{rate}{2\pi f} \sin^{-1}(2\pi f t_n),$$

and

$$y_n = F(x_n) \text{ in } F(t_n) \geq 0, \quad \text{otherwise, } y_n = F(t_n) \text{ in } F(t_n) < 0,$$

in the region  $x_n \geq 0$ , or

$$y_n = F(-x_n) \text{ in } F(t_n) \leq 0, \text{ otherwise, } y_n = F(t_n) \text{ in } F(t_n) > 0,$$

in the region  $x_n < 0$ .

The algorithm includes a delay element and a modulation factor (relative intensity of the function  $F$  with respect to the function  $G$ ). Also the algorithm has a network structure including a Fourier series expansion. Nonlinear function  $F$  is an arbitrary function, especially a periodic function. The important thing is that the nonlinear function  $F$  can be generated by called **Cellular Flow** (SPICE-like SDP circuit simulator) which includes two-variable automatic differential expressions (calculation graphs as basic technology of Google tensor flow), RLC elements, negative resistance, gyrators, etc. The gyrator neuron is expressed by using activation function  $\mathbf{f}(\mathbf{x})$  as

$$\begin{pmatrix} i_1 \\ i_2 \end{pmatrix} = \mathbf{f} \left[ \begin{pmatrix} g_{11} & g_{12} \\ g_{21} & g_{22} \end{pmatrix} \begin{pmatrix} v_1 - v_2 \\ v_3 - v_4 \end{pmatrix} \right].$$

Each conductance  $g_{ij}$  for **forward propagation** can be determined in **back propagation** in the cellular flow learning process same as calculation graphs of the tensor flow. **Error propagation** between teacher and output signals ( $v_1 - v_2$ ) can be done by changing from voltage to current. The neuron element is included in the Jacobian matrix. The Function  $F$  is a waveform in the time domain generated by the circuit analysis which uses implicit numerical integration method. In order to reduce the number of times for analysis, it is important to edit the structure of the attractor which is composed of the left and right sound waveforms. The sequence of output values  $\{y_n\}$  is a time sequence which has frequency set shifted accordance with the change of each pitch frequency  $f_n$  of the note sequence  $\{N_n\}$  over the entire piano keyboard. In general, the MIDI sequence obtained by **automatic music transcription** is incomplete, and the changes in loudness for the expression are lost. In addition, the standard MIDI sound source (GM) has a problem that it is insufficient for music expression because of its low quality. Therefore, there is a need to convert MIDI sequences to Wav sequences to enrich the music expression. By using Sophia Sound, we have tried the mapping modulation method to the MIDI note sequence  $\{N_n\}$  for real Oscar Peterson's jazz music. As a result, a new musical instrument sound was obtained with a rich sense of rhythm, which has a mixture of piano, bass and drum sounds, (<https://www.etlab.jp>).

It will be expected that new instrument sounds with expressions will be generated using the resonance phenomenon of nonlinear elements and large-scale LC circuits.