

# DESC9115: DIGITAL AUDIO SYSTEMS

## Written Review 2

### Tempo-Synchronised Effects Controlled by a Beat Tracking Digital Audio System

Nick Frost SID: 308211448

#### The Problem:

A musician/DJ has a collection of desired effect units that will produce a modulated output sound. However, a common limitation is that the modulation of these effects is not synchronized with the tempo of their input. This problem may be encountered with any time-variant effect such as a wah-wah, phaser, tremolo, or a flanger. Often if these effects are applied with a random rate the outcome will be uncomplimentary to the rhythm. This will sound out of place and could result in the musician/DJ believing that the effect is inappropriate and therefore discarding the idea altogether. It may be advantageous to manipulate other parameters such as the depth of these effects, but often when the rate is varied without regard to tempo the result can be detrimental to the music.

#### The Solution:

In order for these effects to compliment the music, the modulation rate must be linked appropriately to the tempo of their input. When the music and the effect work together the result can significantly enhance the composition. This harmony of the input and the effect is achieved with a beat-tracker that processes the rhythm (based on patterns calculated from the input) and assigns a tempo. For example, if a beat-tracker analyses the tempo of a musical loop, the modulation of a wah filter can be synchronized so that a completely new element will be introduced to suit the groove. Synchronization means that the elements will work as one, therefore the result will sound appropriate and intentional.

The wah-wah filter is used as an example of how modulation can be synchronized with tempo. The wah effect uses a state variable filter that allows a designated 'band' of frequencies through, while attenuating all other frequencies. A band pass filter is simply a low pass filter combined with a high pass filter. The filter acts upon its inputs  $x(n)$  (where  $n$  is the sample number), to produce its outputs  $y(n)$ . Because the state variable filter can produce three outcomes they are named  $y_h(n)$ ,  $y_b(n)$ , and  $y_l(n)$ . These signify the high pass, band pass, and low pass outputs respectfully. The following diagram illustrates the signal flow:

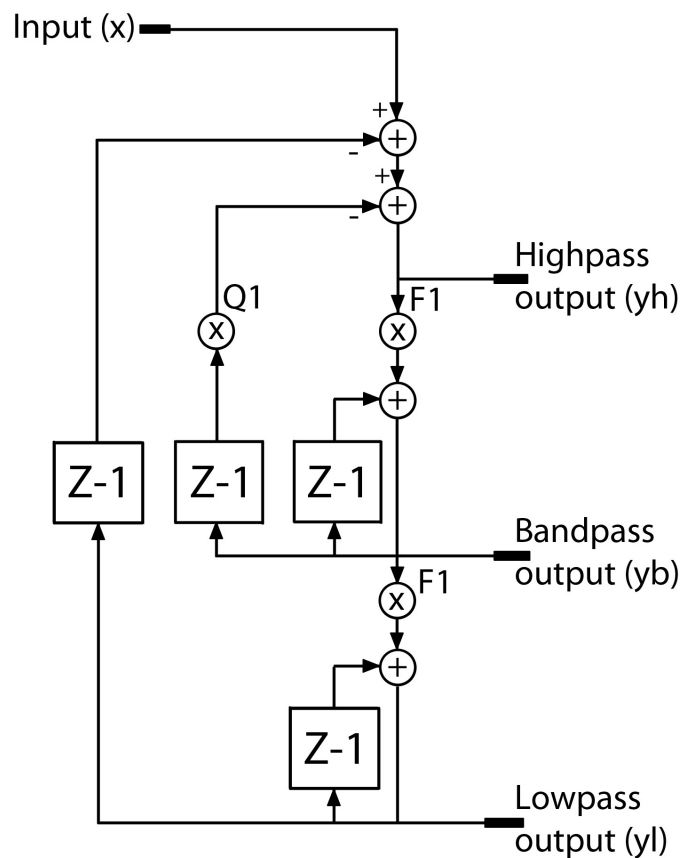


Figure 1: Signal flow of a state variable filter

The filter uses the difference equation<sup>1</sup>:

$$y_h(n) = x(n) - y_l(n-1) - Q1 * y_b(n-1)$$

$$y_b(n) = F1 * y_h(n) + y_b(n-1)$$

$$y_l(n) = F1 * y_b(n) + y_l(n-1)$$

The formula uses the tuning coefficients:

$$F1 = 2 * \sin((\pi * Fc(1)) / Fs) \text{ and } Q1 = 2 * C$$

<sup>1</sup> Zolzer, Udo, 2002, *DAFX: Digital Audio Effects*, Chapter 2, John Wiley & Sons

The band pass filter (yb) band is modulated over time, sweeping between an upper and lower cutoff frequency to create the “wah” sound which gives the effect its name. A low frequency oscillator will control the rate of modulation and the LFO itself will be controlled by the tempo calculated by the beat tracker.

### **How the Beat Tracker Calculates Tempo:**

The aim of a beat tracker is to develop a rhythm or tempo from an audio signal input. It is a complex process and in this case we assume that the tempo is regular and in 4/4 style. The system analyses patterns in the incoming signal, depending on the type of tracker these patterns could be in the frequency spectrum or in the onset strength envelope. The tracker performs calculations to determine the time between each beat and uses this information to predict the timing of future beats. I imported the following drum loop into MATLAB in order to determine its tempo based on the concept of dynamic programming proposed by Jean Laroche<sup>2</sup>.

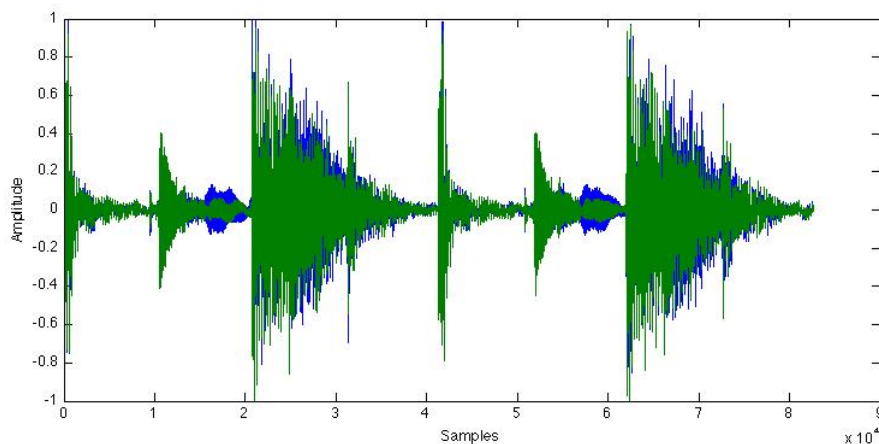


Figure 2: Input signal showing amplitude over time

By examining the peaks in amplitude of the waveform that has been created it is already possible to see patterns emerging. The largest rhythmic elements can be seen at roughly 0, 20000 and 60000 samples. Perhaps these are the kick drum and the evenly spaced peaks between them are other percussive elements such

---

<sup>2</sup> J. Laroche. “Efficient tempo and beat tracking in audio recordings”, *Journal of the Audio Engineering Society*, 51(4):226–233, Apr 2003.

as snares or hi hats. While it is possible to see repetition in the waveform by eye, the following equations from Ellis<sup>3</sup> act on the input to give numerical data.

$$C(\{t_i\}) = \sum_{i=1}^N O(t_i) + \alpha \sum_{i=2}^N F(t_i - t_{i-1}, \tau_p)$$

This formula examines onsets in the audio signal and determines whether these onsets have a rhythm. What must be remembered is that this formula is based on the assumption of a target tempo that provides a reference point for the patterns found in the input signal. Here  $\{t_i\}$  is the sequence of  $N$  beat instants found by the tracker.  $O(t_i)$  is an “onset strength envelope” derived from the audio,  $\alpha$  is a weighting to balance the importance of the two terms,  $F(\Delta t, \tau_p)$  is used to measure the regularity between two beats  $\Delta t$  and the ideal beat spacing  $\tau_p$  defined by the target tempo.

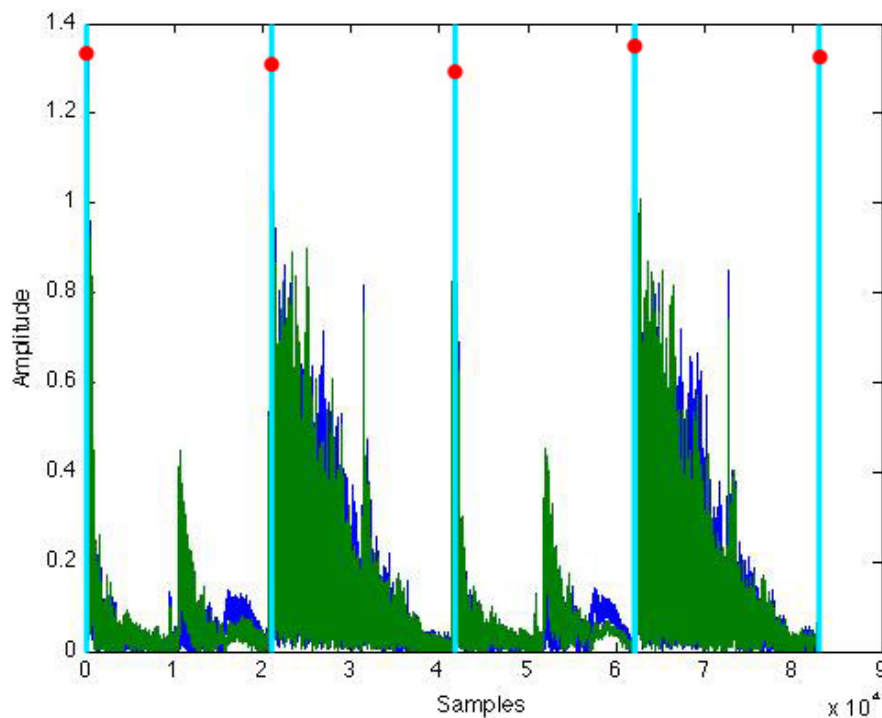


Figure 3: The onset strength envelope of the input audio. Red dots indicate the peak amplitude, Blue lines are estimated beats calculated by the system.

---

<sup>3</sup> D.P.W. Ellis, “Beat Tracking by Dynamic Programming,” *Journal of New Music Research*, Vol. 36(1), 51–60, 2007

A simple calculation based on the results from this graph can convert the answer into beats per minute (BPM). If we assume the peaks occur every 21000 samples (as the graph indicates) at a sampling rate of 44100 samples per second:

$$44100/21000= 2.1 \text{ beats per second}$$

$$2.1 \times 60 = 126 \text{ beats per minute.}$$

Once again, Ellis demonstrates how the final two equations will rate how effectively the sequences can be repeated in order to narrow down the possible tempo range.

$$C^*(t) = O(t) + \max_{\tau=0\dots t} \{\alpha F(t - \tau, \tau_p) + C^*(\tau)\}$$

Where  $C^*(t)$  is used to calculate the best score of all sequences that end at time  $t$ . It is understood that a  $t$  is ideally the closest envelope onset strength and it is calculated in reference to the best score of the preceding beat. i.e.

$$P^*(t) = \arg \max_{\tau=0\dots t} \{\alpha F(t - \tau, \tau_p) + C^*(\tau)\}$$

Finally,  $C^*$  &  $P^*$  get calculated for every sample beginning with sample zero. Next the largest value of  $C^*$  is determined and the systems works backwards via  $P^*$  to find each preceding beat tome until the system reaches the zero sample once again. This is how the tempo is realized when the beat tracker is implemented on a steady rhythm. In simple terms the beat tracker analyses the envelope onset strengths of its input, and delays the signal to find correlations in the peak transients. The pattern is then compared against a defined master tempo. The fact that this system does rely on a constant tempo can limit its application, however when used appropriately its accuracy and computational efficiency will rival alternative systems.

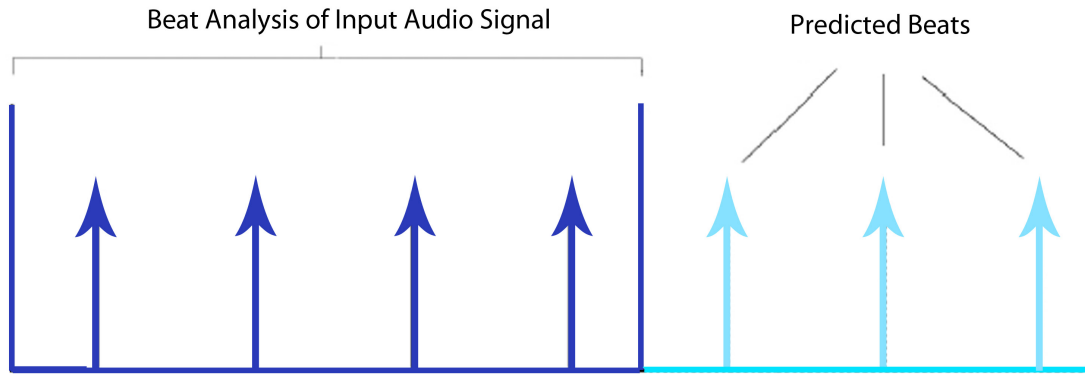


Figure 4: The beat tracker determines beat spacing and uses this information to predict future beats.

Once autocorrelation has identified any regular, periodic structure and a tempo has been established by the beat tracker, an LFO is used to synchronise the effect. We will examine how to implement an LFO with 1 or more cycles per beat. Because of the direct relationship between the number of cycles per beat  $\Omega$  and the frequency of the beats calculated by the tracker  $\tau$ , the following equation is used to determine the length of an LFO cycle<sup>4</sup>:

$$\mathcal{J} = \frac{\tau}{\Omega}$$

Now the LFO must be modulated by a phase value between 0 and 1. If each cycle of the LFO is represented by  $c[n]$ , then  $0 \leq c[n] < 1$ .  $c[n]$  will climb from 0 to 1 over a rate determined by the length of the LFO cycle. To ensure that  $c[n]$  only takes values between 0 and 1, the following fractional equation is used:

$$c[n] = \text{frac}(c[n-1] + \eta)$$

Where  $\eta = 1/\text{length of LFO cycle}$ .

So far the beat tracker has calculated a tempo, the LFO has synchronised a wave to the tempo, and now that LFO must be used to modulate the wah filter. When  $m\Omega[n]$  is the modulating waveform at a rate of  $\Omega$  cycles per beat, then to

---

<sup>4</sup> A.M Stark, M.D Plumbley and M.E.P Davies, "Audio effects for real-time Performance using Beat Tracking", *Audio Engineering Society Convention paper 7156*, 2007.

synchronise the cutoff frequency,  $f_c$ , of the wah we simply use the following equation from A. M Stark *et al.*

$$f_c = f_b + (m\Omega[n] \cdot f_r)$$

Where  $f_b$  represents the base cutoff frequency, and  $f_r$  represents the sweep range of the bandpass filter in Hz.

If all of these systems are correctly implemented, the output will be a tempo-synchronous wah filter. The signal flow of these components is illustrated in the signal flow diagram below.

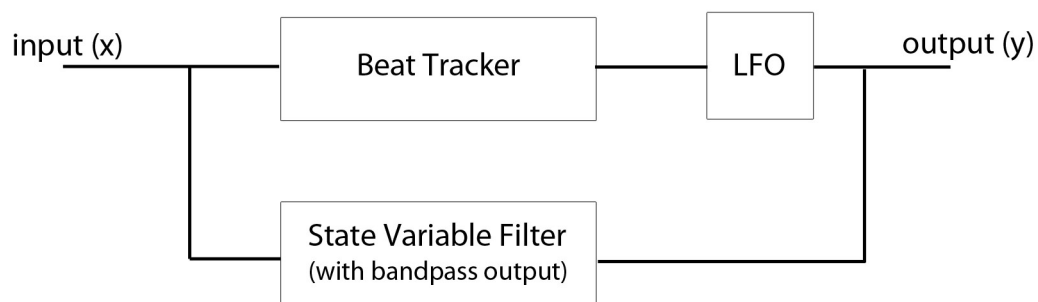


Figure 5: Signal flow diagram showing how the components act upon the input to produce an output.

## Conclusion:

Tempo-synchronized audio effects have diverse application in both live performances and musical composition. Once a tempo dependant LFO has been created, it can be used to modulate a variety of effects such as wah-wah, flanger, phaser and tremolo. The beat-tracker can similarly be used to define delays which may be useful to create a tempo synchronized echo effect. By combining these digital audio systems, we are able to make these effects more intuitive to the tempo of their input so that they can compliment the inherent rhythm in all music.

## REFERENCES:

N. P. W Frost, "Creating and implementing a Stereo 'Wonder-Wah' Filter", *Sydney University E-Scholarship Repository*, June, 2013.

M. E. P Davies and M. D. Plumbley, "Context Dependent Beat Tracking of Musical Audio", *IEEE Trans on Audio, Speech and Language Processing*, Vol 15, Issue 3, pp. 1009-1020, March, 2007.

D.P.W. Ellis, "Beat Tracking by Dynamic Programming," *Journal of New Music Research*, Vol. 36(1), 51–60, 2007.

E. D. Scheirer, "Tempo and beat analysis of acoustic musical signals", *Journal of the acoustical Society of America*, vol. 103(1), p.588:601, 1998.

C. Uhle and J. Herre, "Estimation of tempo, micro time and time signature from percussive music", in *Proceedings of DAFx*, 2003.

J. Laroche. "Efficient tempo and beat tracking in audio recordings", *Journal of the Audio Engineering Society*, 51(4):226–233, Apr 2003.

M. F. McKinney, D. Moelants, M. Davies, and A. Klapuri. "Evaluation of audio beat tracking and music tempo extraction algorithms", *Journal of New Music Research*, this issue, 2007.

S. Dixon. "Automatic extraction of tempo and beat from expressive performances". *Journal of New Music Research*, 30(1):39–58, 2001.

A.M Stark, M.D Plumbley and M.E.P Davies, "Audio effects for real-time Performance using Beat Tracking", *Audio Engineering Society Convention paper* 7156, 2007.