Compositor – the bottom-up approach

Ruslan Yusipov

Independent researcher,

Kuntsevskaya str. 8-1-37

121351 Moscow

Russia

ruslan.yusipov@gmail.com

Abstract

Techno music became popular among many spheres of our life and intelligence. Young people as well as those who founded the genre in 80s got involved in making sounds which characterize a machine like music. My involvement into the scene is to contribute for automatic generation of techno music based on the function derived from Frequency Modulation (FM) for rhythm generation and FM side bands as fundamentals for sound synthesis.

Simple FM side bands relate to each other not as the members of arithmetic progression, because of the Carson's bandwidth rule (Schottstaedt 2007). This rule states $k = I + 1$ harmonics (where k is the side band number and I is the modulation index), which is larger on one side band, than $d = I \cdot F_m$ deviation (Dodge and Jerse 1997) needed for a standard progression:

$$a_k - a_{k-1} = d$$

Where a is a progression member and d is a difference between members (Kramor 1994).

I used an abstraction that simple FM by one sinusoidal carrier and one sinusoidal modulator creates not only a bandwidth of sound but also an arithmetic progression of several useful properties. The main function derived from this analysis, which

run the algorithm, compares frequency modulation side bands to the central frequency as in mathematical operation of measuring. The function graphic is hyperbola.

Introduction

My intention is to design the unified approach which characterizes the event called techno music. For me to distinguish noise from music, one of the following criteria must meet: first of all, the noise is a random process (Dodge and Jerse 1997; Cipriani and Giri 2010) but music is determined by some criterion and talking about algorithmic music it must be a mathematical property of some sort, which drives the tones playback; second, noise is characterized by a broad spectrum and for human to distinguish a tone it needs to include a fundamental (Dodge and Jerse 1997); third, for techno music to be distinguished from a random process it needs to include musical property such as temporal coherence. My mathematical function not only generates a rhythmic structure but I use the same FM spectrum side bands as fundamentals for sound technics of non-linear distortion and physical modeling synthesis.

The whole project named Compositor realized using MaxMSP programming language is intended for automatic techno music generation and live performance.

The idea of the rhythmic structure of Compositor instrument came to me after studying drumming rudiments. The rule behind exponential distribution of sound events is the single ratamaque rudiment (Gadd 1994). The fast flam in the beginning of this rudiment (Figure 1) is compared to sweep of the start time in exponential distribution which characterized by hyperbola graphic.

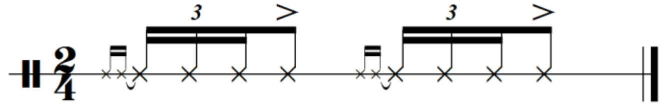

Figure 1. The Single ratamaque rudiment.

During the work on a project I was presented by an article of Nick Collins (see *Computer Music Journal* 36(3):8-23, 2012) in which he describes the top-down approach for autoacousmatic music. This article is an essay on the reverse model, which is as he stated the bottom-up approach to automatic algorithmic music generation.

FM is characterized by three parameters, such as central frequency (Fc), frequency of modulation and modulation index (Dodge and Jerse 1997). These three parameters characterize wave phenomenon with rarefaction and compression of sound wave in time. As stated earlier in simple FM with two sound sources such as sinusoidal oscillators, the following number of harmonics exists:

$$k = I + 1$$

This harmonics were treated by me as a separate frequency bands when filtering them with fourth order band-pass filter. In this model, I use cascade of two second order Butterworth filters. This slope selected by me to extract an exact harmonic when modulator-to-carrier ratio (b) is $2/1$. Together with low Fc from 37.9259 *Hz* to 73.8817 *Hz* and large modulation index such as $I = 9$ it constitutes that lower order harmonics going from Fc into the negative frequencies region change their value to positive *Hz* but change the direction of phase (Cipriani and Giri 2010). I need this to select direct pass-band as the Fc for a current pair of side bands, which coincide when negative frequencies goes over 0 *Hz*.

For the system to work with the fast Fourier transform (FFT) it needs to include the law that sound wave exists during a large period E.g. infinity (Dodge and Jerse 1997; Petelin and Petelin 2001). Using my mathematical function I suggest the mechanism to reveal separate harmonics in time and to distinguish them from spectral representation. I postulate the difference of time that a function based on, with the standard $4/4$ notation of music timing. Then I represent a function on a logarithmic spiral (Figure 2) with every complete cycle of it as a length N and which differentiates together with Fc. Including in my model Doppler Effect tells that lower *b* ratios correspond to a higher frequencies and the spiral cycles are wider while larger values of *b* correspond to a narrower spiral cycles and lower frequencies.

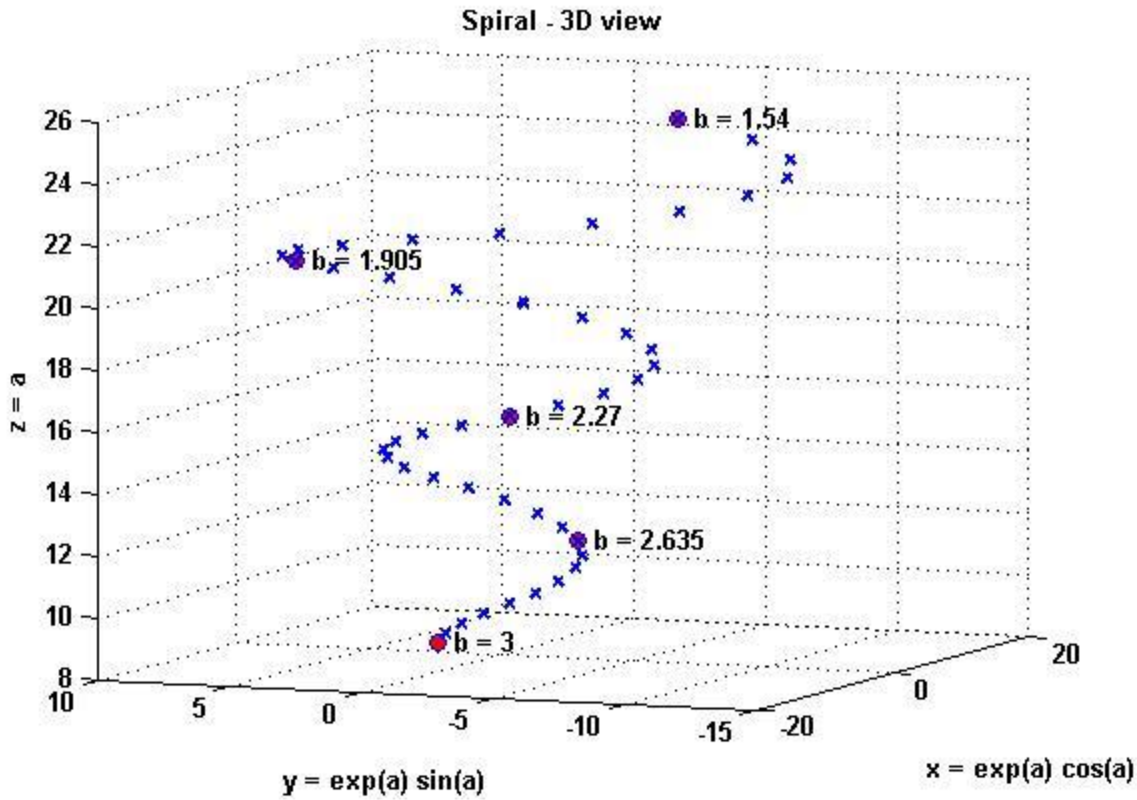

Figure 2. Spiral revolutions according to b ratio value.

To find a spiral values I need to represent them as four segments attaining to a step value of 0.365 between $b = 1.54$ and $b = 3$. This step length is equal to *b* multiplier values of 1.54, 1.905, 2.27, 2.635, 3, which are the points of a harmonics tuning change. To represent the spiral I need to count 8π value for multiplier 1.54 as the outer boundary and to find a starting point I need to divide 8π on 3 as the inner boundary. This will give me three spiral revolutions equal to 46 steps. Dividing on four segments will give me 12 steps for each of the three segments and 10 steps for the last segment. This way I will find out which points of the spiral represent each of the selected multiplier *b* values.

The diameter of complete revolution of spiral is 1,024 *Hz,* which is the N length of samples. Harmonic numbers coincide with FM side bands frequency values and I got a pair of values such as frequency and period $F(k)$. This property together with the pass band of a filter reveals a waveform representation of FM spectrum particle in time domain.

Body

The function exists for all side band values - negative and positive ones. Negative frequency values in effect change their phase and as stated by an FM rule: odd negative frequencies have a phase multiplied by -1 (Dodge and Jerse 1997). This led me to a set of three functions - one for all positive side bands and two for negative odd and even side bands.

Positive side bands:

$$F(k) = \frac{N}{1 + kb}$$

Where *N* – is a window length, *k* – is a harmonic number, and *b* – is the modulator-to-carrier ratio.

Negative odd side bands:

$$F(k) = \frac{N}{|1 - 2kb + b|}\ , \qquad \text{where } k = \frac{I + 1}{2}$$

Negative even side bands:

$$F(k) = \frac{N}{2} + \frac{N}{|1 - 2kb|}\ , \qquad \text{where } k = \frac{I + 1}{2}$$

The deviation is 1024 *Hz* but when negative frequencies fold over 0 *Hz* the highest harmonic is:

$$J_{10}(I) = F_c + 10 \cdot F_m, \qquad \text{when } I = 9$$

The Bessel function ($J_k(I)$) for this harmonic constitutes the highest possible shift on a graph. Therefore, I got a useful way to display Bessel graph motion in time (Figure 3) and Compositor visual system reveals it by using 3-D OpenGL technology (Figure 6).

The motion through harmonics and function values is achieved by the iterative mechanism (Cipriani and Giri 2010). First, values are generated using the three appropriate wavetables for one of the functions. Master clock generator drives all of this wavetables at once synchronizing a motion. The wavetable values of function fires the recursion and using modulo operator I split them on positive and negative odd and even tracks. Only odd modulation indexes are possible, because modulo operator creates two progressions when the argument is even – one for even numbers and another for odd numbers. For negative side bands I multiply the progression by 2 to achieve $n + 2$ steps. Negative even values of side bands coincide with the second function, which is situated in the second half of the window. Function graphics represented on a wavetable display will look like exponential distribution (Dodge and Jerse 1997), which characterized by a belt like structure (Figure 3).

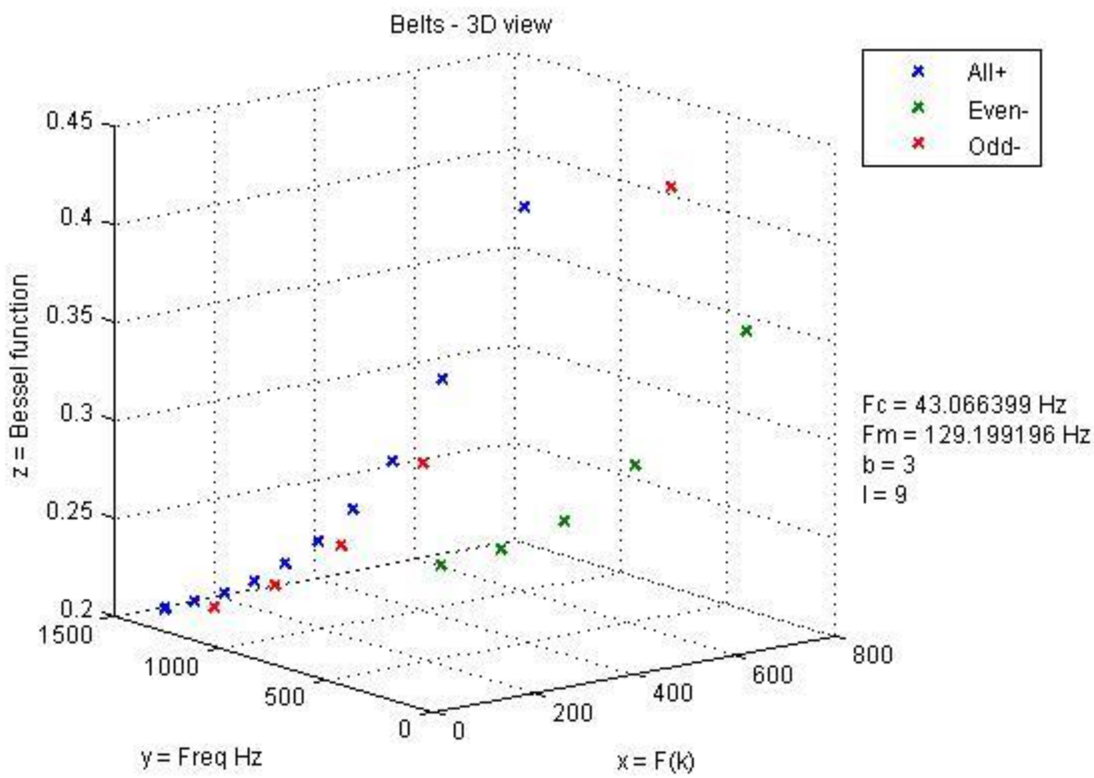

Figure 3. Functions plotted in three dimensions showing a belt like structure.

Center frequency of periodic signal in FFT is:

$$f_0 = \frac{f_s}{N}$$

FFT disturbs the spectrum on k channels, which are in zero to $\frac{N}{2}$ region. The $\frac{N}{2} \cdot f_0$ channel is on the Nyquist frequency, which characterizes the upper limit of sampled audio material (Dodge and Jerse 1997). Compositor can achieve FFT pitch synchronous analysis when $F_c = 43.066399\ Hz, F_m = 129.199196\ Hz, b = 3$. Using the values of the function I can take an FFT window of the length which is equal to the current *F(k)*. The window length, which function can predict for the current harmonic, is equal to the window length for a pitch synchronous analysis with a mantissa of five values of precision. I take in account that window length is an integer value and for it to be pure pitch synchronous analysis of non-harmonic periodic signal I just need to take an integral part of *F(k)*. I need to mention that this pitch synchronous analysis is possible on the base of each harmonic separated in time. This outcome will give me an opportunity to visualize each harmonic and its phase by viewing single FFT band for it. The function operates for sample rate 44.1-*kHz* and window length 1,024 samples. There are sample rate values for all other window lengths, which coincide together. For example, for window length 2,048 samples sample rate will double and will be equal to 88,200 *Hz*. I consider 44.1-*kHz* as $(2 \cdot 3 \cdot 5 \cdot 7)^2$, which are the first four prime numbers, as a row of decimal representation such as:

$$2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 3 \cdot 10^0 \cdot 3 \cdot 10^0 \cdot 5 \cdot 10^0 \cdot 5 \cdot 10^0 \cdot 7 \cdot 10^0 \cdot 7 \cdot 10^0 = 44100$$

I use the same decimal representation for 1,024 samples:

$$2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 \cdot 2 \cdot 10^0 = 1024$$

I see that the first number is base eight (Carruthers 1877) and the second number is base ten. Comparing two systems I got an equality of base eight system with base ten system.

I use the three possible values to show the period length: msec, bpm and angular velocity (ω). For this to convert properly I have chosen several formulas:

From bpm to ω:

$$\omega = \frac{\pi \cdot \theta}{120}, \qquad \text{where } \theta - \text{is beats per minute value}$$

From msec to ω:

$$\omega = \frac{2000\pi}{T}, \qquad \text{where } T - \text{period length in seconds}$$

From ω to bpm:

$$\theta = \frac{240 \cdot \omega}{2\pi}$$

The exponential function coincides with standard time notation on base eight in radians, when $b = 3$.

For positive side bands:

Fifth harmonic on $\frac{\pi}{8}$, first harmonic on $\frac{\pi}{2}$.

For negative odd side bands:

Third harmonic coincide on $\frac{\pi}{4}$, first harmonic on π.

Only negative even frequencies do not coincide with time base eight.

Together with pitch synchronicity for current window, which is equal to 1,024 samples, deviation is 1,162.79 *Hz*. Starting from $F_c = 43.066399\ Hz$ means that the highest harmonic in the specter, when *I* = 9, will be:

$$J_9(I) = F_c + I \cdot F_m$$

It equals to 1,205.86 *Hz*, when Fm = 129.199196 *Hz.*

This is slightly different from the Carson's bandwidth rule, which state $I + 1$ harmonics for the effective spectrum (Schottstaedt 2007; Dodge and Jerse 1997). In fact, this predicts continuous growth of the spiral with cycles that are not closing at the point of 1,162.79 *Hz* deviation (Figure 2). Instead, it goes higher on additional interval equal for pitch synchronous analysis to $129.199196\ Hz$. So, where it comes to the end? The answer predicted by a multiplier value, which states the first cycle of the spiral at $b = 1.54$ and the last cycle at $b = 3$. These values selected for musical reasons for Fc, which equal to a sub frequency body of kick drum sound, to be in a range from 37.9259 Hz to 73.8817 Hz. This equals to a center frequency values of the modern styles of electronic dance music, where the kick drum exists as a fundamental for the whole track.

The rhythmic pattern in my instrument Compositor achieves the temporal coherence (Dodge and Jerse 1997) and it makes this system unique. In music theory, an event is called music, when it has a sequence or a melody (Dubyanskaya 2002). Sequence is a repeating cycle, which creates a melodic event, we recognize as logical by transmitting it from the source to our eardrum membrane and analyzing it in our brain (Dodge and Jerse 1997). This sequential structure is a prominent feature of modern styles of electronic music such as techno (Hawtin 2005). This style is primarily occupied with the creation of kinetic event through an introduction of sequence and sounds, which create temporal coherence. I recognize a harmonic movement in Compositor by studying their downward progression. The pure harmonic is encased in Helmholtz envelope with phases of attack, sustain, and decay (Dodge and Jerse 1997). I use this type of envelope to select the harmonic for pure pitch synchronous FFT analysis (Dodge and Jerse 1997). Every harmonic exists on a time interval, which is equal to Darboux integral (Figure 4). This integral characterizes the step function (Cipriani and Giri 2010).

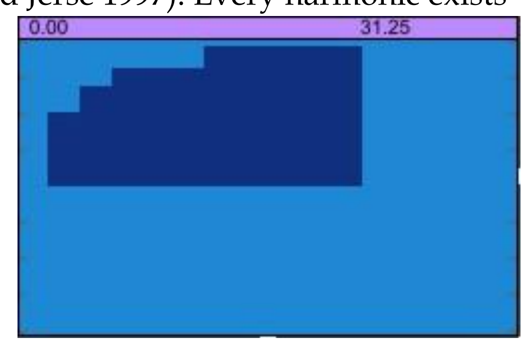

Figure 4. Darboux integral.

Harmonics are not only presented in an enhanced way, they are separated into two streams – upper and lower one, each with its own timbre. I use two systems for enhancing the sound of pure harmonics. The first system is non-linear destruction by wave shaping and it is applied to the higher row of harmonics. Through the number of transfer functions, counted by me using the

Chebyshev polynomials, I enhance the pure harmonic to the level it constitutes the timbre characterized as brass or horn like. Another chain, which is lower harmonics, encounters physical modeling through the waveguide resonator. It is somewhat special, because it does not include a feedback chain and creates two types of delay lines interconnected with each other. The polynomials create the row of harmonics based on the pure tone extracted from FM timbre. This harmonics selected by me to create a progression of special sort. It is even and odd or mixed numbers together for it to create the timbres of different instruments. Together with an envelope on the parameter of distortion index, I can trace the progression of harmonic structure on lower beat rates. The carefully selected timbres and sequential structure tend to induce a meditative trance state. On lower tempos, it can result to an effect of wanting to sleep.

For a demonstration of granular synthesis technology, my instrument includes the granular synthesis module, which coincides exponential function to the grid base eight by approaching the time interval based on my own derived integral value. The underlying formulas of Compositor grain module stated below.

The derived function for base eight rounding for positive side bands:

$$F(k)_i = \frac{N}{q} \cdot int\left(\frac{q}{1+kb} + 0.5\right)$$

Where q – is quantization grid.

The derived function for base eight rounding for negative odd side bands:

$$F(k)_i = \frac{N}{q} \cdot int\left(\frac{q}{|1-2kb+b|} + 0.5\right), \qquad \text{where } k = \frac{I+1}{2}$$

The derived function for base eight rounding for negative even side bands:

$$F(k)_i = \frac{N}{2} + \frac{N}{q} \cdot int\left(\frac{q}{|1-2kb|} + 0.5\right), \qquad \text{where } k = \frac{I+1}{2}$$

Estimated time for a run of a grain for positive side bands:

$$\Delta d = \int_{F(k)_i}^{F(k)} \left|\frac{N}{q} \cdot int\left(\frac{q}{1+kb} + 0.5\right) - \frac{N}{1+kb}\right| dx$$

Estimated time for a run of a grain for negative odd side bands:

$$\Delta d = \int_{F(k)_i}^{F(k)} \left| \frac{N}{q} \cdot int\left(\frac{q}{|1-2kb+b|} + 0.5\right) - \frac{N}{|1-2kb+b|} \right| dx , \qquad \text{where } k = \frac{I+1}{2}$$

Estimated time for a run of a grain for negative even side bands:

$$\Delta d = \int_{F(k)_i}^{F(k)} \left| \frac{N}{q} \cdot int\left(\frac{q}{|1-2kb|} + 0.5\right) - \frac{N}{|1-2kb|} \right| dx , \qquad \text{where } k = \frac{I+1}{2}$$

I take an absolute value of time, because it can be either negative or positive. For MaxMSP play~ module, which I selected for this task, to work properly I need to set the time as a positive value. By using the appropriate function value and function rounded on base eight, grains played back forward or backward, depending on whether they appear earlier or later on the quantization grid.

The base eight rounded function can also be used for rhythm description and can be revealed in musical notation such as in Figure 5.

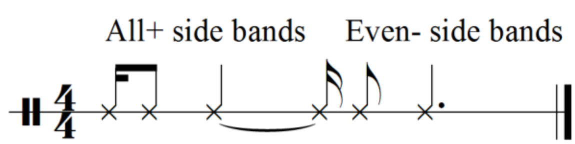

Figure 5. Base eight rounded function rhythm for all positive and negative even side bands when b = 3.

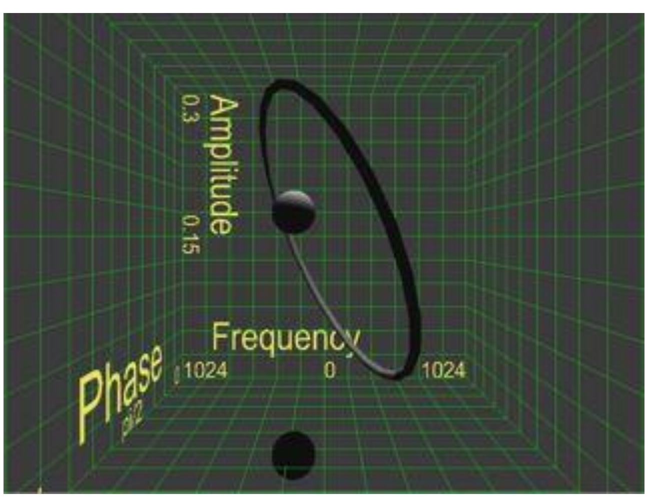

Figure 6. OpenGL representation in Compositor visual system.

The three rhythm section instruments of modern styles of dance music (including jazz and funk genres) such as kick drum, snare and hat (Garibaldi 1992) constitute three possible outcomes for a Compositor OpenGL visual system (Figure 6). They are played back in a sequential order using three sinusoidal

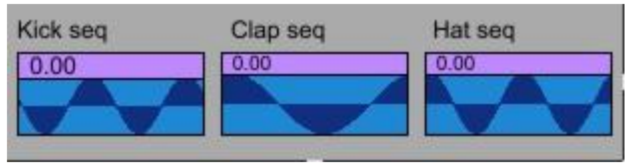

Figure 7. Sinusoids triggering the kick, hat and snare instruments.

oscillators zero crossings to trigger the play back system (Figure 7). Using the OpenGL torus object and phases of two sinusoids, which run snare and hat modules, together with a master clock generator varying from zero to 2π, I set three Euler angles. The snare drum triggered by zero crossings on $\frac{\pi}{2}$ and $\frac{3\pi}{2}$ sets an x angle. The hat triggered by zero crossings on $\frac{\pi}{4}$ and $\frac{5\pi}{4}$ during one cycle of sinusoid and $\frac{\pi}{4}$, $\frac{3\pi}{4}$, $\frac{5\pi}{4}$, $\frac{7\pi}{4}$ during two cycles initiated by "x2" button sets z angle. The master generator running from zero to 2π sets y angle. This way I got three values of Euler angles and to display motion of the torus I need to convert them to quaternion, the four vector, which consists of the real, scalar part, and x, y, z as imaginary part (Watt 2000). The kick drum appears on zero and π, when one cycle of sinusoid completes in the window length, and on zero, $\frac{\pi}{2}, \pi, \frac{3\pi}{2}$, when two cycles of sinusoid complete in the window length. This influences the appearance of z angle in a way described in the Table 1.

	0.0625 – 0.125	0.25	0.3125	0.125
Kick x2 On	x	x		
Kick x2 Off			x	x
Hat x2 On		x	x	
Hat x2 Off	x			x

Table 1. Interrelation of kick and hat phases

The table constitutes the initial state of the first zero crossing triggering the hat instrument and varies together with the phase of sinusoid triggering the kick drum. The numbers are given in normalized phase values.

The initial phase of the kick drum varies from 0.39 to 0.45. It happens because the onset time (Bello et al. 2005) of the function does not begin from the start of the window and function compresses and expands depending on the multiplier *b* value. The movements created by this transformation are explained in the Table 2.

	Deviation from left to right	Around the vertical axis	Strong deviation from left to right
Kick x2 On		x	x
Kick x2 Off	x		x
Hat x2 On	x	x	
Hat x2 Off			x

Table 2. Rotation types

The kick instrument made of the same FM timbre with an amplitude envelope and envelope on modulation index triggered by sinusoid zero crossings. The fast sweep at the beginning of modulation index envelope from upper value to zero achieves the spectrum with characteristic click at the beginning and low frequency body (Dodge and Jerse 1997).

The auto mode of Compositor is what considered by me as the unification between Probability theory and deterministic approach. While function rhythm and derived harmonic rows are the deterministic process, other modules parameters such as waveguide resonator and wave shaping are selected by the either uniform or normal (Gaussian) distribution. Multiplier *b* values selected by an interesting algorithm, which uses Beta distribution (Dodge and Jerse 1997) with values of $a = 0.1$ and $b = 0.9$. It selects one of the two scenarios: continue playback without change or select new multiplier value. The algorithm checks double triggering of the new multiplier value selection and states the rule that value from 0.9 side of the curve cannot be selected twice in 15 seconds period. The value for multiplier selected by an exponential distribution with $\lambda = b$ is recursively quantized. The quantization is based on the function for eights rounding of positive and negative even sidebands. The window length for the function equals $N = 1.46$, which is $0.365 \cdot 4$ preceded by 1.54 as the starting point, from which multipliers are counted. I get several preset values for b and they constitute the most interesting patterns of the function.

The mixing system of compositor consists of channels for outputs of wave shaping and waveguide for three branches of function together with three instruments of rhythm section and granular engine. Automatic panning, auto level and auto send to a Schroeder reverberator effect (Dodge and Jerse 1997) are controlled by the normalized frequency values mapped to a proper range. The auto mixing strategy is to create a bell-like spectral envelope of Compositor output.

Conclusion

As you find reading the article, I was primarily interested in searching for an algorithm, which produces an exact rhythm of function-based music applied to the techno music style. The artificial intelligence that produces such patterns still needs more variations to include but if we consider permutations number together with different harmonic rows derived from function, it will give us a very useful results, which can be included in a number of albums of algorithmic and techno music.

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