Real-time network operating system Compositor

About the author

Ruslan M. Yusipov (1984 - present), sound engineer (diploma specialist)
Russian Federation, 121351, Moscow, Kuntsevskaya st. 8-1-37
ruslanyusipov@icloud.com

Frequency modulation is designed to improve the purity of the transmitted signal in U.S. and North American telephone networks. The quality of the transmitted sound signal in the telephone line is the main interest of the author, as his specialty is television and Radio sound engineering. The objectives of studying the prototype of the Compositor NRTOS, which are given below, relate mainly to wired DSL (Digital Subscriber Line) communication. My task was to provide a preset system and the line of the Compositor NRTOS multiplexer. Over time, the project has grown into a full-fledged radio simulation studio. That is, it is a virtual radio station with multi-channel playlists, quality of service, simulated Ethernet protocols, created to study multi-channel routing on radio relay Ethernet stations.

Introduction

The real-time network operating system Compositor is based on the extension of parameter attributes of the RAD96 kernel and includes the following virtual network functions (VNF):

- L1-L3 L6-L7 vScanner: RIPv1, RIPv2
- L1-L3 L6-L7 vAggregator: OSPF, OSPFv3
- L1-L3 L6-L7 vRouter: BGP, RIPng, EIGRP

These functions form a wireless data transmission environment where they can be connected to the AVP interface to work with an internal kernel extension database consisting of 10171 routing tables. Work with the host operating system on which the Compositor NRTOS runs is carried out through a local database, allowing you to organize a connection environment for these extensions as network devices (NIC, network interface controller) and establish a local connection to them. Based on this, Compositor NRTOS allows you to make sets of configurations of virtual operating systems with different devices and save them in presets for later continuation of work with these environments.

The main tasks of Compositor NRTOS are:

- Equality of audio frequency modulation and radio frequency modulation;
- Proof of Carson's rule;
- The effect of using beatings for signal modulation;
- Study of the problem of irregular signal sampling rate;
- Application of frequency modulation on low-voltage power supply networks (e.g. telephony);

1.Thesis

The periodic time determined by the AC frequency is the current moment of time or an antiderivative hyperbolic function. This system is based on periods of rotation of Euler angles, multiples of 1(2):1:2. Compositor NRTOS number system uses a 64-bit representation, the base of a logarithmic function is power of two.

An original function, which is the basic function of Compositor NRTOS:

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

The graph of the function is hyperbola.

Compositor NRTOS is built on the basis of the law of frequency modulation (FM) and uses it to transmit data on an on-air network. FM is characterized by three parameters, such as central frequency, modulation frequency and modulation index. These three parameters describe a wave phenomenon with compression and wave expansion in the temporal representation. With simple frequency modulation by two sources of sinusoids, there is a number of frequencies *k* calculated by the formula:

$$k = I + 1$$

Where, *k* – number of side-bands, *I* – modulation index.

To separate harmonics when multiplexing FM channels in the upper and lower side-bands, Compositor NRTOS uses a cascade of two Butterworth filters of the 2nd order.

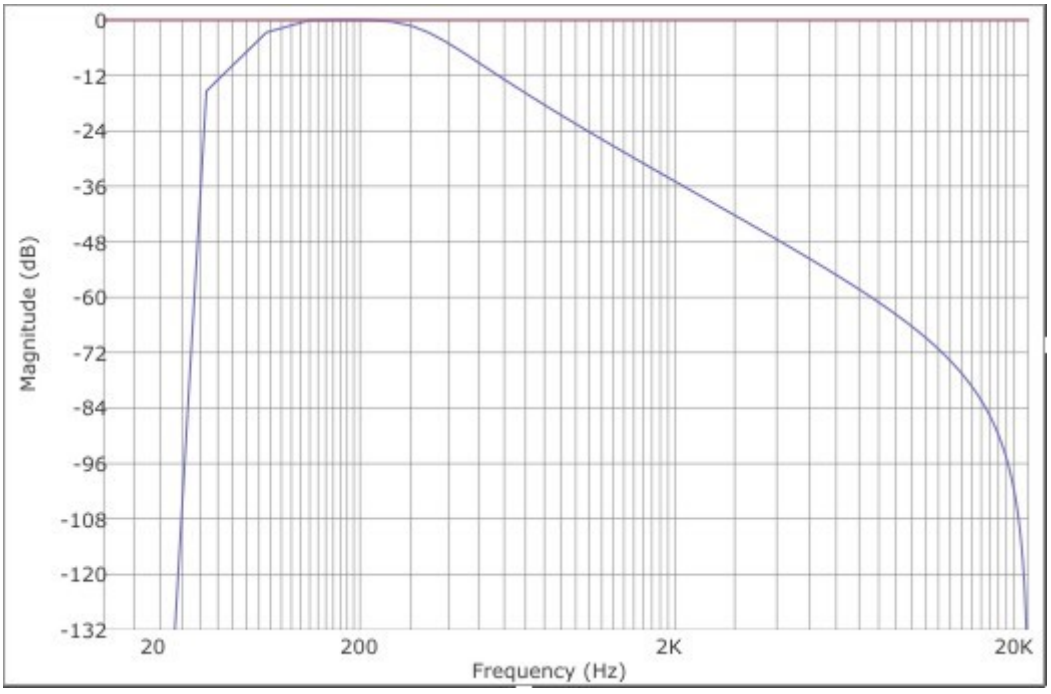

Fig. 1 - Butterworth 2nd order bypass filter

Thus, the bandpass filter characteristic becomes 12 dB per octave (total 24 dB) and a filter pass-band is equal to $\frac{3}{4}$ of the modulation frequency. The filter was created using bilinear transformation and is recursive. Mathematically, it can be represented as follows:

$$y[n] = a0*x[n] + a1*x[n-1] + a2*x[n-2] - b1*y[n-1] - b2*y[n-2]$$

In FM, the upper side-band (USB) and the lower side-band (LSB) are formed. The amplitudes of frequencies in the side-bands are displayed by the Bessel function $J_k(I)$ and are the Bessel functions of the *1st* order. The number of negative side-bands is equal to the number of positive side-bands. Negative side-bands are divided into even and odd. Odd side-bands begin with a negative phase, and even side-bands begin with a positive phase.

2.Main part

The segment of the virtual protractor from 0 to 2π is four planes of the Cartesian coordinate system. Then each of the frequency modulation harmonics can be displayed on the protractor, comparing it with the center frequency, and present it on a spiral passing along the surface of the truncated cone with a diameter of each spiral revolution equal to 1024 Hz. Frequency values for different ratios of modulation frequency to center frequency $\frac{F_m}{F_c}$, which are determined by a single parameter of the divisor *b*, correspond to different rotations of the spiral, which is explained by the expansion and narrowing of the values of the desired function. Then, taking the window of synchronous analysis of the non-harmonic spectrum (SANS) as a deviation, and presenting it in Hz, we get that each harmonica on the spectrum is lined up in a series equal to positive frequencies:

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

For negative odd frequencies:

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

For negative even frequencies:

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

N – SANS window

b – divisor ratio of modulation frequency to center frequency

Figure 2-4 shows graphs of the function for the divisor $b = 3$:

Fig. 2 - Function for positive side-bands

Fig. 3 - Function for negative even side-bands

Fig. 4 - Function for negative odd side-bands

The deviation is 1024 Hz, which corresponds to 1024 samples of the SANS window. It follows from the graph in Figure 3 that the amplitudes of negative bands are multiplied by -1. A positive spectrum when a low center frequency is used, which stays from the base (0 Hz) less than the modulation frequency is possible only when taking the negative frequency module. To do this, in Compositor NRTOS at z=0.5, the center frequency varies from 37.9259 Hz to 73.8817 Hz while maintaining the deviation unchanged and equal to 1024 Hz. Temporal dependences at such a center frequency and deviation correspond to the maximum purity of the perceived signal.

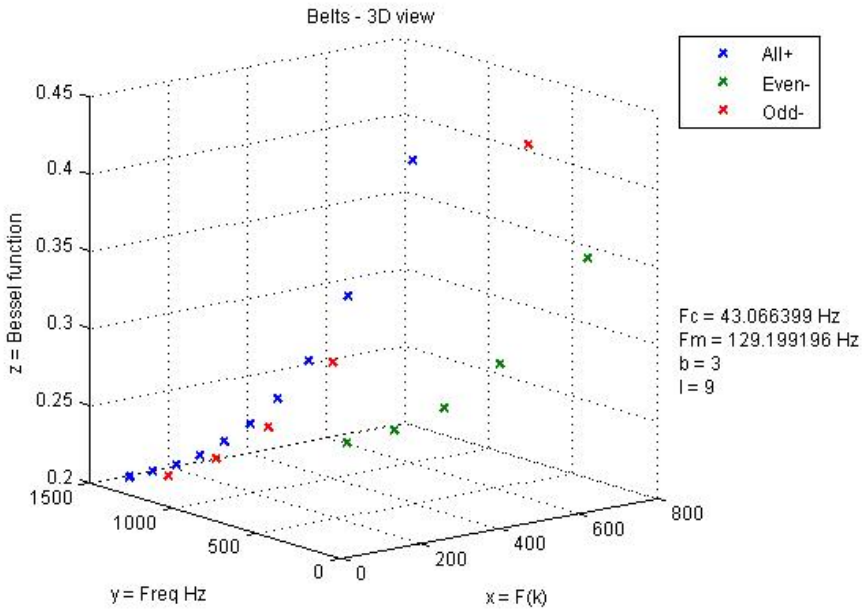

Fig. 5 - Branches of the function in 3-dimensional space

Deviation in frequency modulation is determined by the formula:

$$d = I * F_m$$

Since the frequencies turn around zero, the deviation is less than 1024 Hz for negative side-bands. It follows that the SANS window is equal to 1024 samples.

The center frequency of the periodic signal during the Fast Fourier Transform (FFT) is determined by the formula:

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

FFT divides the spectrum into *k* channels, which are between 0 to $\frac{N}{2}$. $\frac{N}{2} * f_0$ channel sets the Nyquist frequency, which characterizes the upper sampling limit. My method is not based on static reference to the center frequency, as in the phase vocoder, because I use a variable center frequency, and the modulation frequency does not change, so the parameter *b* cannot be a multiplier, but is a modulation frequency divisor, which helps to find the desired center frequency and allows you to determine the communication lines using buffer collisions. This is confirmed by the following:

I assume that 44.1 kHz as a row $\left(2 \bullet 3 \bullet 5 \bullet 7\right)^2$, which are the first four prime numbers as a decimal progression:

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

I use the same representative model for 1024 samples:

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

Compositor NRTOS allows you to perform a SANS in the value of the divisor $b = 2$, but for this the signal sampling frequency must be 58254.2336 Hz.

The center frequency of the upper band falls on $J(I)_{10} = \left(1024 - F_c\right) + F_m$. At a certain value of the divisor *b*, beatings of two neighboring frequencies are formed, which is characterized by amplitude modulation (AM).

The modulation frequency remains unchanged and is equal to 113.778 Hz at $I = 9$, only the ratio of the modulation frequency to the central frequency $\frac{F_m}{F_c}$, namely the divisor, changes.

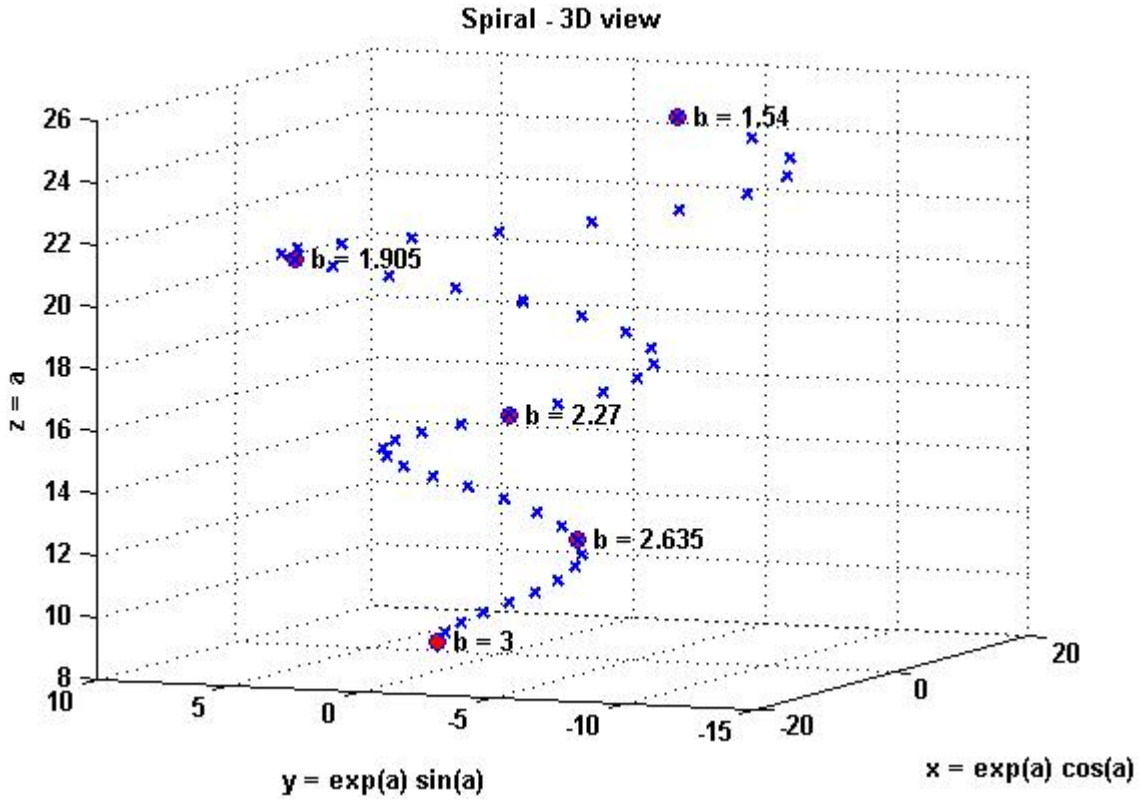

Fig. 6 - Dependence of the divisor on spiral rotation

The entire metric system can be represented as a virtual protractor, where the inner circle corresponds to the radians, and the outer one has frequency gradations and values of the divisor. Then, in order to match the values of the divisor and frequency bands, it is necessary to build the entire metric system in a spiral, where the wide coil corresponds to the values of frequencies and phases at the divisor $b = 1.54$, and the narrowest coil of the spiral corresponds to the value of phases and frequencies at $b = 4$. Accordingly, the function varies with the change of the divisor. Frequency bands *k*, which are according to the formula $F_c \pm kF_m$, corresponds to its own phase value, are expressed in the time segments of the SANS window (radians). For ease of programming, Compositor NRTOS uses digital signal values (samples). Each frequency filtered from the frequency modulation spectrum is characterized by a pass-band frequency, which is set as $\frac{3}{4}$ from the modulation frequency. This value is not selected randomly and corresponds to the bandwidth $\frac{F_m}{2} * 1.5$. This bandwidth is selected based on the divisor $b = 2$. In this case, the negative frequencies wrapped around zero coincide with the positive ones. Sufficient bandwidth is equal to $\frac{F_m}{2}$, but I use an extended spacing and multiply the bandwidth by another 1.5 to compensate for the loss of -3 dB at the filter cut-off frequency. The bandwidth for z=0.5 is 85.3333 Hz and selects the harmonic for which the filter is currently configured by the formula $F_c \pm kF_m$, that is, upper or lower side-band.

Thus, the signal chain of the Compositor NRTOS is as follows: FM arising from the modulation of one sinusoidal oscillator by another is divided into a number of frequency bands, according to the function and formula for determining frequency modulation side-bands. These frequency bands

are reproduced at a speed set by one of three ways: period time in milliseconds, network number in decimal value and angular velocity (ω). All values can be converted, and formulas are used for this:

From the decimal representation to ω: $\omega = \frac{\pi * \theta}{120}$, where θ – network number.

From ms to ω: $\omega = \frac{2000\pi}{T}$

From ω to the decimal representation: $\theta = \frac{240 * \omega}{2\pi}$

Each of the frequency bands holds a certain amount of time equal to the step function, where the increase in values occurs step by step.

This step consists of a window function that contains several cycles of the upper or lower side-band. This allows you to use SANS to determine the desired value of the center frequency of the transmission channel.

The next in the signal path is the module of signal separators into two streams. For the upper zones of the frequency range, nonlinear transformation by wave deformation of the signal is used, and for the lower frequency bands of the spectrum, physical modeling of the waveguide is used, which divides the reproducible frequencies into optical ports and electric ports with PoE support. This allows the reconstitution of the signal spectrum. The optical port parameters are selected according to the nonlinear propagation of the medium according to the rule $\frac{1}{f}$.

Fig. 7 - Phases of IRQ interrupter flags

According to Chebyshev polynomials, Compositor NRTOS calculate transmission functions for various radio spectra with even and odd harmonics. The master generator, whose period is equal to the period of the SANS window, synchronizes wave tables that reproduce the values of the three functions presented. Three sinusoidal signals at the zero-crossing intersection are the IRQ (Interrupter Request) flag sequences of the interrupter. The zero-crossing of the first sine wave can occur both at 0 and π at one period for a window cycle and on $0, \frac{\pi}{2}, \pi, \frac{3\pi}{2}$ at two periods for a window cycle. For the second sine wave whose flag values fall on $\frac{\pi}{2}$ and $\frac{3\pi}{2}$, the phase value is added to $x = 90°$. For the third sine wave, the values of the flag fall on $\frac{\pi}{4}, \frac{3\pi}{4}, \frac{5\pi}{4}, \frac{7\pi}{4}$. Phases in the Compositor NRTOS are recorded in normalized values from 0 to 1, which corresponds to the following table:

Phase		
Degrees	Radians	Normalized values
0°	0	0
45°	π/4	0.125
90°	π/2	0.25
135°	3π/4	0.375
180°	π	0.5
225°	5π/4	0.625
270°	3π/2	0.75
315°	7π/4	0.875
360°	2π	1

The signal of three sine waves in values from 0 to 2π represents Euler's angles. To rotate the flows of spherical space in the Compositor NRTOS, these angles (x, y and z) are transformed into a quaternion, which consists of a real (scalar) part and x, y, z as an imaginary part. Compositor NRTOS displays this process graphically using a ring and flow points, which indicate the position of the server in the distance-vector system.

When changing the divisor b, the values of the function expand and narrow, thereby combining the antiderivative function and the standard mesh expressed in base 8. This approach guarantees the coincidence of certain phases of the frequency bands when divisor $b = 3$ with the grid on base 8 in the following values:

Positive bands - 5th frequency on $\frac{\pi}{8}$, 1st frequency on $\frac{\pi}{2}$.

Negative odd bands - 3rd frequency on $\frac{\pi}{4}$, 1st frequency on π.

And only negative even side-bands do not coincide with the grid on base 8.

The next aspect is the statistical property of the system. If you take for abstraction that the signal is continuous and there is arbitrarily infinite time, then the SANS is applicable. Since the whole row is a sequence, the window is repeated recursively and can theoretically be reproduced indefinitely.

The time segment of the window is divided into segments by step integral. Granular synthesis is carried out by a derivative of a function that combines granules with a rounding mesh of a function based on base 8. N samples are used to indicate the phase, and N Hz is used to indicate deviation equal to the maximum bandwidth of the tone frequency channel. Signal components are played sequentially, one after the other. The values of the antiderivative function are rounded to a mesh on base 8 using the derivative of the function.

Derivative for positive frequencies:

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

Where q is the quantization mesh.

Derivative for negative odd frequencies:

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

Derivative for negative even frequencies:

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

To maintain the general characteristic of the signal spectrum, according to the nonlinearity of the function corresponding to the hyperbola graph, the spectrum of the resulting signal in the Compositor NRTOS is approximated by the characteristic of the two-pole band-pass filter of the 2nd order.

3.Conclusion

Compositor NRTOS considers a number of frequencies both as an arithmetic progression and as a spectrum band that complies with Carson's rule. This rule postulates a sufficient number of harmonics in the FM spectrum equal to $I+1$. According to Chowning, the spectrum of the FM audio signal is equal to $d = I*F_m$ (where d is deviation, I is the modulation index, and Fm is the modulation frequency), which gives the spectrum less than according to Carson's rule. If we consider FM audio signal from the point of view of the formula above, then for audio frequencies at a deviation of 1024 Hz, you get a spectrum with a upper side-band maximum frequency equal to 1058.13 Hz, which at a modulation frequency of 102.4 Hz corresponds to the desired spectrum of 1024 Hz (if we count from $F_c = 34.133 \text{ Hz}$), and when considered as a frequency modulation of radio-signal the upper side-band maximum frequency equal to 1175.7 Hz. I insist on Carson's rule application to FM audio signal, which postulates a spectrum of 1175.7 Hz in this case for $I+1$ harmonics of the signal, while the *k*th harmonic is not a progression, but an additional frequency at frequency modulation. According to Chowning's paper, frequency modulation of the audio signal produces a spectrum of 1058.13 Hz, not 1175.7 Hz. So in the spectrum of frequency modulation of the audio signal for I=10 in his opinion there are 10 harmonics. Compositor NRTOS proves that the FM audio signal is also a radio signal transmitting in the ULF-VLF range. This concludes that there are $I+1$ harmonics in the FM audio signal, and it complies with Carson's rule.