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DIGITAL METHOD OF SSB MODULATION

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Abstract: This paper is dedicated to modeling of the digital method of single-sideband (SSB) modulation. Digital method of SSB modulation provided in this paper is based on Weaver`s method of SSB modulation. The design and calculations of digital blocks are detailed. Such a model allows obtaining the output SSB signal in time and frequency domains at given input signals and system parameters. This could be useful for design and optimization purposes.

In the case of implementation of SSB modulation with the use of analog circuits, it requires mixers, filters constructed with the use of inductors, capacitors, ceramic filters, resonators etc. In the case of the digital implementation of SSB modulation, considered in this paper, no analog filters or multipliers are required. Any filtering and multiplication is performed in digital form. Only analog to digital (ADC) and digital to analog (DAC) converters are required here. The ADC is required for digitizing of input analog signal that is usually in human voice frequency range. The high-speed DAC is used for backward converting of a digital sequence to the analog form of final SSB signal. Such approach allows performing all transformations without any other analog circuits and does not require adjustment in the case of mass-production.

Key words: digital processing, digital SSB modulation, Weaver`s SSB modulation, SSB modulation.

1. Introduction

The digital signal synthesis has become very popular these days. Such popularity is related to the development and availability of the high performance digital devices such as programmable systems-on-chip (PSoC), Field-Programmable Gate Array (FPGA), digital signal processing (DSP) microprocessors. Such technical progress tends to the substitution of typical analog circuits with digital circuits. Such substitution has its advantages and drawbacks. The key advantages of digital signal synthesis are noise resistance, fast processing, simple adjustment, flexibility, low cost. The key disadvantages are the quantization error, less accuracy due to limited data resolution, wide frequency bandwidth of parasitic emission due to the steep edges of digital pulses.

This paper is dedicated to the digital method of modeling of single-sideband (SSB) modulation. There are two best known basic methods of SSB modulation: phasing method and filtering method [1, 2]. In addition, some combined filtering and phasing methods exist. In the case of implementation of SSB modulation by analog means, it requires mixers and filters containing inductors, capacitors, ceramic filters, resonators etc. In the case of filtering method, the steep edge filters are required. Such filters usually consist of the chain of simple filters and their tuning is complicated. The phasing method requires high quality analog multipliers which should have stable characteristics in the wide dynamic, frequency and temperature range that is not often feasible. All mentioned cases of imperfectness of filters and multipliers are caused by the parasitic side-band emission and the distortion of the SSB signal.

In the case of digital implementation of SSB modulation considered in this paper, no analog filters or multipliers are required. All filtering and multiplication is performed in a digital form. The only components required are analog to digital and digital to analog converters (ADC and DAC respectively). The ADC is used for digitizing an input analog signal that is usually located in the frequency range of human voice (typically 300–3400 Hz). The high-speed DAC is required for back converting of the digital sequence to the analog form of a final SSB signal. Such approach allows performing all transformations without any other analog circuits and does not require adjusting in the case of mass production. The drawback of this approach is the limited central frequency of the SSB signal due to limited speed of modern DACs. Nowadays, high-speed DACs make hundreds of millions of conversions per second which corresponds to SSB signals with band center frequencies of dozens or hundreds megahertz [3].

2. Weaver`s Method of SSB Modulation

The method of SSB modulation, considered in this paper, is based on Weaver`s method which is described in [1]. This method is also called the third method of SSB modulation which is the combination of phasing and filtering methods. The block diagram of Weaver`s SSB modulator is shown in Fig. 1.

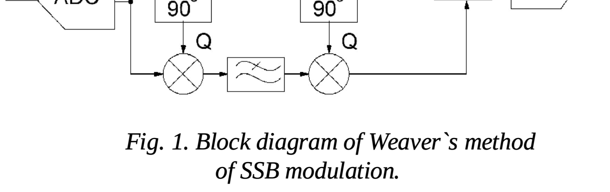

Fig. 1. Block diagram of Weaver`s method of SSB modulation.

The sequence of signal spectrum transformations is shown in Fig. 2. Fig. 2, a shows real and imaginary parts of the spectrum of the input signal.

According to Fig. 1, the input signal is split into two components I and Q (I – in phase, Q – quadrature-phase). Firstly, input signal is multiplied by reference signals at frequency $f_B/2$ located approximately in the middle of the voice spectrum. It causes spectrum shifting to the left by $f_B/2$ frequency value as shown in Fig. 2, b. A resulting signal passes through the low-pass filter (LPF) suppressing the sideband spectrum. Thereafter, the signal is multiplied by I and Q components of a reference signal whose frequency is equal to the central frequency of the SSB signal. Finally signals from I and Q channels are summed and the output SSB signal is obtained as shown in Fig. 2, d.

The suppression of the parasitic sideband and the signal quality mainly depend on cut-off frequency of the LPF, and the distortion of the SSB signal is smaller if an amplitude-frequency characteristic within the pass-band is flatter.

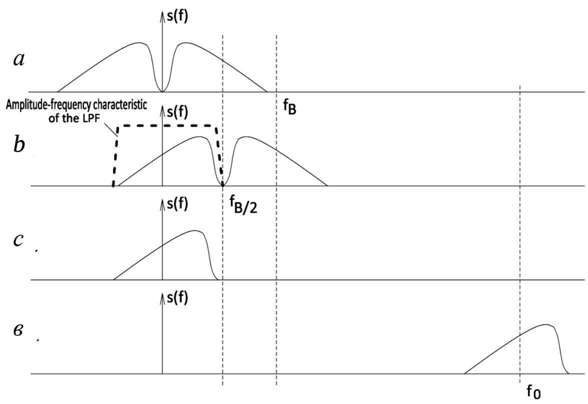

Fig. 2. Spectrum transformations in Weaver`s SSB modulator.

3. Digital Implementation of Weaver's Modulator

The block diagram of the digital design of Weaver`s SSB modulator is shown in Fig. 3.

An analog signal X_a in the voice frequency range is sampled with a frequency f_d and converted to the digital form using the ADC. The sequence of digitized data X is split into two paths: I channel (at the top of Fig. 3) and Q channel (at the bottom of Fig. 3). I and Q channel structures are similar except for the phase of the reference sequences used in each of these channels. There are I and Q sequences used corresponding to the sine and cosine samples respectively. These sequences are the following:

$$Isequence = [0,1,0,-1,0,1,0,-1,0...], \quad (1a)$$

$$Qsequence = [1,0,-1,0,1,0,-1,0,1...]. \quad (1b)$$

There are 4 samples per period in sequences (1a) and (1b).

The stream X of input data is multiplied with each of I and Q sequences and, as a result, the *I1_Y* and *Q1_Y* sequences are obtained. Thereafter, these sequences pass through low-pass filters (LPF) and the *I1_LPF* and *Q1_LPF* sequences are obtained. Then they pass to interpolators. The interpolators are used just to raise the discretization frequency to f_h value. The interpolated sequences *I1_INT* and *Q1_INT* are multiplied by the same quadrature sequences (1a) and (1b) but at frequency f_h. Finally, resulting sequences from I and Q channels are summed and converted to the analog signal using the DAC.

A block diagram in Fig. 3 is divided into the left and right part according to the discretization frequency. Signals in the left part are processed in f_d frequency domain and signals in the right part in f_h frequency domain. Below, important relations used for implementation and modeling of the digital form of SSB modulator in Fig. 3 are given.

A delay unit Δt_b should provide 90-degree phase shift between I1 and Q1 sequences. As equations 1 shows, the full period of Q or I sequences consists of 4 samples. It means that discretization frequency f_d is 4 times higher than Q or I reference frequency. This ratio can be increased, but, in general, f_d frequency should always be related with a period of I or Q sequences as follows:

$$f_d = 4 \cdot k \cdot f_q, \quad (2)$$

where f_q is a frequency of quadrature reference sequences (I and Q sequences); k is a positive integer number.

Let us note that equation (1) is specified for the case of $k = 1$. If k is higher than 1, sequences (1) should be interpolated k times.

An interpolation ratio M is calculated using the following equation:

$$M = \frac{f_h}{f_d}, \quad (3)$$

where f_h is the discretization frequency of processing in the right part of the diagram in Fig. 3. In other words, it also is the sampling frequency of the DAC. Obviously, M should be a positive integer number.

The delay unit Δt_c provides 90-degree phase shift between I and Q sequences.

The central frequency f_c of the SSB signal is calculated as follows:

$$f_c = \frac{f_h}{4 \cdot k}, \quad (4)$$

where k is a positive integer number. Let us note, that if the number of samples in one period of I or Q sequence (1) is multiplied by 4, these sequences should be interpolated k times. For practical usage, the value k = 1 is quite sufficient for qualitative voice communication.

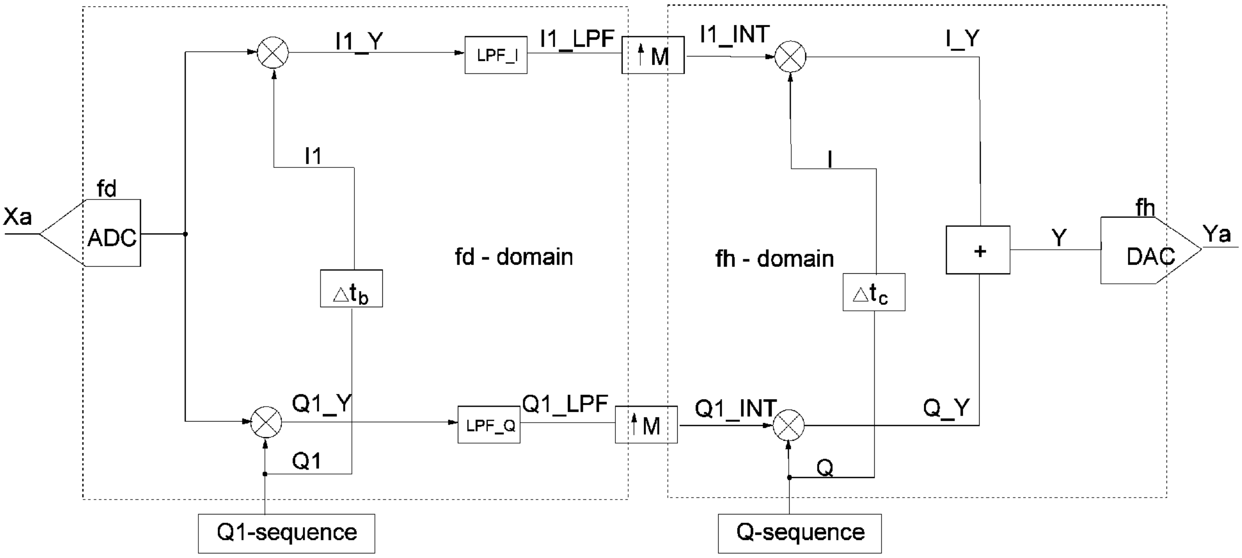

Fig. 3. Block diagram of Weaver`s method of SSB modulation.

The approximate values of the interpolation ratio M, based on a priori known central SSB frequency, can be obtained by substituting (4) to (3):

$$M' = \frac{4 \cdot k \cdot f_c}{f_{d\,\min}}, \qquad (6)$$

where $f_{d\,\min}$ is a minimal ADC discretization frequency considering the maximal frequency of the input signal and the Nyquist theorem. Equation (5) allows estimating the interpolation ratio M' based on the known central frequency f_c of the SSB signal and minimal ADC sampling frequency $f_{d\,\min}$. If M calculated by (5) is a fractional number, the M should be rounded down and exact ADC sampling frequency should be recalculated by the following equation:

$$f_d = \frac{4 \cdot k \cdot f_c}{M}. \qquad (6)$$

For example, there is a requirement to generate the SSB signal with the central frequency f_c=4.9 MHz. The spectrum of the input signal is limited at the frequency 3 kHz. Considering the Nyquist theorem, the minimal ADC sampling frequency f_{dmin} should be at least 6 kHz.

Using (5) and considering k equal to 1, the approximate interpolation ration M' can be calculated as follows:

$$M' = \frac{4 \cdot k \cdot f_c}{f_{d\,\min}} = \frac{4 \cdot 1 \cdot 4900\text{kHz}}{6\text{kHz}} = 3266.666.$$

Since M' is a fractional number, it should be rounded down, and ADC discretization frequency should be adjusted according to (6):

$$f_d = \frac{4 \cdot k \cdot f_c}{M} = \frac{4 \cdot 1 \cdot 4900kHz}{3266} = 6.00122 \text{ kHz}.$$

According to (2) and considering reference sequence as in (1), the frequency of quadrature references can be calculated as follows:

$$f_q = \frac{f_d}{4 \cdot k} = \frac{6.00122MHz}{4 \cdot 1} = 1.50031\,\text{kHz}.$$

The f_q is approximately in the middle of the voice spectrum as it is required.

The LPF can be implemented in the form of a digital filter with infinite impulse response (IIR) or the finite impulse response (FIR). In general case the transfer function of the LPF can be detrmined like those of classical digital filters:

$$H(z) = \frac{b_0 \cdot z^{-1} + b_1 \cdot z^{-2} + \ldots + b_n z^{-n}}{1 + a_1 \cdot z^{-1} + a_2 \cdot z^{-2} + \ldots + b_m \cdot z^{-m}}. \qquad (7)$$

4. Model of digital SSB Modulator

The model of the digital SSB modulator shown in Fig. 3 is created using MATLAB GNU Octave. Such model allows:

1) calculating numerical values of the digital sequences in any unit of the diagram in Fig. 3.

2) obtaining spectrums of the internal and output signals.

3) setting different coefficients of the LPF transfer function and rearranging it.

4) applying different input signals, for example, single-tone signal, biharmonic signal, noise and evaluating output.

Such a model is useful for designung digital SSB modulators in different digital devices (FPGA, PSoC). It allows calculating system parameters at a modeling stage and avoiding time consumption for adjusting a real system. Additionally, it allows comparing characteristics and signals obtained experimentally with simulated ones in the proposed model.

Below, there is an example and simulation results obtained using the model with the following parameters: $f_d = 6$ kHz; $f_q = 1{,}5$ kHz; $M = 64$; $f_c = 96$ kHz. The LPF is the 18-th order digital filter with finite impulse response with cutoff frequency equal to 1,5 kHz and 20 dB signal suppression at 1,8 kHz.

The voltage applied to the input is represented as the superposition of sine signals and expressed as shown below:

$$Vin(t) = A_0 + \sum_{k=1}^{N} A_k \cdot \sin\left(2 \cdot p \cdot f_k + j_k\right). \quad (8)$$

In the proposed example, the biharmonic signal with the parameters $A_1 = A_2 = 2\,\mathrm{V}$; $f_1 = 400$ Hz; $f_2 = 1$ kHz; $j_1 = j_2 = 0$ rad is applied.

Fig. 4 shows the internal signal *I1_Y* in frequency domain obtained after multiplying by the *I1* sequence (see Fig. 3). It is shown that the 400 Hz and 1 kHz harmonics are mirrored relative to the frequency of 1.5 kHz which is the reference *I1* frequency. So, as a result of multiplication, frequency components at 500 Hz, 1100 Hz, 1900 Hz and 2500 Hz are obtained. In Fig. 4, the dashed curve represents the LPF amplitude-frequency characteristic and shows the suppression of the side-band frequencies at 1900 Hz and 2500 Hz.

Fig. 5 shows the output signal *Ya* in the time domain. There are visible amplitude beats in Fig. 5 caused by mixed frequency components equal to 400 and 1000 Hz. As the result, the beat frequency is equal to 600 Hz.

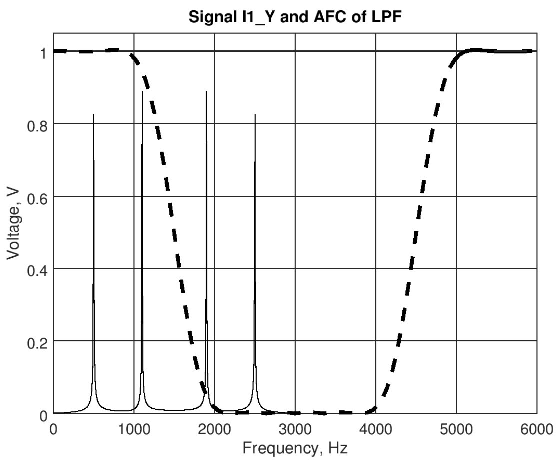

Fig. 4. Internal signal I1_Y in the frequency domain and amplitude-frequency characteristic of the LPF (dashed curve).

Fig. 6 shows the output signal *Ya* in the frequency domain. There are two noticeable frequency components at about 96 kHz which are the input 400 Hz and 1 kHz biharmonic components shifted to the high central SSB frequency f_c.

Additionally, there are a lot of parasitic spectral components, but with significantly smaller amplitudes.

If there is no input signal, the output signal is also absent which is expected in the case of SSB modulation.

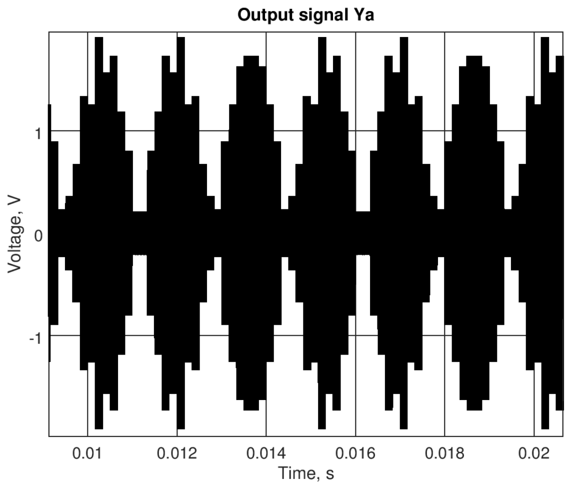

Fig. 5. Output signal Ya in time domain.

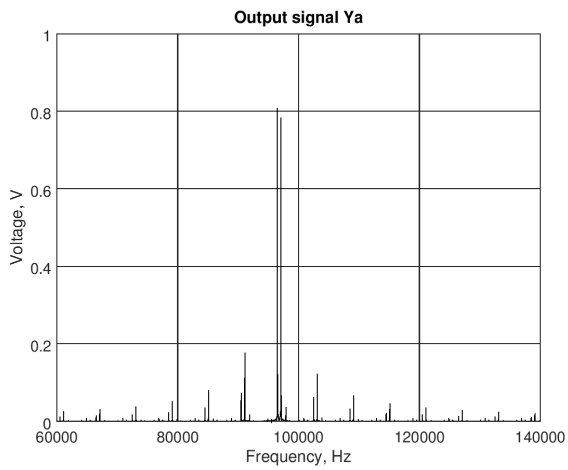

Fig. 6. Output signal Ya in the frequency domain.

5. Implementation of SSB modulator using PSoC

The digital SSB modulator considered in this paper was created using a programmable system on a chip (PSoC) manufactured by Cypress. Due to limited ADC/DAC speed and only 48 MHz clock frequency of the microprocessor CPU, the maximal central frequency of the SSB signal was about 100 kHz. The SSB signal obtained with the use of PSoC4 was passed to the SSB receiver and human voice was clearly understandable at the receiver end.

6. Conclusion

The architecture of the digital method of SSB modulation considered in this paper has the following advantages:

1) It allows performing SSB modulation without analog filters and only with digital devices (except for ADC/DAC).

2) In practice, the considered architecture allows generating SSB signals with frequencies up to several dozen Megahertz.

3) Digital method of SSB modulation is not sensitive to the environment impacts, because all processing is performed in a digital form.

4) No special adjustment required during mass production.

The considered architecture has the following drawbacks and peculiarities:

1) Any value of central SSB frequency can be obtained by changing the interpolation coefficient M and discretization frequency of the ADC. It slightly increases the complexity of the system in the case when output frequency should be varied in the wide range. Such a problem does not occur if output frequency is fixed.

2) Sideband emission occurs due to the quantization.

3) The DAC sampling frequency should be at least 4 times higher than the required SSB central frequency.

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SSB-МОДУЛЯЦІЯ З ВИКОРИСТАННЯМ ЦИФРОВОГО МЕТОДУ

Спартак Маньковський, Мирон Николишин, Емілія Маньковська

З модельовано цифровий метод формування односмугової модуляції (SSB). Цифровий метод SSB модуляції в цій роботі базується на методі Вівера. Подано проектування та розрахунок цифрових блоків. Ця модель дає змогу отримувати вихідний SSB сигнал у часовій та частотній областях за заданих вхідного сигналу та системних параметрів. Це може бути корисним для задач проектування та оптимізації.

У випадку реалізації SSB модуляції на основі аналогових кіл останні вимагають застосування змішувачів, фільтрів, які побудовані на основі індуктивностей, ємностей, керамічних фільтрів, резонаторів тощо. У випадку цифрової реалізації SSB модуляції, розглянутої в цій статті, застосування аналогових фільтрів чи помножувачів не потрібне. Будь-яка фільтрація чи перемноження відбуваються у цифровій формі. Лише аналогово-цифровий (АЦП) та цифро-аналоговий (ЦАП) перетворювачі є необхідні. АЦП є необхідний для оцифрування вхідного аналогового сигналу, який зазвичай є в звуковому діапазоні частот. Швидкісний ЦАП є необхідний для зворотного перетворення цифрової послідовності в аналогову форму результуючого SSB сигналу. Цей підхід дає змогу виконувати всі необхідні перетворення без застосування будь-яких аналогових кіл та не вимагають налаштування у випадку масового виробництва.

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