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Possibilities of Analytical Signal Usage in Numerical Demodulator MSK

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Abstract:

The article deals with applications of a DSP (Digital Signal Processor) in the MSK (FSK) demodulators. It describes simplified method of numerical processing of analytical signal. By simulation procedures it features achieved results and shows possible applications.

Keywords:

Digital signal processing, demodulation

1. Introduction

When modulated signals are processed we receive an "analytical signal":

$$A(t) = A_R(t) + \mathrm{j}A_I(t)\ . \qquad (1)$$

A real part is $A_R(t)$ and it corresponds with originally received signal whose imaginary part $A_I(t)$ is the Hilbert transformation of the received signal. In references [1], [2] important features and possibilities of usage are described. From its parts, it is possible to calculate the instantaneous phase of origin signal

$$j\,(t) = \arctan\left[A_I(t)/A_R(t)\right], \qquad (2)$$

from which an instantaneous angular frequency can be extracted

$$w(t) = \mathrm{d}\,j\,(t)/\mathrm{d}\,t\ . \qquad (3)$$

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These formulas can be used for the utilization of some types of discrete modulations in the DSP. This article describes a possible variant of the digital demodulator MSK (FSK), which uses analytical signal. Its principle is relatively simple. It is based on a real-time sequence calculation $A(t)$, $j(t)$ and $w(t)$ according to the above mentioned relations; i.e. the calculation is done between the following samples of input signal. However, the usage of the DSP leads to some problems, which comes from digital processing and from limited possibilities to implement harder mathematical operations in the DSP. This article draws attention to these problems and offers possibilities to solve them.

2. Theoretical analysis

2.1. The Computation of Analytical Signal

In digital form, the imaginary part of an analytical signal might be derived by using the Hilbert transformation. It is done by the convolution of original input signal $A_R[n]$ with function $h[n]$, by other name:

$$A_I[n] = h[n]A_R[n]\,. \tag{4}$$

Practically, the Hilbert transformation is implemented as a digital FIR filter type with length *m*. The structure of the digital FIR filter is shown in Fig. 1. This structure ensures calculation of differential quadratics:

$$A_I[n] = \sum_{k=0}^{m} b_k A_R[m-k]\,. \tag{5}$$

Coefficients are in this situation:

$$b_k = h[k] = 2/kp \quad \ldots\ldots \text{ for even } k,$$

$$b_k = h[k] = 0 \quad \ldots\ldots \text{ for odd k.}$$

If the real part $A_R[n]$ is delayed for the period of mT_s then, the delay which is created by the filter is balanced and samples of analytical signal in form:

$$A[n] = A_R[n] + \mathrm{j}A_I[n] \tag{6}$$

are obtained.

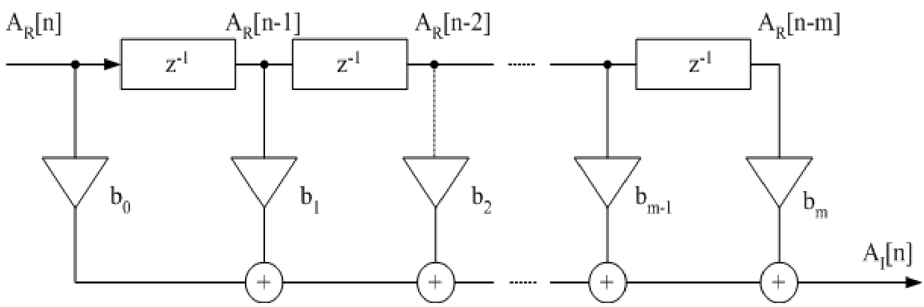

Fig. 1 Direct structure of FIR filter

This procedure can be done without any problem using DSP, including additional level balancing of signal in both lines.

2.2. Enumeration of an Instantaneous Phase

The direct utilization of the formula [2] with DSP can lead to several problems. One of them is a division operation, which cannot be done directly, but only with a separate program which can be time consuming. In addition, it is impossible to avoid zero samples in divider, which would have been necessary to eliminate. The last problem is enumeration of goniometric function by using existing functions in the DSP.

Due to the above mentioned problems other methods and procedures, which are suitable from time and accuracy point of view, were developed. Finally, for enumeration of the instantaneous phase a method which works with the constellation diagram of received signal and with a table of pre-calculated values, was chosen.

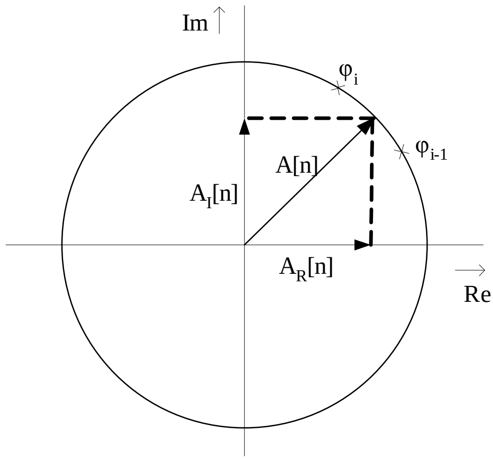

Fig. 2 Constellation diagram with parts of analytical signal

This method is based on the presumption that measurement of the instantaneous phase may not be quite exact and would have certain errors; if it has no effect on the proper function of the system. If we accept the presumption then we can calculate with definite number of *p* of discreet phases and using analytical signal analyze in which period of

$$j = j_i - j_{i-1}$$

is an actual phase. The calculation of the relation (2) can be replaced by a sequence of steps, in which the nearest discrete phase is associated to the actual value. Whole enumeration can be simplified by using absolute value of the real part of the signal (i.e. in the first quadrant) and by using the signs of both parts. Afterwards we re-calculate the phase j to the corresponding quadrant of the constellation diagram. This algorithm is simple; it uses only basic mathematical operations and requires a small amount of time.

2.3. Enumeration of an Instantaneous Frequency

Enumeration of instantaneous frequency using DSP according to formula (3) is also not possible. However, by using digital signals we can avoid this situation and replace differential of the phase by its difference

$$\Delta w = \Delta j \ / \Delta t \ . \tag{7}$$

Practically this formula can be realized by using a difference of phase samples in the time t_i and t_{i-k} . The time interval

$$\Delta t = t_i - t_{i-k}$$

can be, using k = 1, 2, 3… and known sample period T_s , set up as

$$\Delta t = kT_s \ .$$

The calculated instantaneous frequency already includes information on a demodulated signal. By doing consequential operations (for example shaping) we can use it for follow-up processing.

3. Simulation Modelling

Simulation modelling was used in order to test the aforementioned method. The objective was to test the validity of the method, to analyze its possible use for the MSK (FSK) signal demodulation and to reveal possible problems prior to DSP implementation.

The model was designed by program Matlab version 7.1. The simplified method for enumerating of the instantaneous phase and frequency described in chapter 2 is utilized in the program. The model is designed for monitoring of signals behaviour in every important places. Having this information we can evaluate the functionality of the whole system.

Values used in the simulation of the MSK demodulator are:

$f_1 = 1000$ Hz , $f_2 = 1500$ Hz , $v_P = 1000$ b/s , $f_s = 20$ kHz , monitored interval was 5 ms. Achieved results can be found in Figs 3 and 4.

The simulation showed that enumeration of formulas (1), (2) and (3) can be replaced by simplified and faster variants. Although a problem with enumeration of instantaneous frequency according to the formula (3) was also found. When phase changes from minimal to maximal value (see saw-tooth waveform in Fig. 4 an unwanted bounce of ω can appear. The reason is differences in phase samples are much bigger than in the linear part of waveform. This can be eliminated by using a program (SW).

In the model the constellation diagram was divided (Fig. 2) into 32 discrete phase parts. The number was found enough because of variables in input signal. When it was recalculated only onto the 1st quadrant we got 8 discrete parts. That is why only a few steps of decision can be found in program, which leads to a shorter enumeration time. Generally a number of discrete phase parts can be higher, thus it will lead to more accurate outputs and more time necessary for processing.

The simulation confirms that described method can be used for the MSK digital demodulation. Experiments with the model demonstrate that the same method can be used for the FSK demodulation with continuous phase.

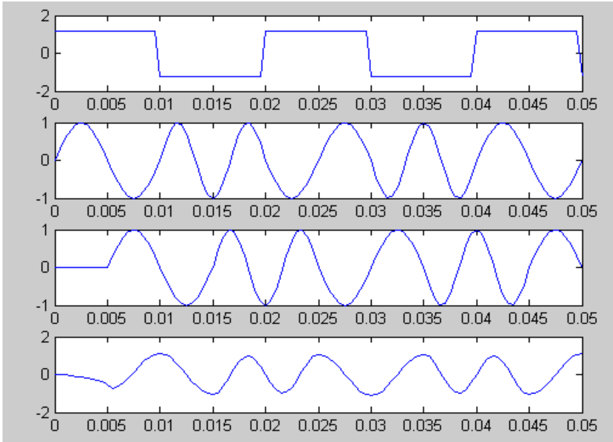

Fig. 3 Waveforms of data and modulated signal, real and imagined parts of analytical signal

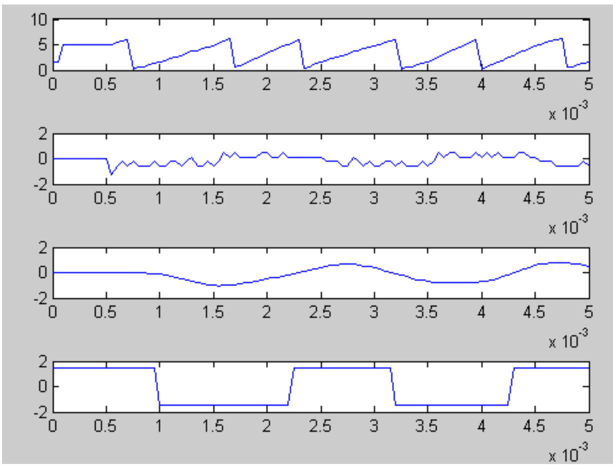

Fig. 4 Waveforms of instantaneous phase, instantaneous frequency, signal at low-pass filter input, and shaping circuit output.

4. Conclusion

Favourable results of the simulation are a prerequisite for successful realization of the described demodulator based on the DSP. Its realization must be solved in two different parts – as a program and as a circuit.

The circuit must be created minimally from three different blocks connected as shown in Fig. 5. A core is the DSP, which works with samples of MSK (FSK) signal and enable the real-time digital evaluation. Samples are obtained by a suitable type of analogue-digital converter ADC with possibility to set up any sampling period. Results (demodulated data signals) are taken from some digital DSP output (port). In minimal configuration is necessary to add Flash memory (EPROM type), in which is putted program for DSP and which is necessary for independent working (without connection with another control computer).

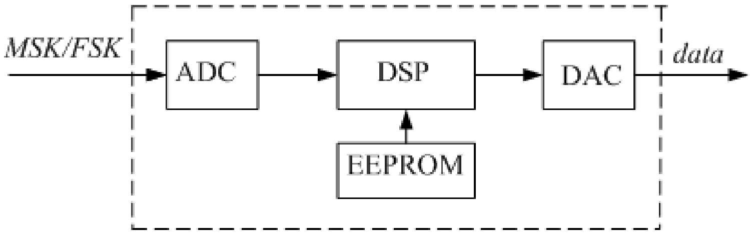

Fig. 5 Circuit schema

SW part can be realized according to the algorithm described in chapter 2. Its parts were chosen with due regards for DPS realization and they do not represent a significant obstacle from the programmer point of view.

In laboratory conditions, creation of the program, debugging, bootstrapping and controlling of correct functionality can be realized through the developmental modules with DSP [3-5], on which blocks are put in onto one main circuit board according to the Fig. 5.

We suppose that one of the components of this demodulator will be the automatic gain control subsystem, described in [6]. The function of this demodulator will be firstly verified in ideal condition, secondly in real ones (depending on the relationship of signal and noise). Achieved results will be described in the next article.

References

[1] *Hilbert Transform*. Available from:< http://wikipedia.org>.

[2] *Analytic Signal and Hilbert Transform Filters*. Available from: <http://ccrma- www.standford.edu>.

[3] *C3X Evaluation Module (EVM).* Available from:< http://focus.ti.com>.

[4] *TMS320C6416 DSP Starter Kit.* Available from:<http://focus.ti.com>.

[5] *Evaluation Kit for DSP56F801*. Available from:< http://e-www.motorola.com>.

[6] SISKA, M. and JOHANIDES, P. *The possibility of DSP application in SW band. Partial development* (in Czech). Brno : Military Academy, 2000.