IMPROVEMENT OF IMPULSE SHAPED CPM MODULATION APLICATION IN DSP

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ABSTRACT

Even today, the high-speed communication systems are in need of simple fast algorithms for data processing. This paper describes an improvement of the high speed continous phase modulation (CPM) with impulse shaping implementation method based on method described in [1]. Research was concentrated on the DSP high speed GMSK modem design for data communication with experimental satellites, but the results can also be used in more conventional devices. The improvement lies in the fact, that by the simplification of the filtering process, inphase and quadrature output samples in baseband remain only in a denumerable number of points. A very simple algorithm has been designed, which brings more accuracy and efficiency in the generation of impulse shaped CPM signals in DSP.

1 INTRODUCTION

Now let us recall the mathematical aspects of impulse shaped CPM modulation scheme using inphase and quadrature outputs in baseband, described in [1]. The block diagram is depicted in Fig. 1.

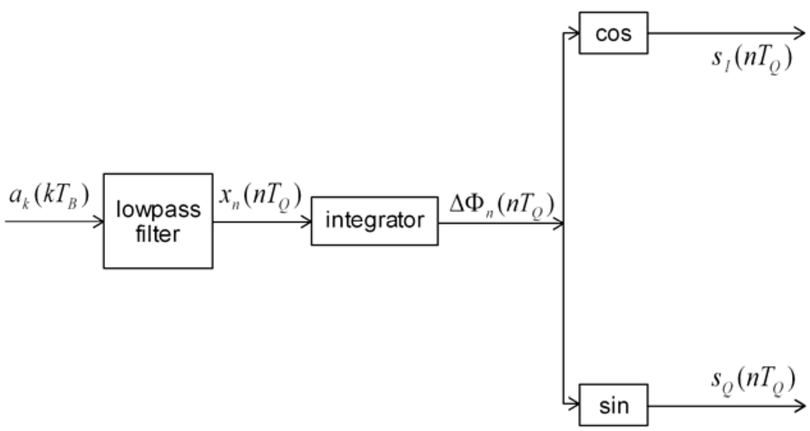

Fig. 1 *DSP baseband impulse shaped CPM modulator*

Input bit sequence $a_k(kT_B)$ (which is the combination of values -1 and 1) is interpolated and filtered by the impulse shaping filter. Bandwidth of the output signal $x_n(nT_Q)$ is given by the relative filter bandwidth $B_G T_B$, where B_G is 3 dB bandwidth of the GLPF filter. T_Q is the selected sampling period in baseband, interpolation rate is $I_r = T_B/T_Q$. Modulation signal $x_n(nT_Q)$ represents a frequency parameter of the modulation. The integrator is used to obtain the phase parameter $\Phi(nT_Q)$, whose function is given by the relation

$$\Phi(nT_Q) = \frac{\pi}{2}\sum_{i=0}^{n} x(iT_Q)\,. \tag{1}$$

Then the complex envelope is obtained by cosine and sine functions. This paper is based on the results in [1], where the CPM modulator DSP implementation is carried out partially in an innovative way as a ROM table for low pass filter outputs and partially in a conventional way using standard techniques for computing the integral and cosine and sine outputs. The new method described here gives a complete solution of impulse shaped CPM modulation in a DSP system as a ROM table which not only result in a faster algorithm but also improves the precision of output samples. This is because all output values are precomputed instead of integration errors given by the limited range of number expression in any digital system, described in [1].

2 MATHEMATICAL ASPECTS OF THE PROPOSED METHOD

The question is: "Are there only denumerable number of output phase values in the interval <0;2π>?" First let us consider the results obtained in [1], where the filter output samples are based on the relation (3). Some modification of this relation has been done, as in equation (2), in order to get more accurate congruence with an analog definition of FIR filter. In addition to that, the FIR length N is not arbitrary, but it must be the product of interpolation rate I_r and $(L-1)$, where L is the number of input bits, which are used for the computation of filter output wave.

$$\begin{aligned} y_n &= \sum_{k=j}^{I_r-2} h_{k-j} a_i + \frac{1}{2} h_{I_r-1-j} a_i + \\ &+ \sum_{l=1}^{L-2}\left(\frac{1}{2} h_{lI_r-1-j} a_{i+l} + \sum_{k=0}^{I_r-2} h_{lI_r+k-j} a_{i+l} + \frac{1}{2} h_{(l+1)I_r-1-j} a_{i+l} \right) + , \\ &+ \frac{1}{2} h_{N-1-j} a_{i+L-1} + \sum_{k=0}^{j-1} h_{N+k-j} a_{i+L-1} \end{aligned} \tag{2}$$

where

$$\begin{aligned} i &= n \operatorname{div} I_r\,, \\ j &= n \bmod I_r\,, \\ N &= (L-1) I_r\,. \end{aligned} \tag{3}$$

Considering, that the filter energy transfer is at maximum equal to one, we get

$$\sum_{k=0}^{I_r-1} h_k = \frac{1}{I_r}. \tag{4}$$

Now we can suppose, that we have an infinite input sequence $a_k(kT_B)$, where there are many occurrences of all possible combinations of (L-1) bits. When the feature of h given by the relation (4) is used in (2), maybe we will be able to find sub-sequences, where the output phase $\Phi(nT_Q)$ changes in integral multiplies of $\pi/2$. Now a very large simplification will be used. We do not need to compute directly all FIR output samples and integrate them, but we only need to integrate the number of occurrences of each input bit weighted by the bit value. Firstly, we have to find the weighted number of occurences $\Lambda_{i,l}$ of each bit while computing one FIR output wave according to the selected subsequence of L input bits:

$$\Lambda_{i,0} = \frac{1}{NI_r}\left(\sum_{j=0}^{I_r-1} x_i + \frac{1}{2}x_i\right) \tag{5}$$

$$\Lambda_{i,l} = \frac{1}{NI_r}\left(I_r^{\ 2} x_{i+l}\right) \qquad \text{, where } l = 1,..,L\text{-}2$$

$$\Lambda_{i,L-1} = \frac{1}{NI_r}\left(\sum_{j=0}^{I_r-1} x_{i+L-1} + \frac{1}{2}x_{i+L-1}\right).$$

Note that no value of the impulse response h is used. If we take a more detailed look into these functions, we can say that, when the sum of all Λ functions computed over much more than L input bits is an integral number, see (6), then the real phase $\Phi(nT_Q)$ changes exactly in the integral multiple of $\pi/2$:

$$\sum_{i=i_1}^{i_2}\sum_{l=0}^{L-1} \Lambda_{i,l} = k, \tag{6}$$

where k is an integral number and $i_2 >= i_1 + 2(L-1)$. Considering all aspects mentioned above, equation (6) has two general solutions:

$$x_{i_2-l} = x_{i_1+l},$$

$$x_{i_2-l} = -x_{i_1+l} \qquad \text{,where } l = 1,..,L\text{-}2 \tag{7}$$

The analysis of the solution of equations given in (7) implies that the phase changes only in a few discrete points, so we are able to use the look-up table, where sine and cosine values of the phase $\Phi(nT_Q)$ trajectories are directly stored. It follows from the analysis too, that the number of all possible phase trajectories in one phase quadrant is 2^L.

3 PRACTICAL REALIZATION OF IMPULSE SHAPED CPM MODULATOR

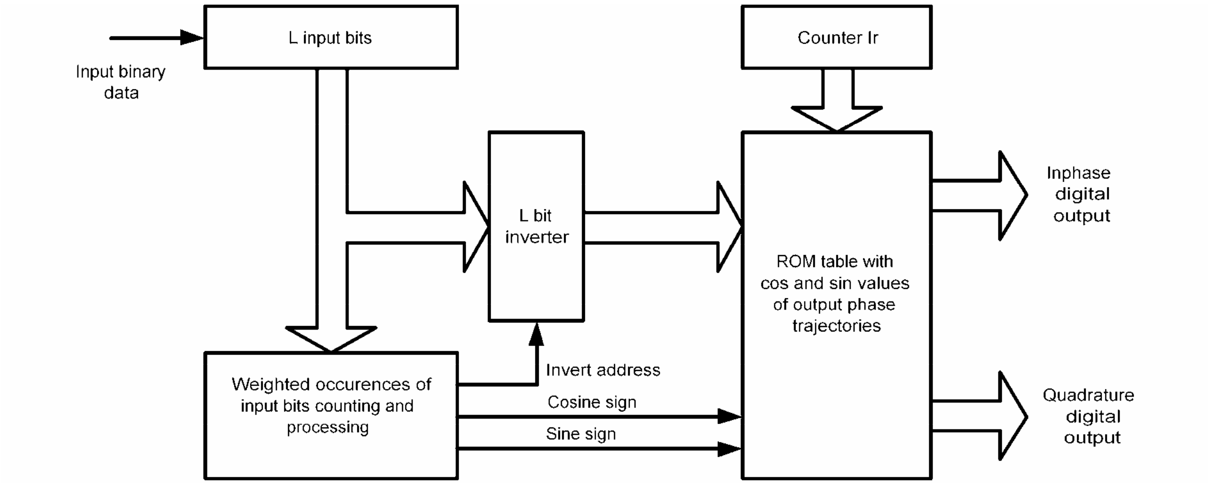

Fig. 2 *Practical realization of impulse shaped CPM modulator based on the new method*

Block diagram of the practical realization of the impulse shaped CPM modulator in DSP is depicted in Fig. 2. All possible combinations of L input bits form one part of ROM address and represent the selection of a phase trajectory. The number of phase trajectories is equal to 2^L. All 8 possible shapes of the phase trajectories in first quadrant for L=3 are depicted in Fig. 3.

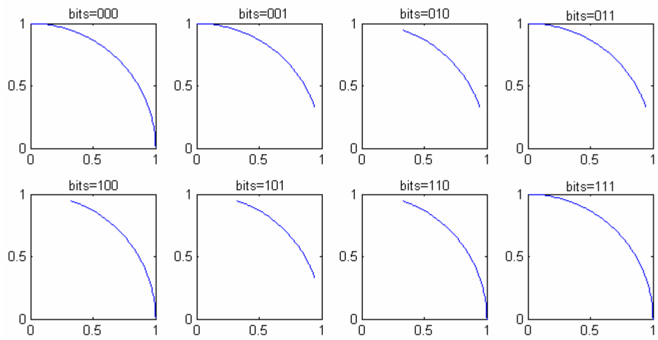

Fig. 3 *All possible shapes of phase trajectories in first quadrant for L=3*

Counter I_r forms the second part of the address and it represents the computation of output values while processing samples per one input bit. The block for counting and processing weighted occurrences of the input bits has three outputs, which determine one of four phase quadrants. Data picked from ROM are inphase and quadrature digital outputs of GMSK modulator in baseband, the next step would be the frequency translation into interfrequency as described in [1].

4 CONCLUSION

The new method of generating impulse shaped CPM modulated signals has been proposed. The main advantages are the speed of the algorithm and the high accuracy of output samples, which does not worsen with time. The low pass filter impulse response can be arbitrarily random, the only limitation is that the bit count L should be small (accordingly the length N). With the signal processor ADSP2189, which has a maximum system clock frequency of 80 MHz, we can achieve modulation rates for impulse shaped CPM well above 1 MSample/s. Further work will be aimed at realizing other modulation schemes and evaluating their performance in narrowband satellite systems.

ACKNOWLEDGEMENTS

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