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Analysis and Implementation of Efficient GMSK Modulation in Spectrum Sensing

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Abstract - This paper describes an implementation of GMSK modulator on the TMS320C6713 16-bit Floating Point Processor. GMSK is a highly bandwidth efficient scheme which is currently popular in GSM based cellular mobile systems. The modulator is configured for a data rate of 8 kbps and a BT product of 0.3, both of which are fully configurable. The implementation has been done based around a numerically Controlled Oscillator, which provides a very flexible means of implementing most of the modulation schemes. The improvement lies in the fact, that by the simplification of the filtering process, in phase 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 GMSK signals in DSP. Simulation work has been done using Elanix software package.

Keywords— GMSK, Wireless Communication, Cellular system, Computer Simulation

1. INTRODUCTION

There are a number of factors that influence the choice of a modulation scheme for use in a wireless communication. The performance of a mobile communication system is dependent on the efficiency of the used modulation method. Linear and constant envelope modulation techniques, such as BPSK, QPSK, GMSK, etc…, were used to examine the features of the required modulation scheme, and to illustrate their use in the cellular environments. The goal of a modulation technique is not only to transport message signal through a radio channel, but also to achieve this with the best quality, power efficiency, and with the least possible amount of bandwidth (1).Cellular mobile radio systems enable high-density geographical co-channel reuse and are effective for achieving efficient spectrum utilization. These systems are adopted in advanced mobile radio system plans. Digital technologies are also effectively applied to achieve not only high-speed, highly reliable data transmission but also high-grade, highly flexible system control (2).The distribution of channels and channel frequency must be ensures that assignments within one geographical cell area will not interfere with channels assigned in adjacent cell locations (4).

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The special characteristics of the mobile-radio environment can have an adverse effect upon radio propagation, and consequently can affect this quality of services provided (5). It is therefore essential to know the cause, extent, and methods for minimizing these effects in order to improve the quality and reliability of mobile-radio telephone communication. In light of many problems, many observers are now begin to look toward a next generation of mobile radio based on the digital signal processing and voice coding technology that have come so far under the impetus of increasingly powerful processors.

Digital mobile radio shares three basic characters: coding, modulation, and multiplexing. Modulation is the technique by which the digital bit stream is transferred onto radio carrier. Any modulation technique can be used to carry a digital signal of particular interest are the highly spectrum efficiency. The choice of modulation technique for a mobile radio system will rest on two considerations.

First, it should be spectrum-efficient and should occupy as narrow a transmission bandwidth as possible with a sharp power roll-off (6). Second, as with the voice orders, it should be able to withstand a high error environment. Some spectrum efficient techniques like QAM are highly efficient, but are not suitable for modulation applications because they require a stable amplitude reference not present in a fluctuating mobile signal. Conventional PSK also is unsuitable because it requires an absolute phase reference (7).First, the specific requirements on the digital modulation for mobile radio use are described. Then, premodulation Gaussian filtered minimum shift keying (GMSK) with coherent detection is proposed as an effective digital modulation for the present purpose, and its

Fig.1.DSP Based GMSK Modulation

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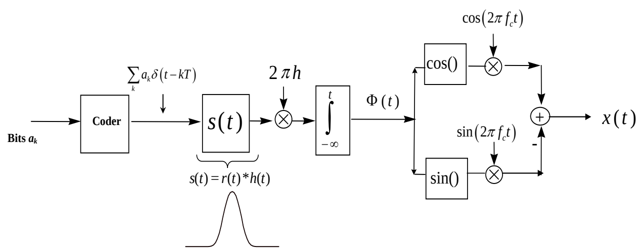


fundamental properties.The purpose of this paper is to modeling and simulates a GMSK system and exploited this system in the mobile wireless applications. To achieve this we studied the behaviour of radio waves during propagation in the radio channel, from the transmitting antenna to the receiving antenna. The majority of signal degradation occurs, and so to develop effective transmission methods it is necessary to have knowledge of what actually happens in the channel. So we simulated this channel using ELANIX package. In order to fulfil this task we modeled and simulate the transmitter and the receiver. The use of computer simulation has greatly aided this process. The adaptability of simulated communication systems is of a great advantage. Simulation also allows ease of comparison between various models.

2.IMPLEMENTATION AND PRACTICAL REALIZATION OF GMSK MODULATOR.

2.1 Processor Specifications

The implementation has been done on a TMS320C6713Processor on a DSP Starter Kit. The Starter Kit has an ELANIX software that. The sampling frequency and gain of the AIC can be programmed by changing the values in certain registers. Because of the ease of implementation, this platform was chosen.

2.2 Implementation Considerations

The Gaussian Filter needs to be implemented digitally. It has been implemented as an FIR Filter. To meet this end the frequency response of the analog filter can be sampled and its inverse fourier transform found (or) the Impulse Response could be sampled. In this implementation, the Gaussian Filter does not have considerable response beyond 2B => therefore it is sufficient if its Impulse Response is sampled (because there is no aliasing). A Window (Hamming Window) has been used to minimize the ripple. The GMSK Modulator requires the following Modules to be available:-

- Gaussian FIR Filter
- FM Modulator with Modulation Index 0.5

The Gaussian FIR Filter requires a real-time FIR Filter with configurable order and sampling Frequency to be implemented on the Processor. The Modulator is built around a real-time Numerically Controlled Oscillator with configurable parameters. Block diagram of the practical realization of the GMSK modulator in DSP is depicted in Fig. 2.

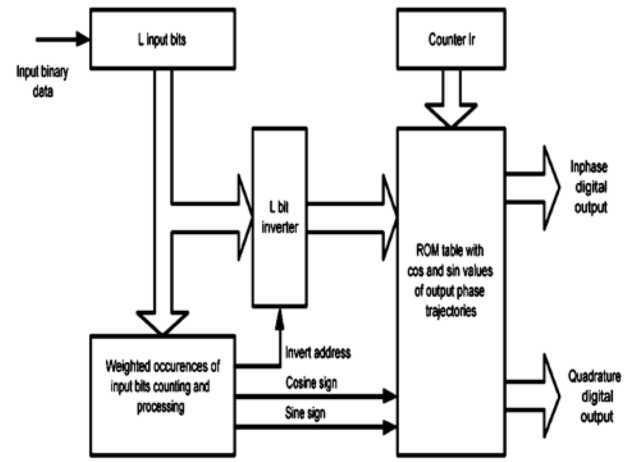


Fig 2 . Practical Realization of GMSK Modulator

Input bit sequence $a_k(kT_B)$ (which is the combination of values -1 and 1) is interpolated and filtered by the Gaussian filter. Bandwidth of the output signal $x_n(nTQ)$ is given by the relative filter bandwidth B_GT_B, where B_G is 3 dB bandwidth of the GLPF filter.

A FIR Filter needs to be implemented so the Gaussian filter coefficients can be entered in. The FIR Filter has the following considerations.The Fir Filter should respond to every ADC output. Therefore it has to be interrupt-based.Before the next sample arrives from the ADC output the previous sample has to process and its output sent out to the DAC. The past N inputs must be stored as a memory block.At each input, all the data in the block must be shifted down to make for the new input.

Finite Word length: - All the coefficients have to quantise before storing in the registers. This error (Coeffecient Quantisation Error) has been simulated and the coeffs quantised to ensure minimum error.

T_Q is the selected sampling period in baseband, interpolation rate is $Ir=T_B/T_Q$. Modulation signal $x_n(nT_Q)$ represents a frequency parameter of GMSK modulation. The integrator is used to obtain the phase parameter $\Phi(nT_Q)$, whose function is given by the relation. Denumerable number of output phase values in the interval $<0;2\pi>$" First let us consider the results obtained in [1], where the filter output samples are based on the relation (2). Some modification of this relation has been done, as in equation (1), 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 Ir and (L-1), where

L is the number of input bits, which are used for the computation of filter output wave.

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

where

$$i = n \operatorname{div} I_r, \quad j = n \bmod I_r, \quad N = (L-1)I_r. \qquad (2)$$

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

$$\sum_{k=0}^{I_r-1} h_k = \frac{1}{I_r}. \qquad (3)$$

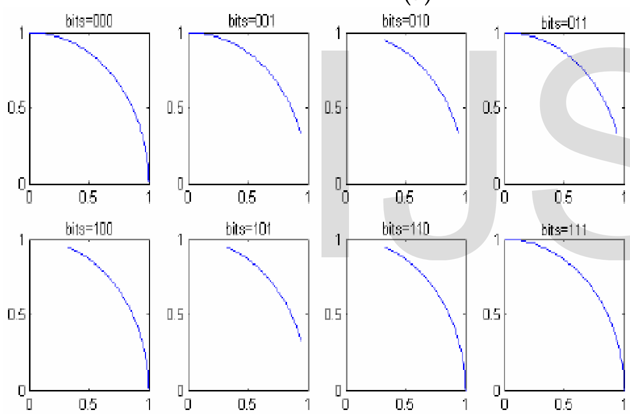


Fig3. All Possible shapes of Phase trajectories in first quadrant for L=3.

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 (3) 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 occurrences $\Lambda_{i,l}$ of each bit while computing one FIR output wave according to the selected subsequence of L input bits:

$$\Lambda_{i,0} = \frac{1}{NI_r}\left(\sum_{j=0}^{I_r-1} x_i + \frac{1}{2} x_i\right) \qquad (4)$$

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 (5), then the real phase $\Phi(nT_Q)$ changes exactly in the integral multiple of $\pi/2$:

$$\sum_{i=i_1}^{i_2} \sum_{l=0}^{L-1} \Lambda_{i,l} = k, \qquad (5)$$

Where k is an integral number and $i2 >= i1+2(L-1)$. Considering all aspects mentioned above, equation (5) has two general solutions:

$$x_{i_2-l} = x_{i_1+l}, \quad x_{i_2-l} = -x_{i_1+l}, \quad \text{where } l = 1,\dots,L-2 \qquad (6)$$

The analysis of the solution of equations given in (6) 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.

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.NCO is a very powerful tool based on the DSP concept called Direct Digital Synthesis (DDS). Direct digital synthesis (DDS) is a technique for using digital data processing blocks as a means to generate a frequency- and phase-tunable output signal referenced to a fixed-frequency precision clock source. In essence, the reference clock frequency is "divided down" in a DDS architecture by the scaling factor set forth in a programmable binary tuning word.

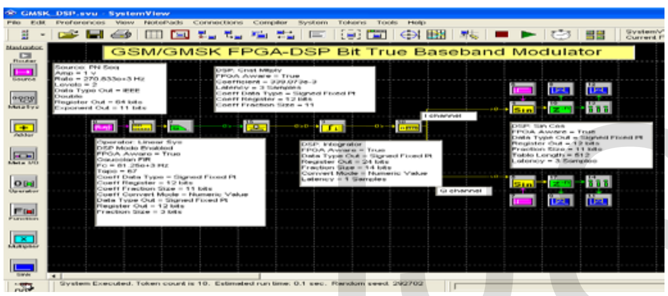
In this architecture the sine wave samples are stored in the processors memory. They are being read and output at every clock rising edge. This simple technique can be used to produce any frequency output between 0<f<fs/2.The Value of phase that has to be looked up in the LUT is determined by the frequency word (fword). That is a fixed increment is given to the phase at each clock edge. This value of phase is referred in the LUT and the sample output.In this figure the jump size, m is the fword ; 2^N is the total number of phase values for which the corresponding amplitudes are stored in the LUT. Fc is the

sampling frequency.Thus the basic architecture of an NCO can be described as follows. For PM the phase jumps by the amplitude of the input in addition to the fword every time. For FM the frequency word is altered as some Fc + Fctrl where Fctrl is proportional to the input FM Using NCO

3.Results & Discussion

To demonstrate the GMSK modulation, we are using the following randomly chosen binary data stream. {1,1,–1,1,1,–1,–1,1,–1,1,–1,–1, 1,1,–1,1,1,–1,–1,1,–1,1,–1,–1,…}.

The beginning of this data stream can be represented graphically. As the data passes through the filter it is shaped and ISI (Inter Symbol Interference) is introduced since more than one bit is passing through the filter at any time. For B_N = 0.3, since the bits are spread over two bit periods, the second bit enters the filter as the first is half way through, the third enters as the first leaves etc is shown in fig4.The beginning of the data stream being sent through the filter.



(a)

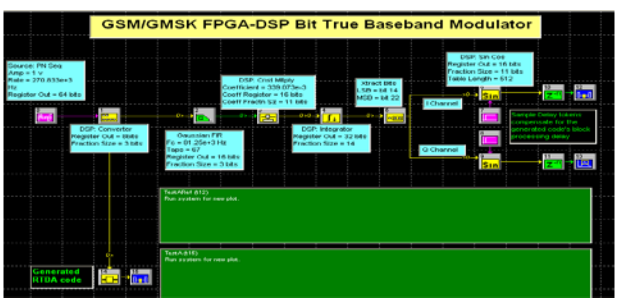


Fig4(a)GMSK Base band Modulator(b)DSP Implementation

Once we have obtained the function $c(t)$, its sine and cosine functions are found to produce the I and Q-baseband signals. Taking the cosine of $c(t)$ produces the I-baseband signal $I(t)$, i.e., $I(t) = \cos(c(t))$, which can be represented graphically show in fig 6.

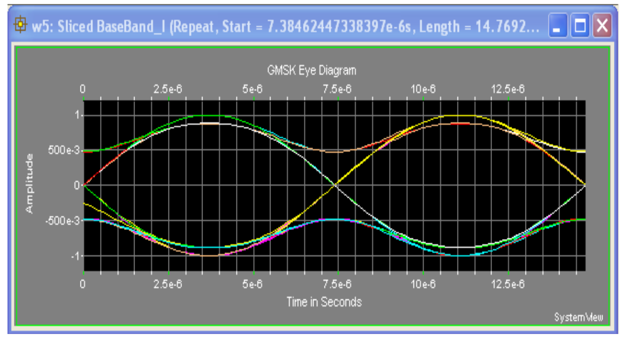


Fig5. Eye diagram for GMSK Frequency Trajectories

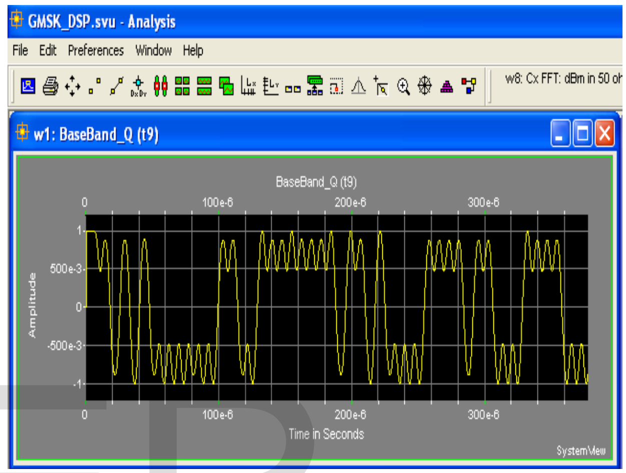


Fig6: I and Q-baseband signals

4.Conclusion

The GMSK Modulator was constructed and the output of the modulator verified with the simulation results. The MSK Modulator and the Gaussian Low Pass Filter were independently verified. The results closely matched the simulation outputs. The filter impulse response (in our case GLPF) can be arbitrarily random, the only limitation is that the bit count L should be small (accordingly the length N). Hence there is the possibility to use this approach in any other CPM modulator with impulse shaping. With the signal processor TMS320C6713, which has a maximum system clock frequency of 80 MHz, we can achieve modulation rates for GMSK well above 1 M Sample/s.

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