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DIGITAL COMPRESSING OF A BPCM SIGNAL ACCORDING TO BARKER CODE USING FPGA

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Abstract- In this paper, we introduce a practical mechanism of compressing a binary phase code modulation (BPCM) signal according to Barker code with 13 chips in presence of additive white Gaussian noise (AWGN) by using a digital matched filter (DMF) corresponding to time domain convolution algorithm of input and reference signals using Cyclone II EP2C70F896C6 FPGA from ALTERA placed on education and development board DE2-70 with the following parameters: frequency of BPCM signal fIF=2 MHz, sampling frequency $f_{SAM} = 50MHz$,pulse period $T = 200\mu s$, pulse width $\tau_S = 13\mu sc$, chip width $\tau_{CH} = 1\mu sc$, compressing factor $K_{COM} = 13$, SNRinp=1/1, 1/2, 1/3, 1/4, 1/5 and processing gain factor SNRout/SNRinp=11.14 dB.
The results of filter operation are evaluated using a digital oscilloscope GDS-1052U to display the input and output signals for different SNRinp.
Key words: Barker code, BPCM, DMF, FPGA, DDFS.

I. INTRODUCTION

Digital matched filtering is widely used for signal processing in modern Radar receivers, so the filter which realizing the digital matched filtering algorithm considers the basic and important element in Radar system. This filter defines the basic features for Radar such that, measurement accuracy, resolution, detection zone range and jamming resistance [1].

In modern Radar, structure-complicated signals with spread spectrum are used such as liner frequency modulation (LFM) signals, BPCM according to Barker codes signals, BPCM according M series signals, which have a long base $(B = \tau_S .\Delta f >> 1)$, to increase the detection range, resolution in range and velocity, and so important jamming resistance. Now a days, different digital processing algorithms are used, such as digital convolution algorithm in time domain, digital convolution algorithm in frequency domain [2], and FFT algorithm.

Complex digital convolution algorithm between input and reference signals considers the most rapid and practical one, and operates in real time, so we will use it in this research [3].

In Ref [4],Thottempudi Pardhu et al. present a compressing mechanism of LFM signal using FFT algorithm for LFM signal and stored replica.

In Ref [5], H. A. Said1 et al. present a design and realization of digital pulse compression in pulsed Radars based on LFM waveforms using FPGA.

In Ref [6], A.Naga Jyothi et al. present a generation and implementation of Barker and Nested binary codes using auto correlation function of Barker code length 13.

In Ref [7] C. D. Rawat et al. present a modern signal processing in Radar which based on the basic concept of matched filtering to achieve high signal-to-interference ratio.

II. RESEARCH IMPORTANCE AND ITS OBJECTIVES

- Using the digital matched filtering for BPCM signal according to Barker code to maximize the SNRout in the presence of AWGN effect.
- Using modern digital techniques to design the BPCM signal synthesizer according to Barker code with 13 chips.
- Using modern digital techniques to design the digital matched filter which allow getting on the desired processing gain factor under effect of interference and AWGN signals.
- Using parallel digital convolution algorithms which makes the processing operation within the real time.

III. RESEARCH MATERIALS AND ITS WAYS

To design, and test the DMF for BPCM signal according to Barker code with 13 chips in the presence of AWGN, the following tools and software are used:

- PC computer for designing and injecting the design in the chip.
- Cyclone II EP2C70F896C6 FPGA chip from ALTERA with highly accuracy, speed, and level specifications, placed on education and development board DE2-70 [8].
- DDFS which is considered as highly accuracy techniques in BPCM signal synthesizing with synchronized coherent according to Barker code.
- Digital pseudo noise generator DPNG to synthesize AWGN designed on FPGA chips.
- Digital FIR filters of highly accuracy specifications in filtering and stability and linear phase response.
- VHDL programming language with Quartus II 9.1 design environment [9].
- MATLAB11 programming environment for digital filter simulation, designing and filter coefficients computing [10].
- GDS-1052U digital oscilloscope with Free Wave program to take the results.

IV. DIGITAL CONVOLUTION ALGORITHM IN TIME DOMAIN FOR DMF

Fig.1 shows the analog BPCM signal according to Barker code with 13 chips, the width of every chip is (τ_{CH}), this signal is given by the following relation [11]:

$$S(t) = U_0(t).\sum_{n=0}^{N-1} g(n).\sin(\omega_0 t) = U_0(t).\sum_{n=0}^{12} g(n).\sin(\omega_0 t) \qquad (1)$$

Where:

$g_n = \pm 1$

$$U_0(t) = \begin{cases} 1 & \text{for } 0 \le t \le \tau_S \\ 0 & \text{for another t} \end{cases} \qquad (2)$$

For g(n)= +1, the initial phase for S(t) signal equals (0) and for g(n)= -1, the initial phase for S(t) signal equals to as shown in Fig.1.

The response of DMF can be represented according to convolution function in time domain by the following relation [12]:

$$Y(n)=\sum_{m=0}^{M-1}\{S(n-m).g(m)\}=\sum_{m=0}^{12}\{S(n-m).g(m)\} \qquad (3)$$

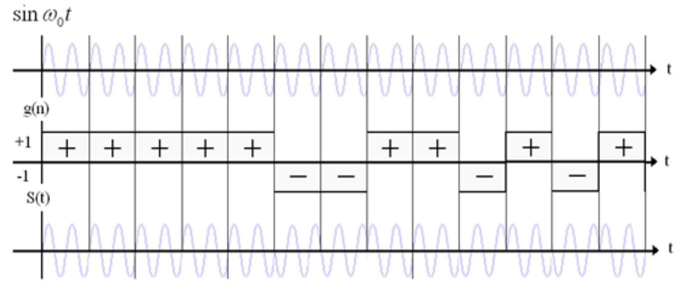

Fig. 1: BPCM signal according to Barker code with 13 chips

Fig.2 shows the pulsed signal U(t) of (τ_S) width and T time period, where this pulse is replaced by the constant Barker code from pulse to pulse of length M=13 chips and every chip is $(\tau_{CH}=1\mu sc)$ width, then this code is changed to a reference signal consists of g(n) functions with -1, +1 values which used then coefficients to the DMF.

Fig.3 shows the digital convolution algorithm diagram between the input signal and the reference signal of 13 length, it consists of 13 digital delay lines DD by amount of one chip width and of 13 shift registers RAM to record the values of g(n) function.

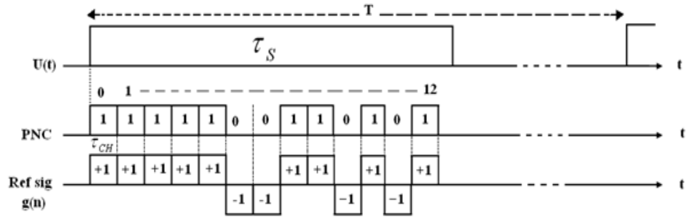

Fig. 2: Barker code of 13 chips and g(n) signals within the pulse width

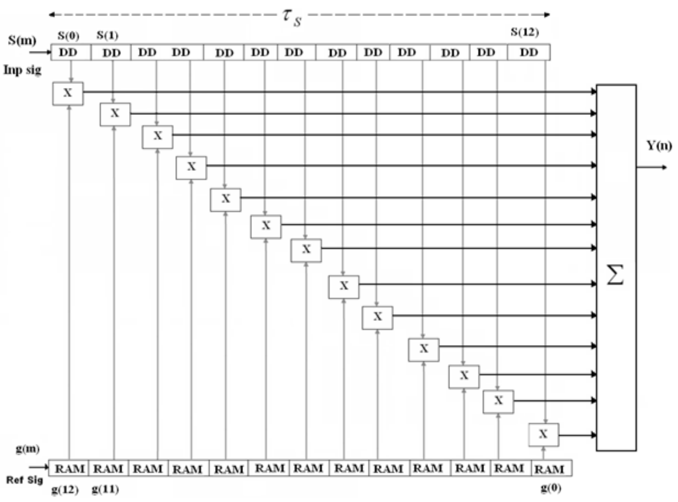

Fig. 3: Time convolution algorithm $Y(n)$ for input and reference signals with length M=13

Fig.4 shows the studied diagram of DMF for a convolution algorithm in time domain [3]. It consists of :

- Direct digital frequency synthesizer (DDFS) to create the BPCM signal according to 13 chips Barker code[13].
- Digital pseudo-noise generator (DPNG) to synthesize AWGN signal [14].
- DMF with digital convolution algorithm in time domain of compressing factor 13.
- Two DAC of 8 bits to convert the signal from digital to analog form , before filtering (DAC1) and after filtering (DAC2).
- PC to link DE2-70 through USB port to inject the design in Cyclone II EP2C70F896C6 FPGA chip [8].
- Digital oscilloscope GDS-1052U with USB port for taking the input and output signal figures of DMF in time domain for different cases of SNR inp.

This research is carried out for the BPCM signal according to 13 chips Barker code and DMF of the following specification.

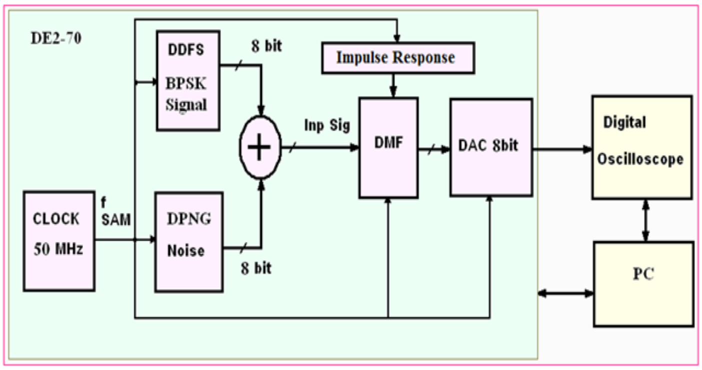

Fig. 4: The research and studying diagram for DMF

V. BPCM SIGNAL ACCORDING TO 13 CHIPS BARKER CODE SPECIFICATIONS

- Processing is done at fIF=2 MHz.
- Modulation type is BPSK according to 13 chips Barker code.
- Sampling frequency is: $f_{SAM}=50MHz, T_{SAM}=0.02\mu s=20ns$
- Pulse width before compressing is: $\tau_S=13\ \mu s$, with $\tau_{CH}=1\mu s$ for each chip.
- Pulse width after compressing is : $\tau_{COM}=1\ \mu s$, and this equals to one chip width ($\tau_{CH}=1\mu s$).
- Number of samples (reference signal length): $M=\tau_S/\tau_{CH}=13/1=13$
- Pulse period is : $T=200\mu s$.
- Delay step is: $\delta\tau=T_{SAM}=20nsc$.
- Number of delay stages for one chip is DD and given by the following relation:

$$\tau_{CH}=D_D.T_{SAM}\Rightarrow D_D=\frac{\tau_{CH}}{T_{SAM}}=\frac{1}{20.10^{-3}}=50 \qquad (4)$$

Every delay stage consists of 50 parallel shift registers (lpm shiftreg0…..lpm shiftreg49) with 8bits, all delay stages connected serially according to Fig. 5.

-Signal base is : $B = \Delta f.\tau_S = (M/\tau_s).\tau_s = M = 13$.

-Compressing factor is: $K_{COM} = \tau_S / \tau_{COM} = 13$.

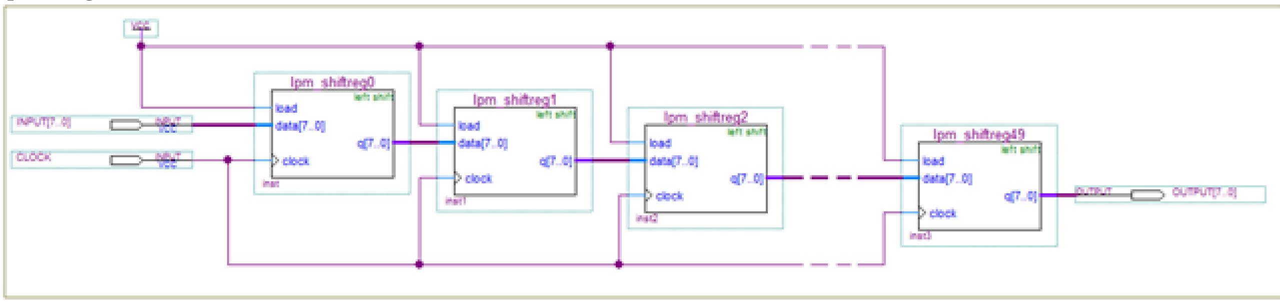

Fig.5 diagram of digital delay(DD) stages.

- Algorithm for generating Barker code with 13 chips shown in Fig.6 where the value of (decimal value 7989 or binary value 1111100110101) stored in serial-parallel shift register as an initial value for shifting and Barker code of 13 chips which constant from pulse to pulse according to Fig.7 .

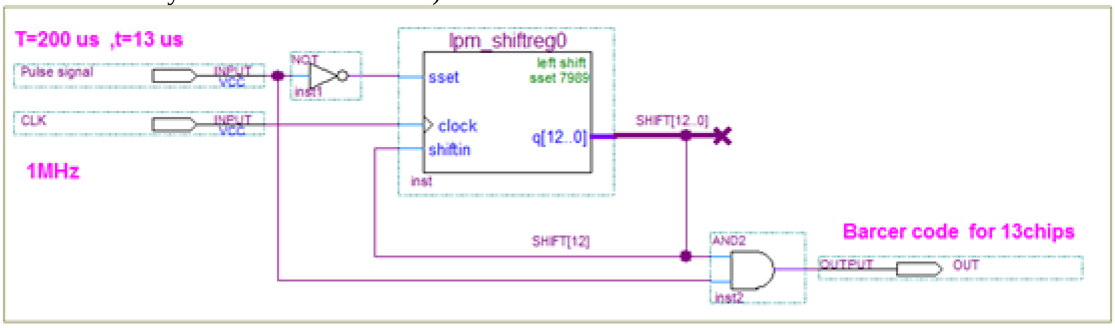

Fig. 6 Algorithm generation for Barker code with 13 chips

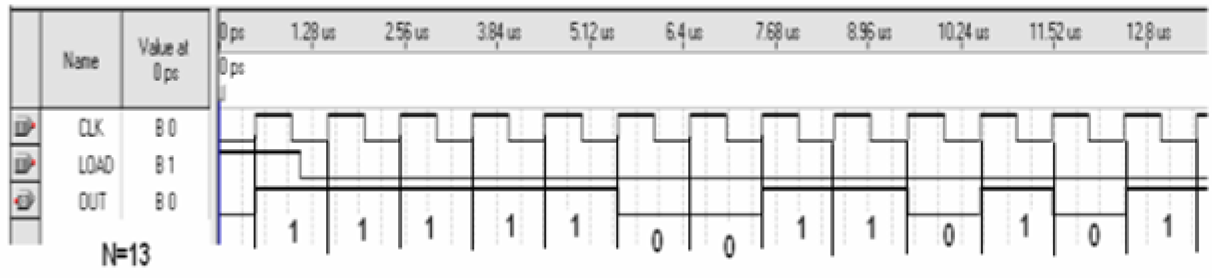

Fig. 7: Barker code signal of 13 chips during pulse width duration

- Algorithm for generating BPCM shown in Fig .8 where using DDFS with code frequency for fIF=2 MHz given by the following relation:

$$L_{f_{IF}} = \frac{2^n.f_{IF}}{f_{SAM}} = \frac{2^{32}*2}{50} = 171798692 \qquad (5)$$

and code phase (00,1800) for BPCM using DDFS given by the following relation:

$$X_{\varphi 1} = \frac{2^n.\varphi}{2\pi} \qquad (6)$$

$$CODE\ \text{PHASE}\ 0 = X_{\varphi 0} = \frac{2^n.\varphi}{2\pi} = 0$$

$$CODE\ \text{PHASE}\ 180 = X_{\varphi 1} = \frac{2^n.\pi}{2\pi} = \frac{2^n}{2}$$

Barker Code	0	1
Phase	π	0
Code Phase	$X_{\varphi 1} = \frac{2^n.\pi}{2\pi} = \frac{2^n}{2} = \frac{2^{32}}{2} = 2147483648$	$X_{\varphi 0} = \frac{2^n.\pi}{2\pi} = \frac{2^{32}.0}{2\pi} = 0$

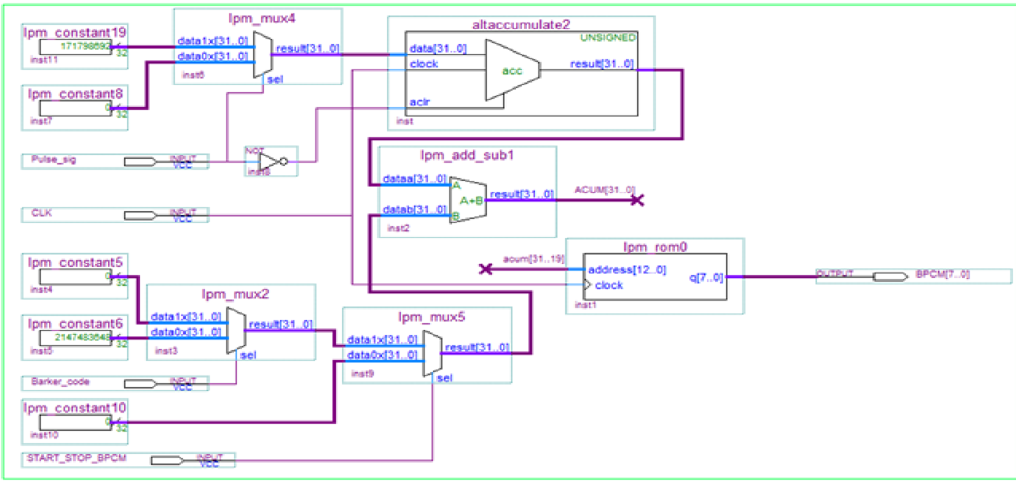

Fig. 8 Algorithm generation for BPCM signal using DDFS

- $SNR_{INP} = 1/1, 1/2, 1/3, 1/4, 1/5$
- Number of radio signal periods during the pulse width is:

$$N_{PER} = \tau_s / T_{IF} = \tau_s . F_{IF} \Rightarrow N_{PER} = 1x2 = 2 \qquad (7)$$

Where: $T_{IF} = 1/F_{IF}$ the high frequency signal period for BPCM.

VI. DMF SPECIFICATIONS

- Length of processing word for input signal is signed 8 bits.
- Number of digital multipliers is 13 with 8x3 bits.
- Number of shift registers number is 650 with 8 bits.
- Number of adder inputs is 13 with 11 bits for each one and one output with 14bits.
- Different logic and mathematic operations (AND, NOT, XOR, etc).
- Capacity of the used memory for filter confections is : 2x13 bits.
- Capacity of the used memory for DDFS BPCM is: 10KB.
- Filter order is N=M-1=13-1=12.
- Filter coefficients, where every coefficient equals to (+1) or (-1):

$$g_{12} = +1, g_{11} = -1, g_{10} + 1, g_9 = -1, g_8 = +1, g_7 = +1, g_6 = -1,$$
$$g_5 = -1, g_4 = +1, g_3 = +1, g_2 = +1, g_1 = +1, g_0 = +1$$

- Processing algorithm: digital convolution algorithm in time domain on-line.
- Input data flow speed 8bit every 20 ns:

8x50x1000000/(8x1024x1024)=48 MBPRS

-Processing speed is 13 multiplying, adding, shifting and conversion operations through 20 ns which equal 13x50x1000000=650000000 operations per second =650 million operations per second by using parallel processing (adding, shifting, multiplying, and dividing 13 digital samples with 8-bits length through one period for sampling pulses, that is, 20ns), this equivalent to 650 MHz processor clock frequency, so the processing is done simultaneously on-line.

-Processing gain at the filter output is:

$$K_{MF} = M = SNR_{out} / SNR_{inp} \qquad (8)$$

$$K_{MF}(dB) = 10\log M = 10\log 13 = 11dB$$

And it is possible to develop it up to 36 dB.

Fig.9 shows a digital convolution algorithm with M=13 in case of constant signal (Barker code is constant for all pulses), so the filter coefficients g(0)….g(12) of values (+1) or (-1) are fed to the first inputs of the multipliers ,the delayed samples with $\tau_{CH} = 1\mu sc$ are fed to the second inputs of the same multipliers.

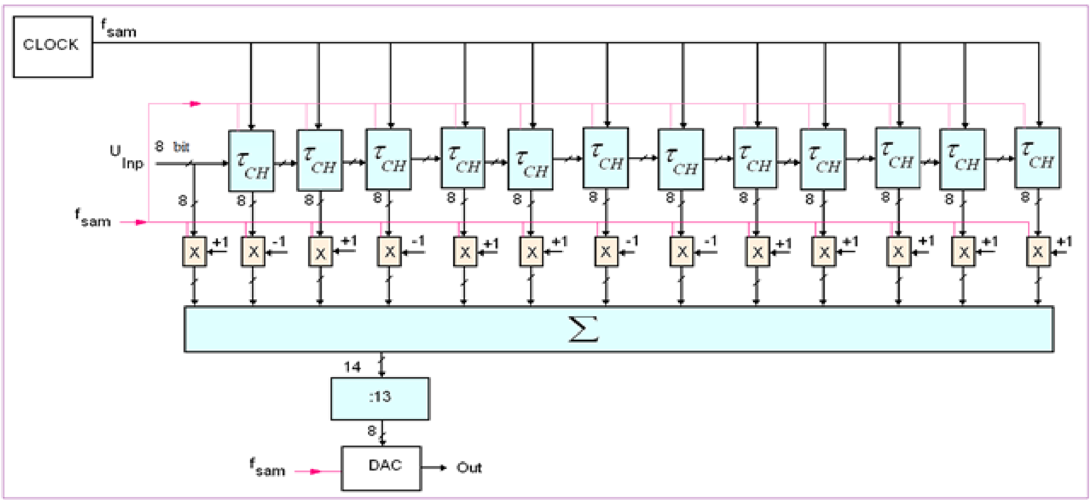

Fig. 9: digital convolution algorithm for DMF in time domain of length M=13

VII. PRACTICAL DESIGN RESULTS OF THE DMF FOR THE BPCM SIGNAL ACCORDING TO BARKER CODE

The practical design results in time domain for input and output signals of the DMF were taken by digital oscilloscope of type GDS-1052U.

Fig.10 shows on channel1 of the oscilloscope the radio pulse signal without phase coding and on channel2 the modulation pulse signal.

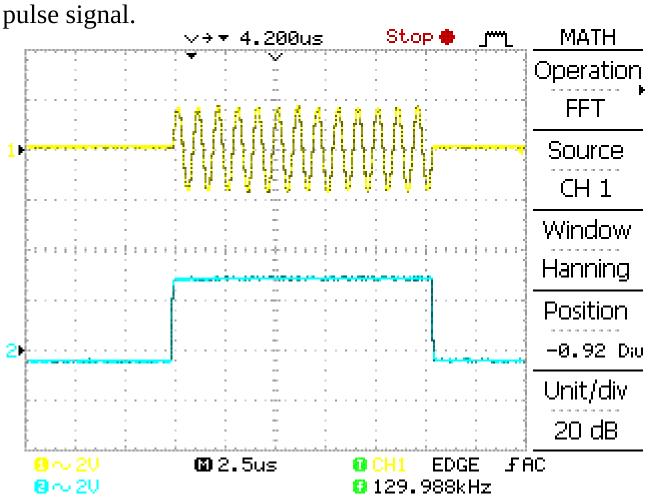

Fig. 10: pulse modulation signal

Fig.11 shows on channel1 of the oscilloscope the BPCM signal according to Barker code of 13 chips and on channel2 the modulation Barker code.

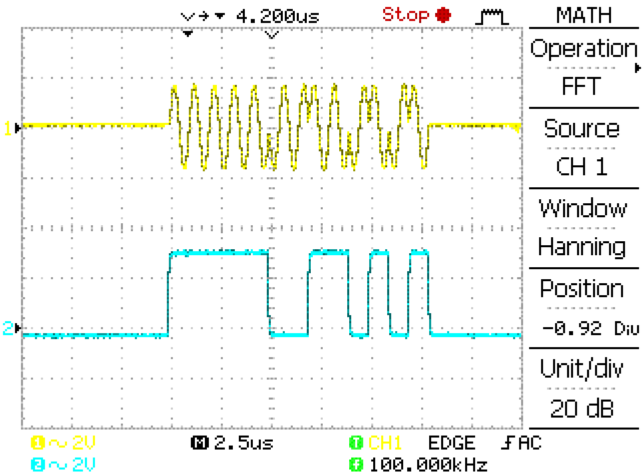

Fig. 11: BPCM signal according to Barker code 0f 13 chips

Fig.12 shows on channel1 of the oscilloscope the radio pulse signal without binary phase coding which applied on DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the filter output signal is nearly zero concerning that the designed DMF is suitable for the BPCM signal according to Barker code of 13 chips.

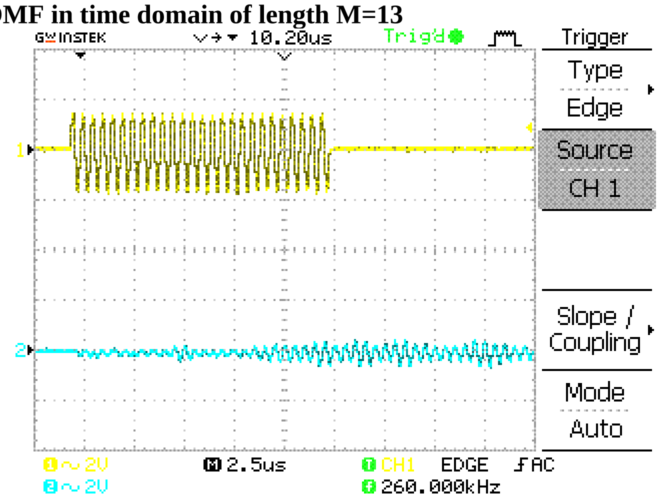

Fig.12: the input and output of the DMF for pulse modulation signal

Fig.13 shows on channel1 of the oscilloscope the radio pulse signal of BPCM according to Barker code of 13 chips without AWGN effect which applied on the DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the pulse was compressed on the filter output by 13 concerning that the filter is designed especially for this signal.

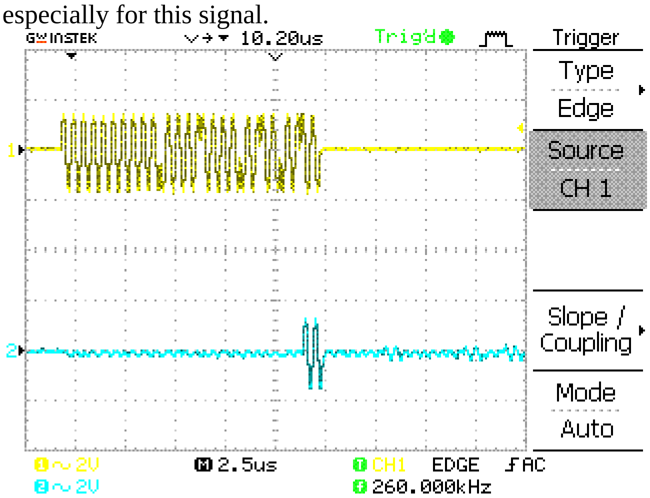

Fig. 13: the input and output of the DMF for BPCM signal according to Barker code of 13 chips without AWGN signal effect

Fig.14 shows on channel1 of the oscilloscope the radio pulse signal of BPCM according to Barker code of 13 chips under the effect of AWGN of SNR inp=1/1, which applied on the DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the pulse was compressed on the filter output by 13 concerning the filter is designed for this signal and it may possible to filter the signal with level less than the previous case because of AWGN existence with SNR inp=1/1.

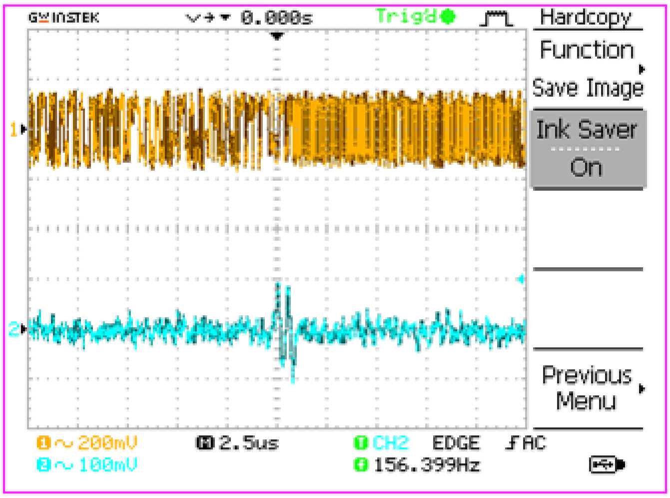

Fig. 14: the input and output signals of the DMF for the BPCM signal according to Barker code of 13 chips due to SNR inp=1/1

Fig.15 shows on channel1 of the oscilloscope the radio pulse signal of BPCM according to Barker code of 13 chips the under effect of AWGN of SNR inp=1/2, which applied on the DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the pulse was compressed on the filter output by 13 concerning the filter is designed for this signal and it may possible to filter the signal with level less than the previous case, because of AWGN existence with SNR inp=1/2.

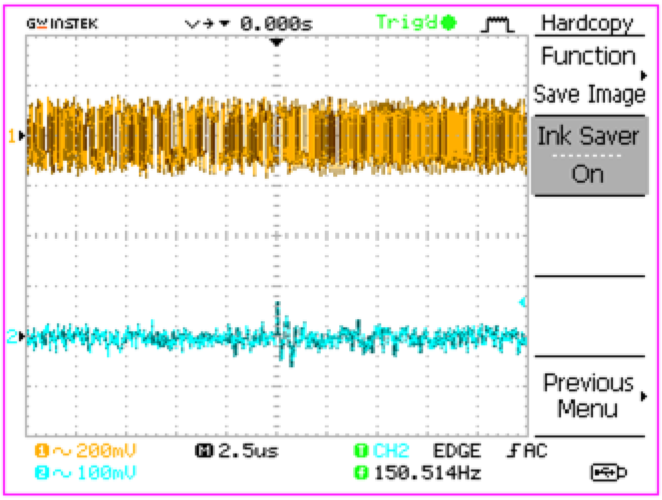

Fig. 15: the input and output signals of the DMF for the BPCM signal according to Barker code of 13 chips due to SNR inp=1/2

Fig.16 shows on channel1 of the oscilloscope the radio pulse signal of BPCM according to Barker code of 13 chips under the effect of AWGN of SNR inp=1/3, which applied on the DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the pulse was compressed on the filter output by 13 concerning the filter is designed for this signal and it may possible to filter the signal with level less than the previous case, because of AWGN existence with SNR inp=1/3.

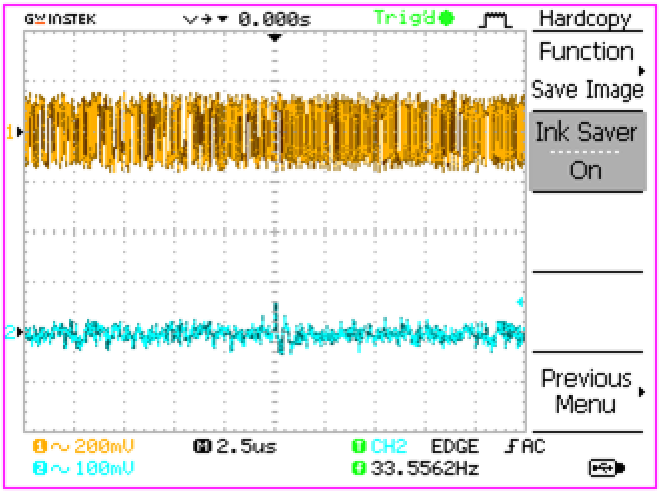

Fig. 16: the input and output signals of the DMF for the BPCM signal according to Barker code of 13 chips due to SNR inp=1/3

Fig.17 shows on channel1 of the oscilloscope the radio pulse signal of BPCM according to Barker code of 13 chips under the effect of AWGN of SNR inp=1/4, which applied on the DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the pulse was compressed on the filter output by 13 concerning the filter is designed for this signal and it may possible to filter the signal with level less than the previous case, because of AWGN existence with SNR inp=1/4.

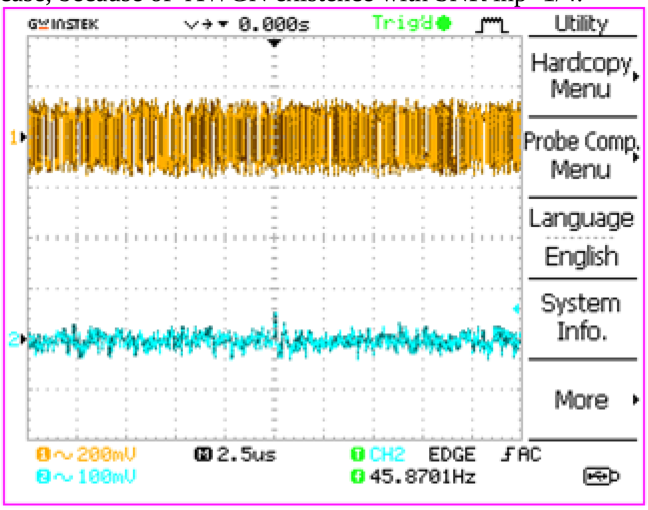

Fig. 17: the input and output signals of the DMF for the BPCM signal according to Barker code of 13 chips due to SNR inp=1/4

Fig.18 shows on channel1 of the oscilloscope the radio pulse signal of BPCM according to Barker code of 13 chips under the effect of AWGN of SNR inp=1/5, which applied on the DMF input and on channel2 the same signal is shown for the DMF output. We note from this figure, that the pulse was compressed on the filter output by 13 concerning the filter is designed for this signal and it may possible to filter the signal with level less than the previous case, because of AWGN existence with SNR inp=1/5.

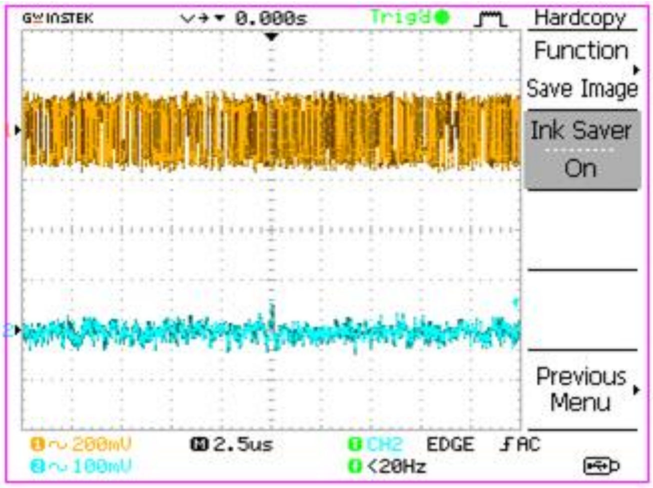

Fig. 18: the input and output signals of the DMF for the BPCM signal according to Barker code of 13 chips due to SNR inp=1/5

VIII. CONCLUSIONS

- Using of modern digital techniques by FPGA permit of design digital matched filters by digital convolution algorithms between input signal and the pulse response of the filter to obtain the required specifications for special processing gain, these techniques have high accuracy in design and performance speed (up to 250 MHz) and high integrated level (hundred thousands of digital integrated functions within one digital chip FPGA).
- FPGA techniques permit of developing DMF algorithm through serial connection for some algorithms of M order or more in input and output to obtain a long signal base and processing gain up to 36 dB for BPCM and LFM signals, this makes the radio pulses have a high effectiveness under AWGN and jamming existence.
- From practical results which obtained, we note the possibility of receiving and processing a BPCM signal according to Barker code of 13 chips under AWGN effect in cases of SNR inp=1/1, ….1/5 and this mean that the signal on the input of the filter is not seen at all, but on output, the signal is so clear because of digital matched filtering operation which achieve a matched processing gain proportional to number of samples:

$$K_{MF}(dB) = 10\log M = 10\log 13 = 11.14dB$$

- By increasing (M) through increase the pulse width and remaining the sample frequency constant or increase number of samples (M) within the pulse width (τ_S) through increase the sample frequency. So it may be increase the processing gain and extract the signal under the condition of SNR inp<1/5.

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