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DIGITAL MATCHED FILTERING FOR RADIO PULSE SIGNAL IN THE PRESENCE OF ADDITIVE WHITE GAUSSIAN NOISE USING FPGA

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***Abstract*— In this paper we discus the mechanism of synthesizing and filtering digital radio pulses practically in the presence of additive white Gaussian noise (AWGN) using digital matched filter (DMF) according to digital convolution algorithm in the time domain for the input signal(radio pulse) and the reference signal (copy of the signal) using a digital programmable device (Cyclone II EP2C70F896C6 FPGA, Altera), which was placed on an education and development board (DE2-70, Terasic) according the following parameters: sampling frequency Fsam=50MHz , pulse width , pulse period , number of samples M=300 , signal to noise ratio at filter input SNRinp=1/1,1/2,1/3,1/8 ,gain processing factor SNRout/SNRinp=24.7dB.**

Results of the filter operation studied using digital oscilloscope for input and output signals for the values of SNRinp menshend above.

Key Words: Radio pulse, DMF,FPGA ,DDFS ,AWGN..

I. INTRODUCTION

Digital matched filtering is being used widely for processing radio pulse in Radar ,GSM and GPS receivers and so ,where the filter considered as the main and the most important element in the receiver, this filter gives the most important specification of the receiver :the ability of signal extraction from noise , sensitivity of the receiver ,immunity from counter measure and so [1].

Different algorithms for digital processing are used now a days :digital convolution algorithm in the time domain and digital convolution algorithm in the frequency domain [2].

Digital convolution algorithm in time domain between the input signal and the reference signal is considered as fast and practical in the real time ,so we will give some of special and basic mathematical relation for this 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.

A. Importance of the research

-Using digital matched filtering for radio pulse to increase SNR in the presence of AWGN.

-Using modern digital technology to design the digital matched filter which enable to get on processing gain factor required.

-Using digital convolution algorithm with parallel processing for the filter and this will make the processing in the real time range.

B. The element of the research and its methods

For design and test the DMF in the presence of AWGN the following elements and program has been 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.

C. Digital convolution algorithm in time domain for DMF

The radio pulse of width (τ_S) in analog form is shown in Fig.1 and is given by the following mathematical formula[11].

$$S(t) = U(t).\sin(\omega_0 t) \qquad (1)$$

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

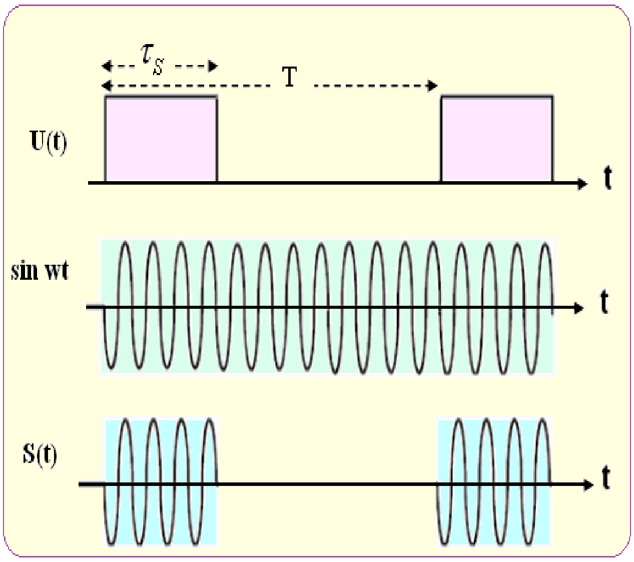

Fig1 phase coherent radio pulse

The output of DMF can be represented according to the following mathematical convolution formula[12]:

$$Y(n) = H(n) * S(n) = \sum_{m=0}^{M-1} H(m).S(n-m) \qquad (2)$$

Where :

-$H(n)$:impulse response of the filter (reference signal) and it is the time invers of the radio pulse according to the time diagram in Fig.2 where the samples of radio pulse are registered only one time in RAM.

-$Y(n)$: output of time convolution for the input signal with the reference signal.

-$S(n)$:samples of radio pulse signal (input signal).

-M: number of samples (length of reference signal).

Fig.3 shows the diagram of convolution algorithm $\{Y(n)\}$ between the input signal and the reference signal of M length.

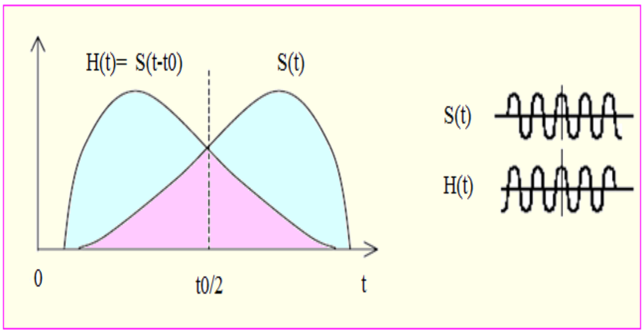

Fig.2 radio pulse and impulse response for DMF in time domain

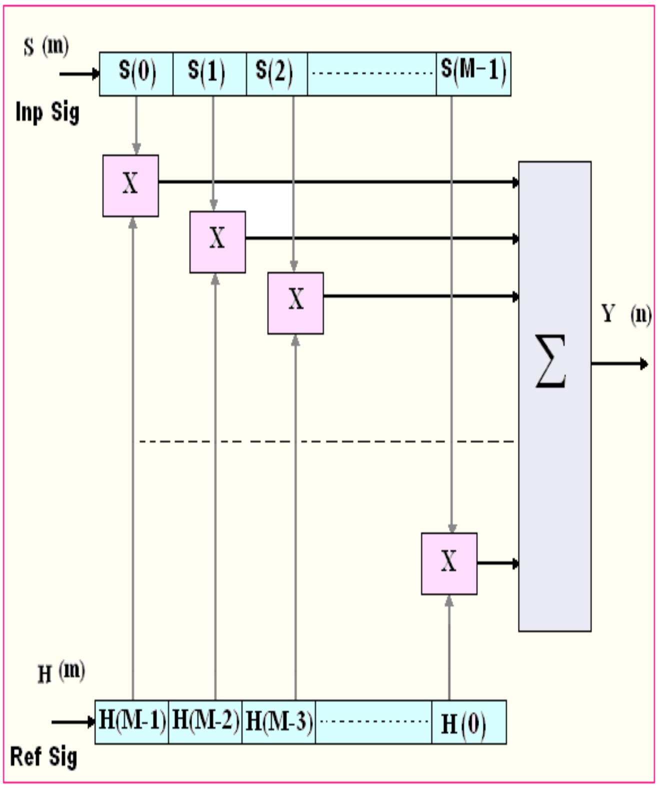

Fig.3 time convolution algorithm $\{Y(n)\}$ for input and reference signals of M length

The diagram of research procedures and study of DMF are shown in Fig.4 for digital convolution algorithm {Y(n)}and it is composed of:

-Direct digital frequency synthesizer DDFS [13] to synthesize the radio pulse signal with pulse modulation.

-Digital adder to add samples of DDFS and DPNG signals

-Digital pseudo-noise generator (DPNG) to synthesize AWGN signal [14].

-DMF with digital convolution algorithm in time domain of compressing factor 300.

-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 implemented for radio pulse signal and DMF with the following specifications.

The diagram of DDFS for radio pulse signal in Quartus II 9.1 design environment are shown in Fig.5.

D. Specifications of the radio pulse

$F_{IF} = 1MHz$ - IF processing frequency:

-Type of modulation :pulse modulation.

$F_{sam} = 50MHz, T_{sam} = 0.02\mu s$ -Frequency sampling :

$\tau_S = 6\mu s$ -Pulse width:

$M = \tau_S / T_{sam} = 6/0.02 = 300$ -Number of samples (length of reference signal):

$T = 100\mu s$ -Pulse period :

- $SNR_{\text{inp}} = 1/1, 1/2, 1/3, 1/8$

$\delta\tau = T_{sam} = 20ns$ -Delay step

-Number of radio signal periods during the pulse width is:

$$N_{PER} = \tau_s/T_{IF} = \tau_s.F_{IF} \tag{3}$$

$$N_{PER} = 6*1 = 6$$

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

E. Specifications of DMF for radio pulse

-The length of processing word for input signal :signed 8bits.

-Number of used digital multipliers:300 multipliers 9x9 bits.

-Number of shift registers with length 8bits :300SR according Fig.6.

-One adder of 300 input 16 bits and one output with 27bits.

-Capacity of used RAM :10KB.

-Filter order: N=M-1=300-1=299.

-The speed of input data flow 8bits every 20 ns(8x50x1000000/(8x1024x1024)=48 MBPRS).

-Processing speed 300 operations (multiply ,add ,shift, etc .) through 20 ns time ,and this equal to $15x10^9$mathematical operations (multiply and add)every one second through the use parallel processing multiply and add for 300 digital samples with 8bits length ,shift ,division and other operations at the same time through one period for pulse sampling 20 ns and this equal to frequency of clock pulse 15.0GHz.So that the processing operation considered as real time (On-Line).

-Matched processing gain factor:

$$K_{MF} = SNR_{\text{out}} / SNR_{\text{inp}} = M \tag{4}$$

$$K_{MF}(dB) = 10\log M = 10\log 300 = 24.7dB$$

-Digital convolution algorithm is shown in Fig.6 with number of samples (length) for reference signal M=300 for statement signal parameters (frequency) so that the value of samples for reference signal are registered once symmetrical with the input signal in shift registers H(0)…..H(299) through the pulse width by signal Single M and then the time convolution between the two signal is implemented with every samples pulse.

-Capability of developing the algorithm through the series link of multiples of algorithms at input and output to get on higher order of the filter and gain processing factor up to 36dB.

-Chang in the specification of radio pulse will change the specification of DMF, so for each radio pulse there is a DMF for it ,to have a constant structure (algorithm) for DMF the ratio between (τ_S) and (T_{sam}) must be:

$$M = \tau_S / T_{sam} = const \tag{5}$$

$$F_{IF} = \text{const}$$

i,e for the designed filter where: $\tau_S = 6\mu s$, $T_{sam} = 0.02\mu s$:

$$M = \tau_S / T_{sam} = 6/0.02 = 300$$

$$\tau_S = 300\text{x}(1/60) = 5\mu s$$ And for F_{sam}=60MHz

pulse width must be:

$$\tau_S = 300\text{x}(1/50) = 6\mu s$$ And for F_{sam}=50MHz

pulse width must be:

$$\tau_S = 300\text{x}(1/30) = 10\mu s$$ And for F_{sam}=30MHz

pulse width must be:

And so.

FIF=1MHz						
F_{sam}[MHz]	10	20	25	30	50	60
$T_{sam}=1/F_{sam}$ [μs]	0.1	0.05	0.04	0.033	0.02	0.016
$\tau_S[\mu s]$	30	15	12	10	6	5
$M=\tau_S/T_{sam}=\tau_S\cdot F_{sam}$	300	300	300	300	300	300
$N_{PER}=\tau_s/T_{IF}$	30	15	12	10	6	5
DMF	The same filter					
Filter coefficients	Variable					

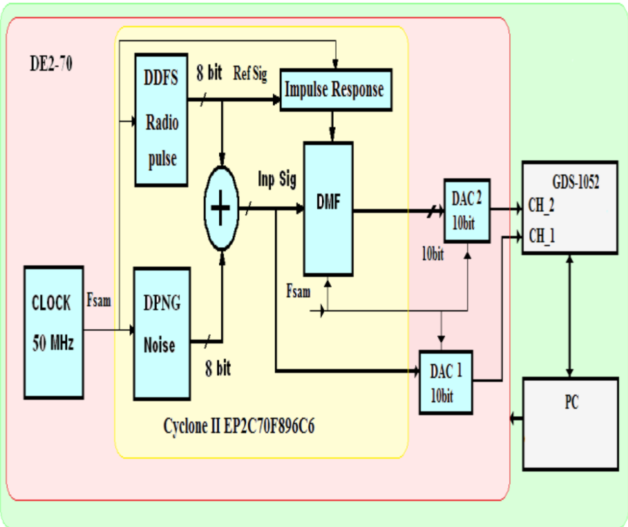

Fig.4 diagram of research and studying of DMF

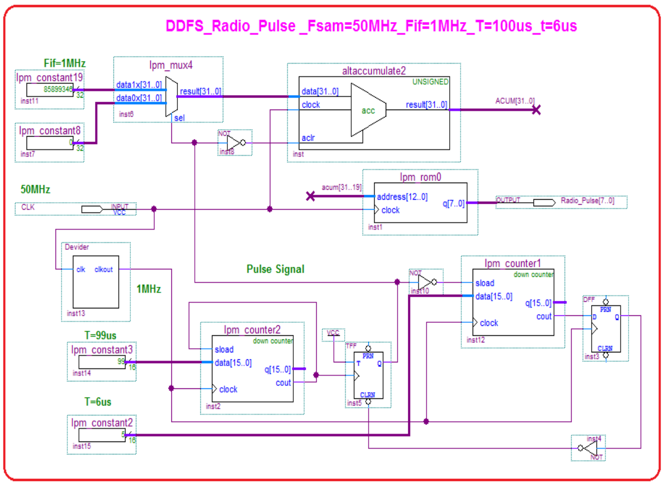

Fig.5 diagram of DDFS radio pulse in Quartus II

9.1 design environment

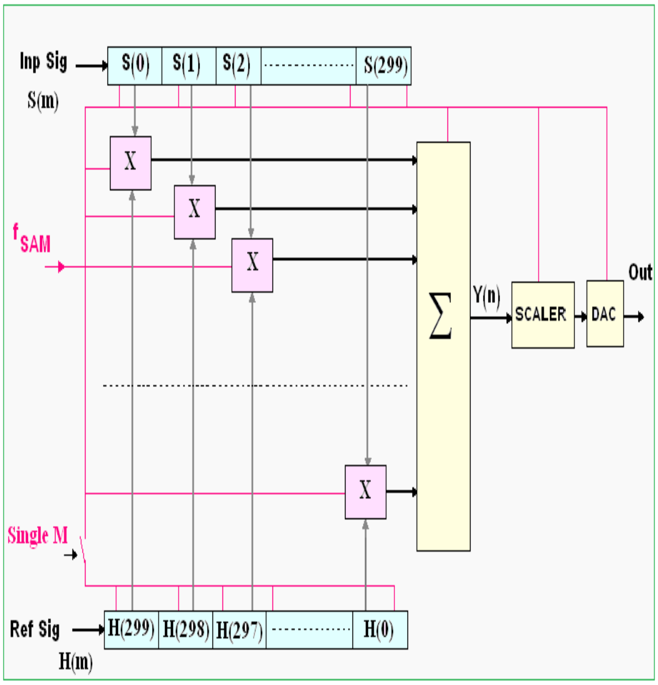

Fig.6 digital convolution algorithm of DMF in time domain

F. The results of practical design of DMF for radio pulse

The results of practical design of DMF for radio pulse in time domain for input and output signals are taken from digital oscilloscope type GDS-1052U.

Fig.7 shows on the first channel (CH_1)radio pulse signal without the influence of noise for pulse modulation at the input of DMF and on second channel (CH_2)the output of DMF for the same signal .We note that the pulse width at the output of DMF has been increased twice where the filter is designed especially for this signal.

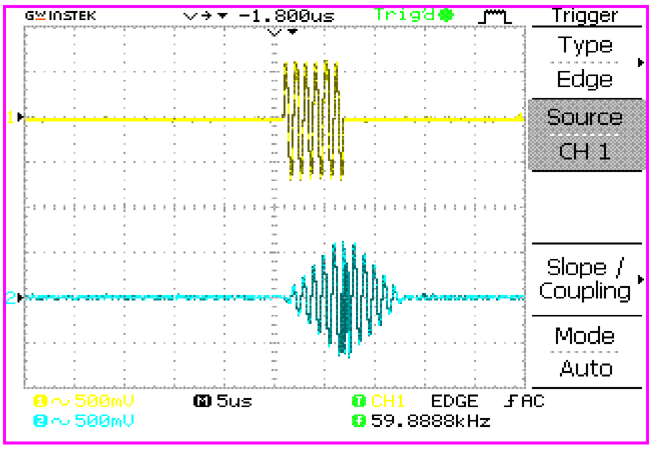

Fig.7 input and output signals of DMF without noise

Fig.8 shows on the first channel (CH_1) radio pulse signal with AWGN and SNR_{inp}=1/1 for pulse modulation at the input of DMF and on second channel (CH_2) the output of DMF for the same signal .We note that the signal has been extracted at the output of DMF considering the filter is designed especially for this signal but the level is less than the previous case because of the presences of AWGN with SNR_{inp}=1/1.

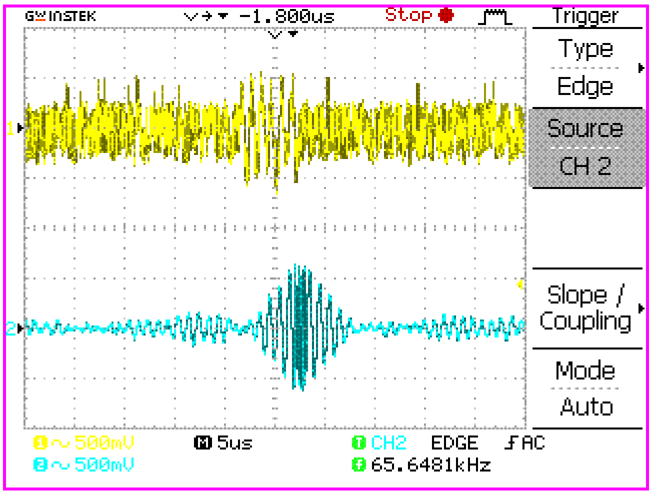

Fig.8 input and output signals of DMF for SNR_{inp}=1/1

Fig.9 shows on the first channel (CH_1) radio pulse signal with AWGN and SNR_{inp}=1/2 for pulse modulation at the input of DMF and on second channel (CH_2) the output of DMF for the same signal .We note that the signal has been extracted at the output of DMF considering the filter is designed especially for this signal but the level is less than the previous case because of the presences of AWGN with SNR_{inp}=1/2.

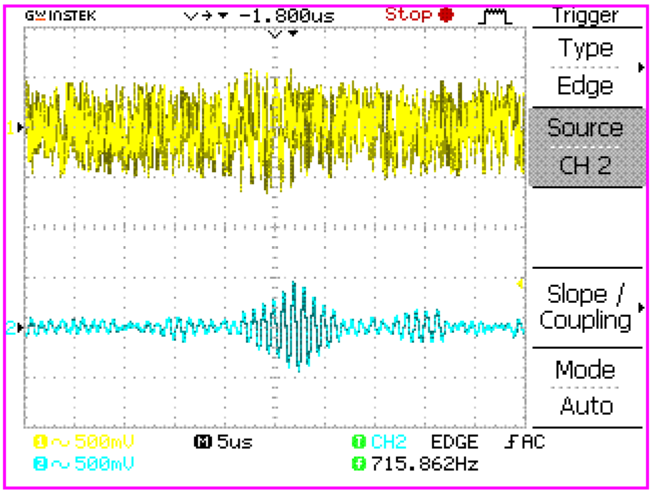

Fig.9 input and output signals of DMF for SNR_{inp}=1/2

Fig.10 shows on the first channel (CH_1) radio pulse signal with AWGN and SNR_{inp}=1/3 for pulse modulation at the input of DMF and on second channel (CH_2) the output of DMF for the same signal. We note that the signal has been extracted at the output of DMF considering the filter is designed especially for this signal but the level is less than the previous case because of the presences of AWGN with SNR_{inp}=1/3.

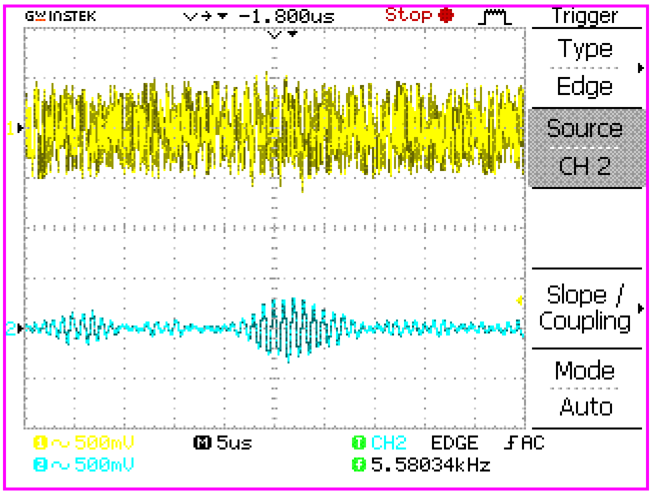

Fig.10 input and output signals of DMF for SNR_{inp}=1/3

Fig.11 shows on the first channel (CH_1) radio pulse signal with AWGN and SNR_{inp}=1/8 for pulse modulation at the input of DMF and on second channel (CH_2) the output of DMF for the same signal. We note that the signal has been extracted at the output of DMF considering the filter is designed especially for this signal but the level is less than the previous case because of the presences of AWGN with SNR_{inp}=1/8.

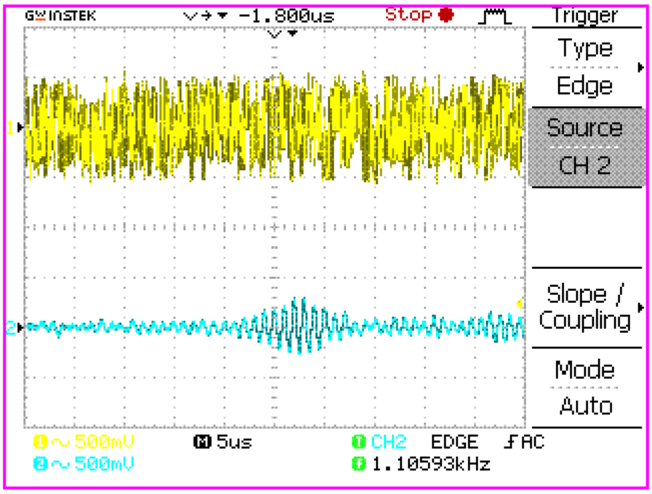

Fig.11 input and output signals of DMF for SNR_{inp}=1/8

II. Conclusion

-Using modern digital techniques (FPGA) enabled designing DMF by digital convolution algorithms for signal input and impulse response of the DMF to get on the required specification with processing factor .Those techniques have high accuracy specification design, speed (up to 250MHz) and high integrated level (up to hundreds of thousands of integrated digital function inside one digital FPGA).

-FPGA techniques enabled the development of DMF through the serial link of multiple algorithm (300 or more) at input and output to get greater signal base and processing gain up to 36 dB, and this will make the radio pulse more effective in presence of noise.

-From the practical results obtained we note the possibility of reception and processing for case $SNR_{inp}=1/8<<1$ when the signal at the input of DMF un-noticeable while at the output the signal is very clear even for $SNR_{inp}=1/8$ because of the operation of matched filtering which implements matched processing gain factor proportional to number of reference signal samples $K_{MF}(dB) = 10\log M$,by increasing M we can increase matched processing gain factor and extract the signal in worst environments ($SNR_{inp}<1/8$).

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