Design and Software Implementation of Analog and Digital Filters For Coherent Receivers under Noisy Channels

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Abstract. One of the most important stations in any communications system is the filtering station. Due to the noise power that effects on the signal, the system must include filtering process to eliminate this effect. In this paper, analog and digital filters are designed and implemented using Matlab tools to solve the problems occurred during receiving the signal that effects on the synchronization such as those for coherent receivers. The offsets of Phase and frequency are major problems in the identification of signals that results reduction in the performance; such as, whenever higher order modulations are used. Two systems are designed and simulated. The first one is analog system and the second one is digital system. 3rd order Butterworth filter is selected for system one and digital IIR filter is selected for system two. Filter order was selected according to diagnosis the level of noise power and the transmitted frequency to meet certain frequency domain design specifications. Obtained results showed excellent performance in the receiving and demodulating the received signal. Evaluation of the results is basically depend on comparison between the transmitted and received samples.

Keywords — Butterworth, IIR, PLL, VCO.

1. Introduction
The signal processing is a mathematical operation to modify or derive the response which basically depends on the operation of the filters. The filtering processes are used so as to produce a linearity invariant time system used to perform spectral shaping [1]. This process is largely used in schemes, like to remove the unwanted noisy signals from the signals, shaping of the spectral like the adaptation of channels, signal detection and identification in cognitive radio, communications and for obtaining the analysis of the spectrum for different signals [2]. Signal processing may be programming and reprogrammed with no need to great change in hardware. Signal processing include properties like the accuracy, perfect re-producibility, the flexible and good quality. Generally, DSP signal is derived from the analogy signal that sampled within a certain time to change it into digitally form [2].

The filter is the heart of any communication system at the signal processing stage. As mentioned early, there are two filters are used in this paper: Butterworth and IIR filters. The first one (Butterworth filter) is designed due to its flat frequency response in the passband. It is first discovered at 1930 due to British engineer and the physicist Stephen. It is known as flat magnitude. It has no ripple. Its becoming flat from passband to stopband. As increasing number of filter order (n), the phase response of the Butterworth filter becomes more nonlinear. Several filter orders along with the ideal frequency response which is known as a brick wall. If the filter order nth increase, then brick wall response and filters gets closer as shown in Figure 1 [3, 4, 5].
Equation 1 is frequency response of low pass Butterworth filter.

\[
G(w) = |H(jw)| = \frac{1}{\sqrt{1 + w^n}}
\]  

(1)

The filters are very important schemes in DSP design in order to change frequency properties of the signal so as to produce the design properties. Digital filters considered to remove noise, spectrum shaping and interference reduction in communication architectures [6, 7]. Generally, the classes of the filters in the digital form are:

(i) FIR: Finite Impulse response filter
This filter has impulse response is in finite duration, because it approximate to zero in specified time. This filter is designed and implemented usually using numbers of adders, multipliers, delay and adders. FIR filters also called as non-recursive feed forward. It has a structure which may be used to implement most of any sort of freq. response digitally. The unit of impulse response is finite; so FIR filters are stable system [6, 7].

(ii) IIR: Infinite Impulse response filters
IIR filters can be designed and implemented as analog or digital form. In the digital forms, the output feedback is immediately seen on the equations defining the output [8, 9]. Digital IIR filters Design is greatly depend on analogy counterparts because there are plenty of works and straightforward design methods concerning analogy feedback filter design. Unlike FIR filters, in the design it is necessary to be carefully consider the time zero case in which the filter outputs has not yet clearly defined [10, 11, 12].

The designed filters in this paper submitted to coherent receivers to recover the original transmitted signal. As mentioned early, this paper is divided into two parts: the first one related to analog system and the second one is the digital system.

2. Proposed system design
2.1. System 1
In this system, the Butterworth filter is used together with the phase locked loop (PLL) to recover the signal that changed due to the phase and frequency offsets; however, the PLL is a feedback loop has the ability to recover phase and frequency of the received signals. The PLL is combined from: multiplier, voltage controlled oscillator and the loop filter. Figure 2 shows the basic construction of PLL [11, 12].

In Figure 2, the input to the multiplier is the received noisy signal and the output of the vco. This multiplier is simply an XOR gate that compare between two input to produce the output, vco is an oscillator that controlled by input voltage [14, 15, 16, 17]. The loop filter is what we want to design in this paper. The Butterworth is preferred here because it is not has a ripple. In general, two PLL is used
Figure 2. PLL basic construction [12].

Here to accommodate the complex received signal; therefore, those two PLL will be separated by shifting 90° between the output of VCO as shown in Figure 3.

Figure 3. Combinational of two PLL [15].

Where: \( i(t) \), \( q(t) \) are In phase and Quadrature phase respectively. \( u_i(t) \) is the received noisy signal. \( u_{o1}(t) \) and \( u_{o2}(t) \) are the output of VCO for I and Q channels respectively. M1, M2 are the multipliers on I and Q channels respectively.

The block diagram of Figure 3 consists of two PLL shifted by 90°. The two arm filters is selected to be Butterworth filter. Those filters (I and Q arms) are selected carefully to be the exactly the same using filter design and Analysis Tool (FDA) in Matlab program. The design parameters of the proposed system and the characteristics of each filter are shown below in Table 1:

Table 1. Characteristic of the proposed system.

| Filter type | Butterworth filter |
|-------------|--------------------|
| Filter order | Third Order        |
| Cutoff frequency | 10 KHz            |
| Modulation type | BPSK              |
| Channel type | AWGN               |

The structure of the detailed proposed system and the designed filter based on Matlab program (i.e. FDA tool) is shown below on Figure 4.

2.2. Simulation Results of System 1

The simulation results obtained from Figure 4 are shown below:
First of all, the general characteristics of the designed filter are seen from the Figures 5 and 6. Figure 6 represents the magnitude response of this filter. It is seen that at -3dB the cutoff frequency is 10 KHz as planned before. Figure 6 is the group delay and the pole/zero location plot respectively. The designed filter is stable as shown in the lower plot of Figure 5.

AWGN is additive to the signal as shown in Figure 7 that shows the generated AWGN on the scope 1.

Figure 8 represents input/output of the Butterworth filter that designed in this paper. As mentioned above, the order of this filter is 3rd order for both sides (I and Q) to meet the requirements for the coherent receiver that proposed in this paper. The order of the filter is selected according to the experimental results to obtain the best output at specified cutoff frequency.

As seen from Figure 8, the upper plot represents the signal that produced from the multiplication between the output of the first VCO and the received signal in the I-side (before filter), whereas the lower plot represents the output signal from the filter that appear on scope 2. The quality of the PLL
Figure 6. Magnitude response of 3\textsuperscript{rd} Butterworth filter.

Figure 7. The generated AWGN.

is evaluated according to ability of the proposed block to track the phase offset and freq. offset of the transmitted signal; however, at the time when output is measured on I-side, the Q-side is approximately is zero. This idea can be shown in Figure 9 that plot the output of the Butterworth filter in Q-side which is approximately is zero.

The error that generated on scope (3) of Figure 4 is stand for the DC offset voltage which is the input to the VCO. This error is preferred to be zero at the lock state that determines that the true signal is received. This error is seen on Figure 10.

The designed system shown in Figure 4 represents coherent receiver for the BPSK signal. The basic job of this system is recover the phase and frequency offset occurs due to the synchronization errors. The heart of this system is the filter; however, the real test for the performance evaluation of this system is the comparison between the transmitted and the received (demodulated) symbols of the proposed system.
Figure 8. Input and output of the Butterworth filter.

Figure 9. The output of filter on Q-side.

Figure 11 represents the comparison between the transmitted and the output of the system.

As clearly seen in red colour from Figure 11, the system need a time to reach and get the fully synchronization between the receiver and the transmitter. The system will stay in lock state and therefore the error is zero as shown in Figure 12 after rerun the system, the output is fully synchronized with the transmitted signal. This fully compatible was not obtained without the designed filter.

2.3. system 2 (digital form)

The same block diagram that proposed in the system 1 (Figure 4), is designed except replacing the filter by digital IIR filter and discrete VCO instead of continuous VCO. The designed was based on Matlab program tool (FDA tool). IIR filter is designed according to Table 2.

Figure 13 represents the proposed system 2 that consists of digital IIR filter, discrete VCO and digital
received signal.

This diagram is implemented by Matlab program and the values of each block is selected to obtain the results as shown in Table 2. However, the simulation results are shown in the next section.

2.4. Simulation results of system 2

The generated sample is modulated using carrier signal. The resultant BPSK is additive by AWGN to reach the coherent receiver that will have the ability to recover the original signal without noise and then finally demodulate the original samples. The core of this diagram is the infinite impulse response (IIR) filter that is low pass filter. The characteristics of designed IIR filter is shown in Table 3.

The structure of this IIR filter is basically based on Z-Transform that consists of numbers of delays as shown in Figure 14.
The filter is stable filter consists of two section as shown in Figure 14, is 3\textsuperscript{rd} order at cut off frequency of 10 KHz. The magnitude response is shown in Figure 15.

Figures 16 and 17 represent the group delay and the zero pole plot respectively.

It is very important to check the response of the filter and evaluate the performance. Figure 18 shows this test. The upper represents the received signal before filtering, the lower plot represents the filtered signal, it is seen that the filter is working well.

As mentioned early in this paper, when the lock state is happen, the Q-side will be approximately zero. This is can be seen in Figure 19 that represent the signal after the filtering process of the Q-side. The red rectangle means that the system need a time to get the lock state and the error to be zero.

Final and real test for the performance of the proposed system is to compare between the generated and the recovered samples. This shown in Figure 20.

As expected, the system need a time to reach the lock state. This clearly seen in the start of lower part of Figure 20 which indicated by red rectangle. Finally, it is necessary to see the freq. spectrum of the received signal as shown in Figure 21.

3. Conclusion
As a results from the designed systems of this paper it is easily to see the good response and good filtering process. Designed filters and selection the orders of the filters is very important to obtain the required
Figure 13. Block diagram of the proposed system 2.

Table 3. Characteristics of designed IIR

| Discrete Time IIR Filter (real) |
|--------------------------------|
| **Filter Structure**            | Direct Form -II, Second Order Sections |
| **Number of Sections**          | 2                                       |
| **Stable**                      | Yes                                     |
| **Linear Phase**                | No                                      |
| **Scale Norm**                  | No Scaling                              |
| **System Object**               | False                                   |
| **Design Specifications**       |                                         |
| **Sample Rate**                 | N/A (normalized frequency)              |
| **Response**                    | Lowpass                                 |
| **Specification**               | N, Fp, Ap                               |
| **Filter Order**                | 3                                       |
| **Passband Edge**               | 0.4                                     |
| **Passband Ripple**             | 1 dB                                     |
| **Measurements**                |                                         |
| **Sample Rate**                 | N/A (normalized frequency)              |
| **Passband Edge**               | 0.4                                     |
| **3-dB Point**                  | 0.42779                                 |
| **6-dB Point**                  | 0.45816                                 |
| **Stopband Edge**               | Unknown                                 |
| **Passband Ripple**             | 1 dB                                     |

output. In communication system, receiving the same phase and frequency of the transmitted signal is very important. Therefore, the PLL is a device that must be founded in the system to recover the signal.
Figure 14. Structure of designed IIR filter.

Figure 15. Magnitude response of 3rd order IIR filter.
Figure 16. Group delay of IIR filter.

Figure 17. The pole/zero plot.
Figure 18. Received noisy signal before and after the filter.

Figure 19. Received signal on the Q-side at lock state.
Figure 20. Comparison between the generated and received samples.

Figure 21. The frequency spectrum of the received signal.
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