



# LPFFIR IP Core Specification

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## Revision History

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# Contents

<b>INTRODUCTION.....</b>	<b>1</b>
<b>SPECIFICATIONS.....</b>	<b>2</b>
FEATURES.....	2
<b>ARCHITECTURE.....</b>	<b>3</b>
<b>APPLICATION.....</b>	<b>5</b>
<b>IO PORTS.....</b>	<b>8</b>
<b>APPENDIX A .....</b>	<b>9</b>
IMPULSE RESPONSE .....	9
POLE ZERO PLOT .....	10
MAGNITUDE AND PHASE RESPONSE.....	11
STRUCTURE .....	12
<b>APPENDIX B .....</b>	<b>13</b>
<b>APPENDIX C .....</b>	<b>15</b>
FULL ADDER BOOLEAN ALGEBRA EXPRESSIONS .....	15
FULL ADDER SIMPLIFIED BOOLEAN ALGEBRA EXPRESSIONS.....	16
<b>INDEX.....</b>	<b>17</b>

## 1

# Introduction

Lowpass filter with finite impulse response (LPFFIR) IP core is characterized by one passband and one stopband, each specified by passband  $\omega_p$  edge frequency and stopband  $\omega_s$  edge frequency. The LPFFIR ideal filter  $H_i(e^{j\omega})$  gain is 6 in the passband and ideal attenuation in the stopband is zero, the filter design specifications include tolerance limits by which the ideal gains in the passband can be attenuated by  $\delta_p$  value and ideal stopband can be gained by  $\delta_s$  value. The LPFFIR tolerance scheme with edge frequencies and tolerance limits is shown in Figure 1.

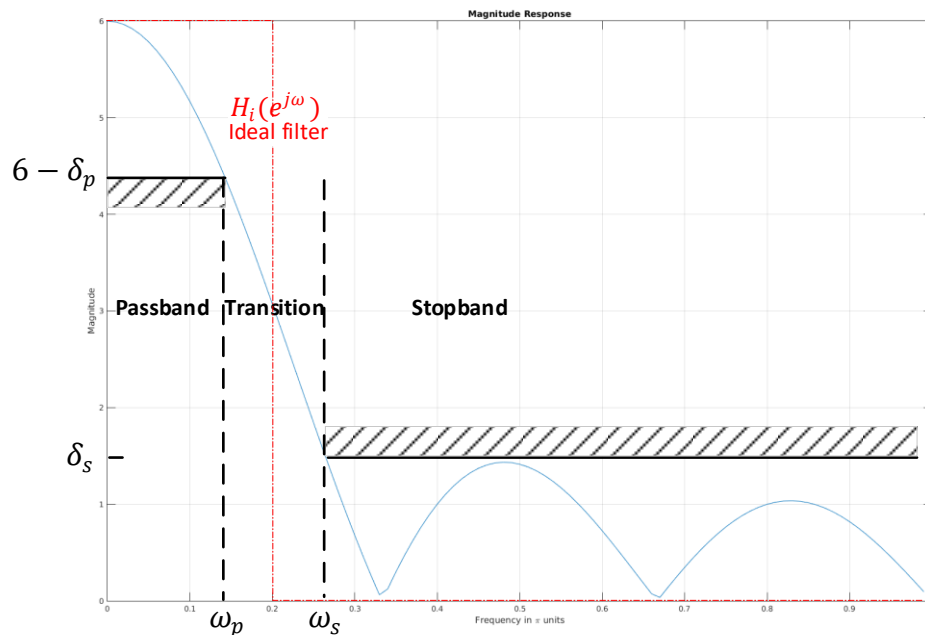


Figure 1 LPFFIR tolerance scheme.

# 2

## Specifications

The LPFFIR Figure 1 specifications are shown in Table 1

Table 1 LPFFIR specifications.

	Passband	Stopband
Ideal filter	6 gain	0 attenuation
Edge frequencies	$\omega_p = 0.14$	$\omega_s = 0.26$
Tolerance	$\delta_p = 1.56$	$\delta_s = 1.605$

### Features

- LPFFIR IP core RTL is fully synthesizable into logic gates targeting ASIC devices i.e. FPGA devices CLB slices or DSP units are not required for synthesis.
- High precision 16-bit fixed-point arithmetic is used for DSP RTL implementation.

# 3

## Architecture

The Figure 2 architecture is a realization of Figure 8 DSP structure which is made up of addition (+) and delay ( $Z^{-1}$ ) elements. The addition (+) element function is implemented by Full Adder (FA) module and Ripple Carry Adder (RCA) module with hierarchy of Figure 3. The delay ( $Z^{-1}$ ) element is implemented by Flip Flops (FF) in a series.

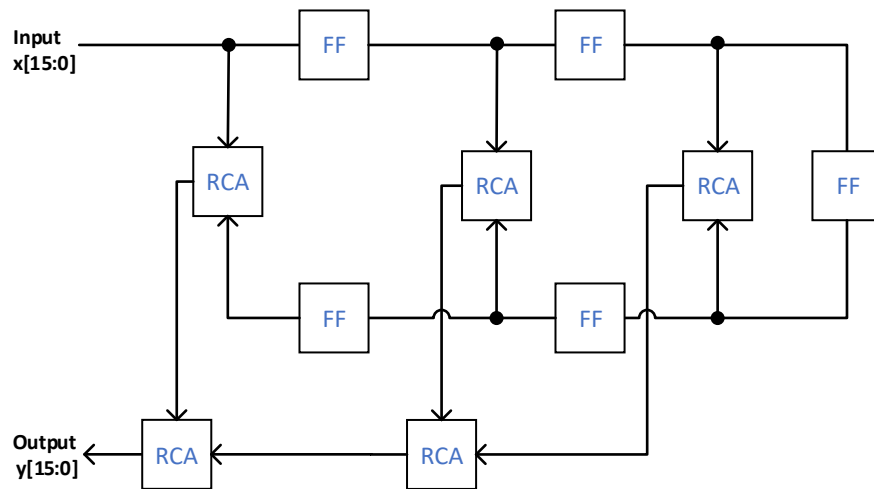


Figure 2 LPFFIR module block diagram.

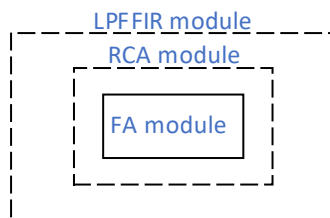
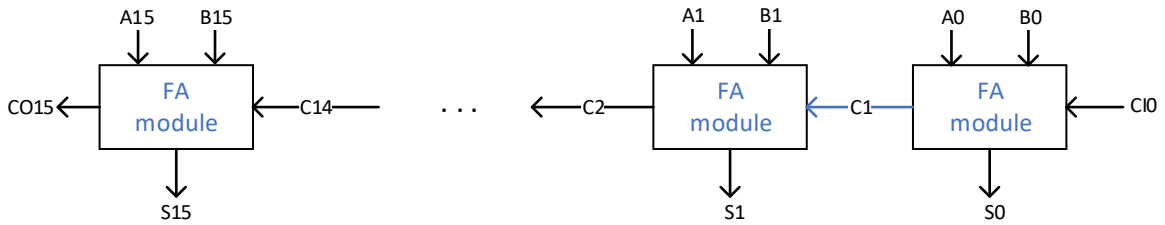


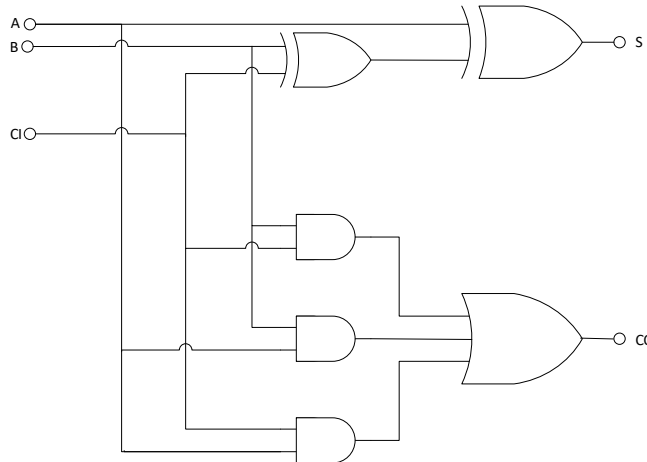
Figure 3 Module hierarchy block diagram.

The RCA module adds two 16-bit and one 1-bit binary number inputs A, B, and  $C_{in}(CI)$  respectively and outputs one 16-bit and one 1-bit binary numbers S and  $C_{out}(CO)$  respectively. The Figure 4 shows how multiple 1-bit add function FA modules are used to create 16-bit add function of RCA module.



**Figure 4 RCA module block diagram.**

The FA module adds three 1-bit binary number inputs A, B, and  $C_{in}(CI)$  and outputs two 1-bit binary numbers S and  $C_{out}(CO)$  as gate diagram shown in Figure 5 which is an implementation of [Full adder simplified Boolean algebra expressions].



**Figure 5 FA module gate diagram.**



# 4

## Application

Application example of LPFFIR IP core is Discrete-Time Processing of Continuous-Time Signals[1] with block diagram of Figure 6 and frequency-domain illustration of Figure 7, if the input is bandlimited and the sampling frequency is high enough to avoid aliasing, then the overall system behaves as an LTI continuous-time system with the output is related to the input through an equation of the form

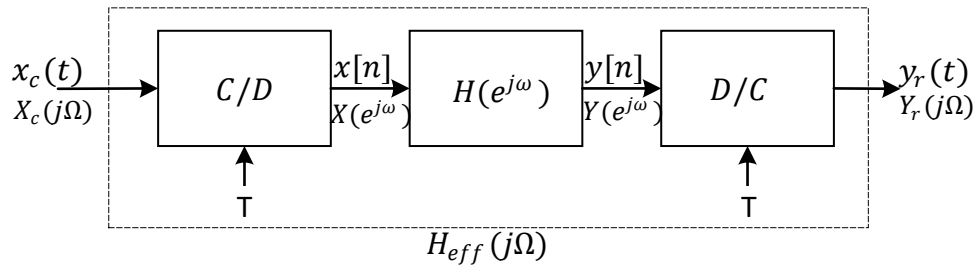
$$Y_r(j\Omega) = H_{eff}(j\Omega)X_c(j\Omega)$$

where effective continuous-time frequency responds

$$H_{eff}(j\Omega) = \begin{cases} H(e^{j\Omega T}), & |\Omega| < \pi/T \\ 0, & |\Omega| \geq \pi/T \end{cases}$$

Using  $\omega = \Omega T$  relation to convert from effective continuous-time filter specification to the discrete-time filter specification results an equation of the form

$$H(e^{j\omega}) = H_{eff}\left(j\frac{\omega}{T}\right), \quad |\omega| < \pi.$$



**Figure 6 Discrete-time filtering of continuous-time signals system application.**

The  $|H_{eff}(j\Omega)|$  continuous-time overall system of Figure 6 with following requirements

1. Sample period shall be  $T = 10^{-4}s$
2. The passband gain shall be 6.
3. The attenuated tolerance at the passband shall be 1.56 in the frequency band  $0 \leq \Omega \leq 2\pi(1400)$  .
4. The gain tolerance at the stopband shall be 1.605 in the frequency band  $2\pi(2600) \leq \Omega$ .

The mapping between the continuous-time and discrete-time frequencies only affects the passband and stopband edge frequencies and not the tolerance limits on frequency response magnitude [2].

The  $|H(e^{j\omega})|$  discrete-time block of Figure 6 with following requirements

1. The passband gain shall be shall be 6.
2. The attenuated tolerance at the passband shall be 1.56 in the frequency band  $0 \leq \omega \leq 0.14\pi$  .
3. The gain tolerance at the stopband shall be 1.605 in the frequency band  $0.26\pi \leq \omega$ .

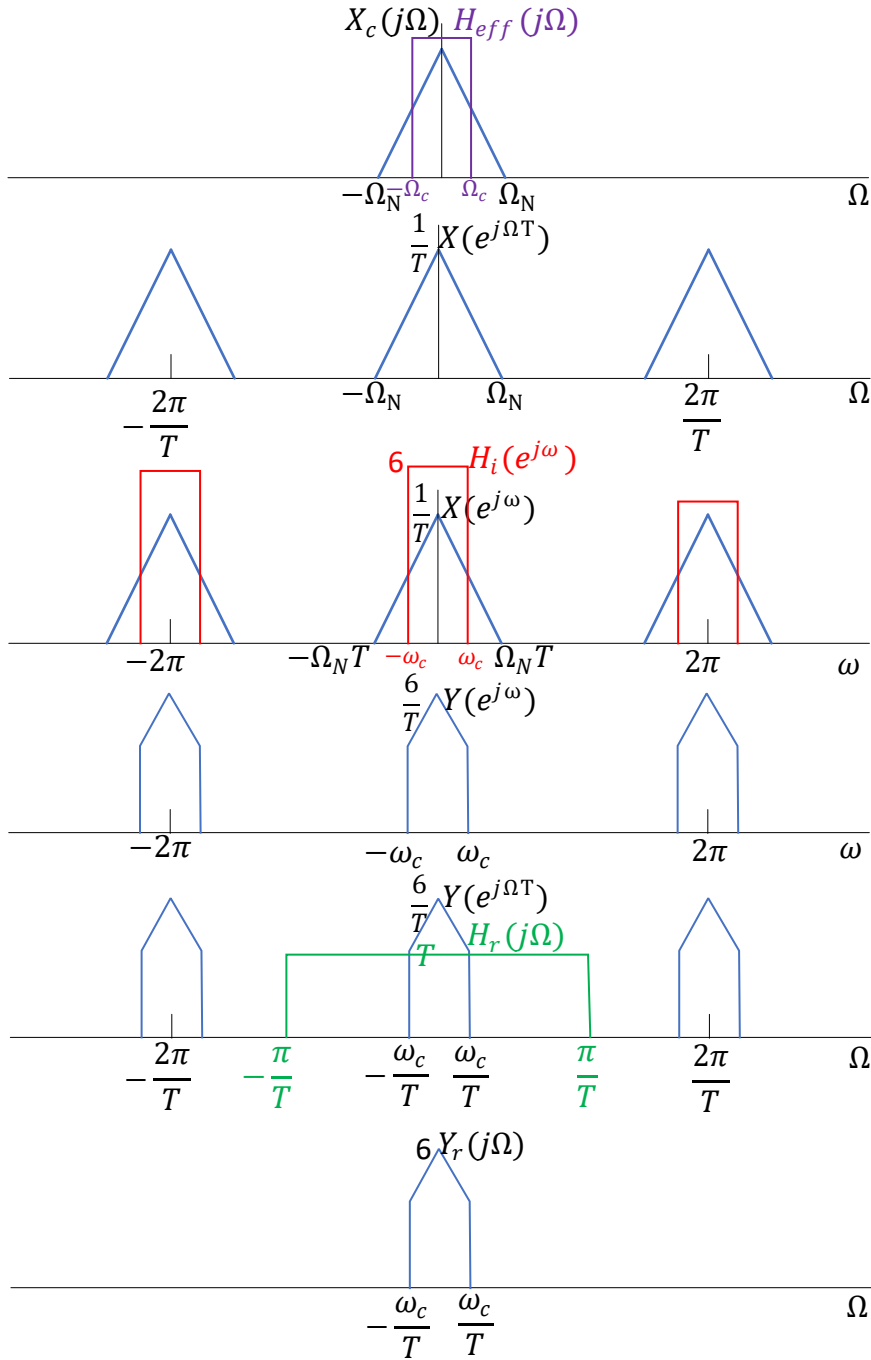


Figure 7 Frequency-domain illustration of discrete-time filtering of continuous-time signals.

# 5

## IO Ports

Port	Width	Direction	Description
clk_i	1	Input	Clock Input
x_i	16	Input	Low-pass filter Input
y_o	16	Output	Low-pass filter Output

**Table 2: List of IO ports.**

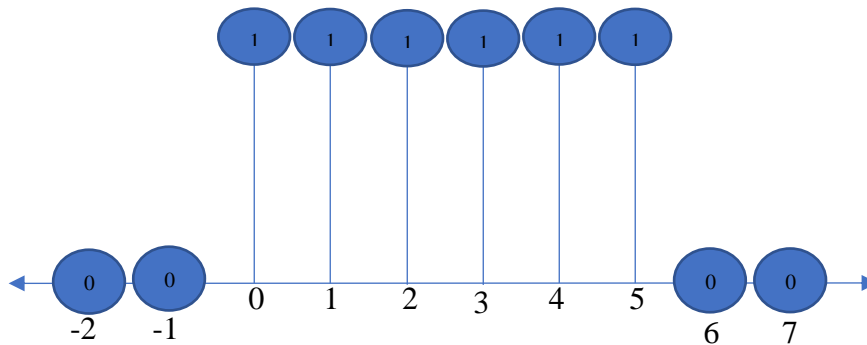
# Appendix A

## Structure

The LPFFIR uses a direct form structure for a FIR linear-phase system. The DSP theory [3] is used for derivation and structure is shown in Figure 8.

### Impulse Response

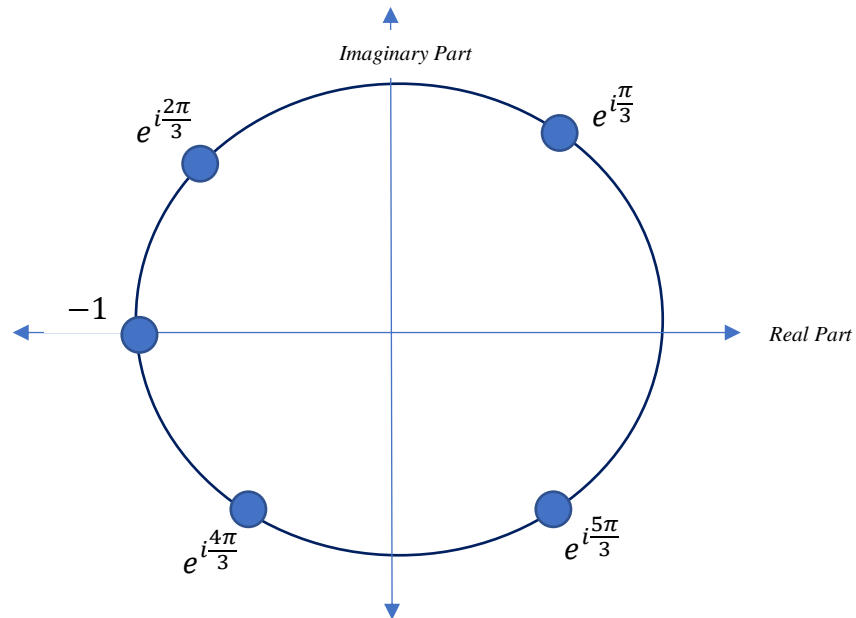
$$h[n] = \begin{cases} 1, & 0 \leq n \leq 5 \\ 0, & \text{otherwise} \end{cases}$$



## Pole Zero Plot

$$H(z) = 1 + z^{-1} + z^{-2} + z^{-3} + z^{-4} + z^{-5}$$

$$= (1 - e^{i\frac{\pi}{3}}z^{-1})(1 - e^{i\frac{2\pi}{3}}z^{-1})(1 - e^{i\pi}z^{-1})(1 - e^{i\frac{4\pi}{3}}z^{-1})(1 - e^{i\frac{5\pi}{3}}z^{-1})$$



## Magnitude and Phase Response

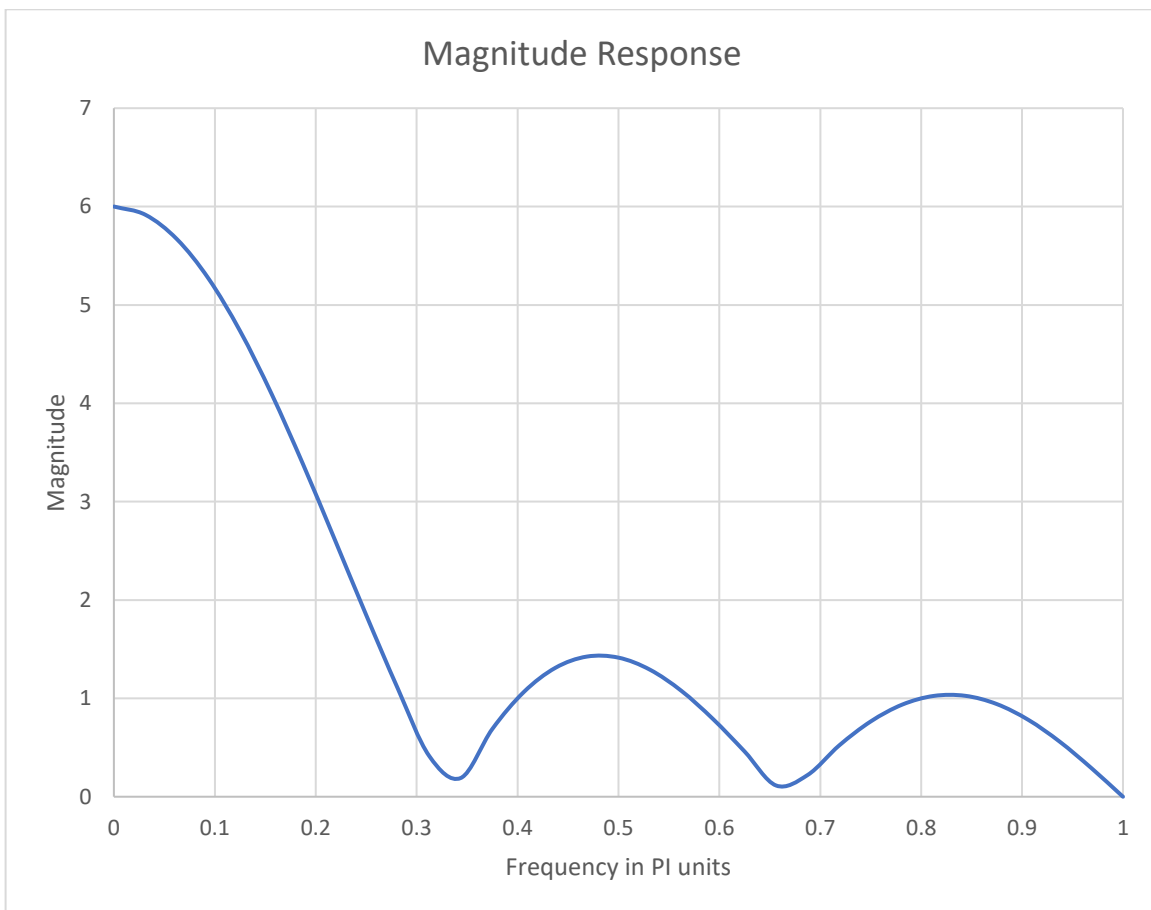
$$H(z = e^{i\omega}) = 1 + z^{-1} + z^{-2} + z^{-3} + z^{-4} + z^{-5}$$

$$= 1 + e^{-i\omega} + e^{-2i\omega} + e^{-3i\omega} + e^{-4i\omega} + e^{-5i\omega}$$

$$= e^{-i\omega\frac{5}{2}}(e^{i\omega\frac{5}{2}} + e^{i\omega\frac{3}{2}} + e^{i\omega\frac{1}{2}} + e^{-i\omega\frac{1}{2}} + e^{-i\omega\frac{3}{2}} + e^{-i\omega\frac{5}{2}})$$

$$= e^{-i\omega\frac{5}{2}}(2\cos\frac{5}{2}\omega + 2\cos\frac{3}{2}\omega + 2\cos\frac{1}{2}\omega)$$

$$\therefore |H(z = e^{i\omega})| = 2 \left| \cos\frac{5}{2}\omega + \cos\frac{3}{2}\omega + \cos\frac{1}{2}\omega \right| \text{ and } \angle H(z = e^{i\omega}) = -\frac{5}{2}\omega$$



## Structure

$$\begin{aligned}
 y[n] &= h[n] * x[n] \\
 &= \sum_{k=0}^M h[k]x[n - k] \\
 &= \sum_{k=0}^{\frac{M-1}{2}-1} h[k]x[n - k] + \sum_{k=\frac{M-1}{2}+1}^M h[k]x[n - k] \\
 &= \sum_{k=0}^{\frac{M-1}{2}-1} h[k]x[n - k] + \sum_{k=0}^{\frac{M-1}{2}+1} h[M - k]x[n - M + k] \\
 &= \sum_{k=0}^{\frac{M-1}{2}} h[k](x[n - k] + x[n - M + k])
 \end{aligned}$$

Let:  $h[n] = \begin{cases} 1, & 0 \leq n \leq 5 \\ 0, & \text{otherwise} \end{cases} \Rightarrow M = 5$  Note:  $M$  is an odd integer.

$$\begin{aligned}
 &= \sum_{k=0}^2 h[k](x[n - k] + x[n - 5 + k]) \\
 &= h[0](x[n] + x[n - 5]) + h[1](x[n - 1] + x[n - 4]) + h[2](x[n - 2] + x[n - 3])
 \end{aligned}$$

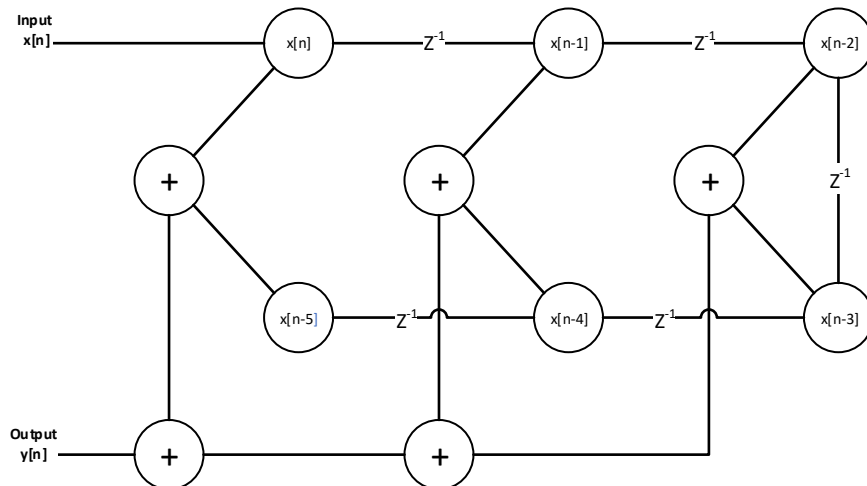


Figure 8 Direct form structure for a FIR linear-phase system.



# Appendix B

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## Expected Behavior

The LPFFIR expected behavior is generated from MATLAB simulation. The simulation source code and result plot are shown in Figure 9 and Figure 10 respectively.

```
1. % FIR difference equation of lowpass filter
2. b = [1, 1, 1, 1, 1, 1]; a = [1];

3. % Response
4. n = [0:7];
5. h = impz(b,a,8);
6. [H,w] = freqz(b,a,100);
7. magH = abs(H); phaH = angle(H);

8. % Plot
9. subplot(4,1,1); stem(n,h);
10. title('Impulse Response'); xlabel('n'); ylabel('h(n)')

11. subplot(4,1,2);zplane(b,a);grid
12. title('Pole-Zero Plot')

13. subplot(4,1,3);plot(w/pi,magH);grid
14. xlabel('Frequency in \pi units'); ylabel('Magnitude');
15. title('Magnitude Response')

16. subplot(4,1,4);plot(w/pi,phaH/pi);grid
17. xlabel('Frequency in \pi units'); ylabel('Phase in \pi units');
18. title('Phase Response')
```

Figure 9 MATLAB simulation source code.

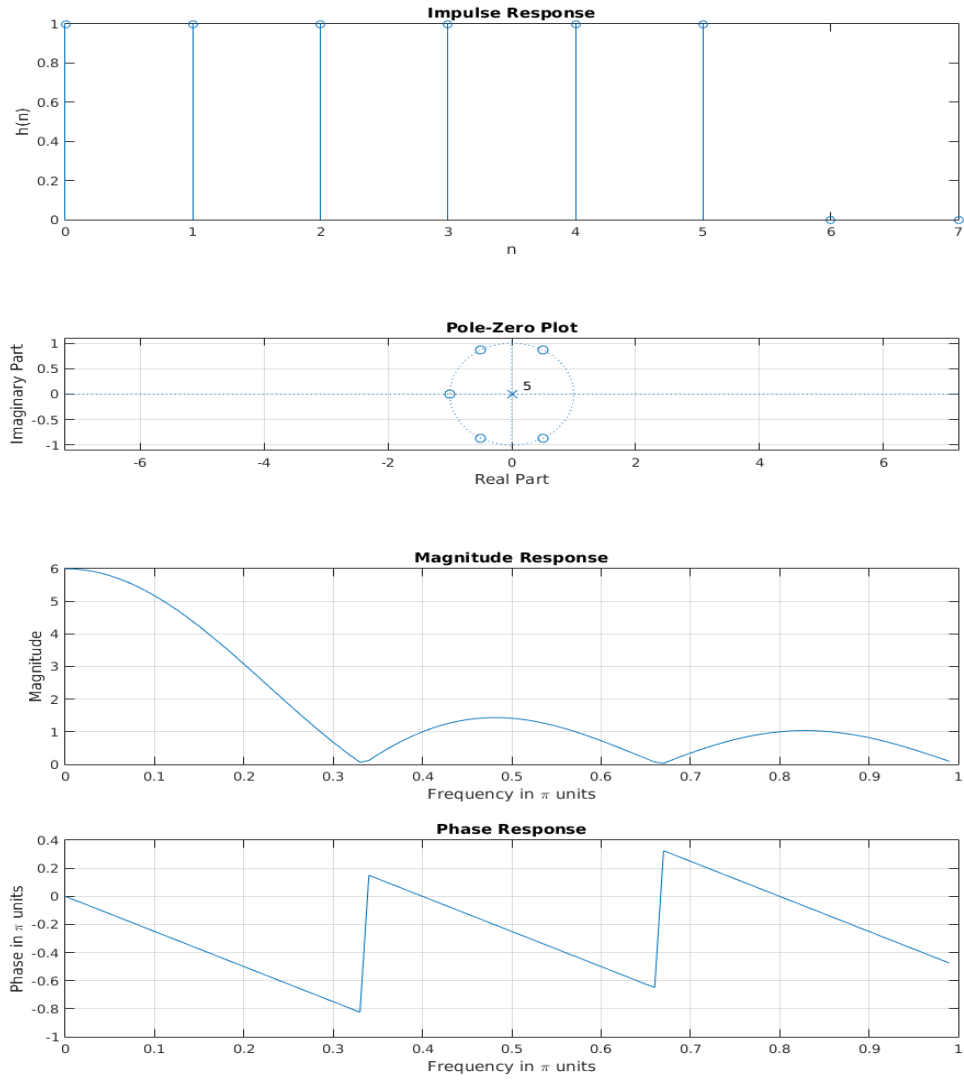
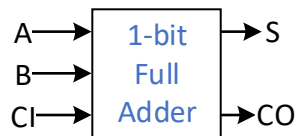


Figure 10 MATLAB simulation result plot.

# Appendix C

## Full adder Boolean algebra expressions



- $CO = B \cdot CI + A \cdot CI + A \cdot B + A \cdot B \cdot CI$
- $S = \bar{A} \cdot \bar{B} \cdot CI + \bar{A} \cdot B \cdot \bar{CI} + A \cdot \bar{B} \cdot \bar{CI} + A \cdot B \cdot CI$

The full adder Boolean expressions are derived from truth Table 3.

Table 3 Full adder truth table.

Input			Output	
A	B	CI	CO	S
0	0	0	0	0
0	0	1	0	1
0	1	0	0	1
0	1	1	1	0
1	0	0	0	1
1	0	1	1	0
1	1	0	1	0
1	1	1	1	1

## Full adder simplified Boolean algebra expressions

- $CO = B \cdot CI + A \cdot CI + A \cdot B$
- $S = A \oplus B \oplus CI$

The K-map of Table 4 and Table 5 are used for simplifying Boolean algebra expressions of full adder.

Table 4 Full adder K-map of CO.

	$\bar{A} \cdot \bar{B}$	$\bar{A} \cdot B$	$A \cdot B$	$A \cdot \bar{B}$
$\bar{CI}$	0	0	1	0
$C$	0	1	1	1

Table 5 Full adder K-map of S.

	$\bar{A} \cdot \bar{B}$	$\bar{A} \cdot B$	$A \cdot B$	$A \cdot \bar{B}$
$\bar{CI}$	0	1	0	1
$C$	1	0	1	0

# Index

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1. Oppenheim, A. V., & Ronald, W. S. (2009). Discrete-Time Processing of Continuous-Time Signals. In *Discrete-Time Signal Processing 3rd Edition* (pp. 197-172). Upper Saddle River, NJ: Pearson
2. Oppenheim, A. V., & Ronald, W. S. (2009). Filter Specifications. In *Discrete-Time Signal Processing 3rd Edition* (pp. 494-496). Upper Saddle River, NJ: Pearson
3. Oppenheim, A. V., & Ronald, W. S. (2009). Structures for Linear-Phase FIR Systems. In *Discrete-Time Signal Processing 3rd Edition* (pp. 403-405). Upper Saddle River, NJ: Pearson.