A Study of the Impact of Hardware Design Choices on the System Impulse Response of a Signal-level Radar Simulation

A thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Engineering

by

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ABSTRACT


The main goal of this research is to study the effect of hardware design choices at the signal-level of an end-to-end radar system. An implementation of waveform diversity concepts, or the use of various waveforms in both transmit and receive is employed. An end-to-end Matlab simulation is developed such that the system impulse response to hardware imperfections and waveform parameters such as bandwidth and frequency channel spacing can be analyzed. Multiple frequency channels as well as multiple pulses are also considered in the research.

All hardware components are nonlinear to some degree. The nonlinearities of the hardware give rise to unwanted spectral components in the output. Therefore, models that simulate the behavior of both ideal and non-ideal hardware components are developed. A user friendly interface is developed in which each hardware component can be interchanged between the ideal and non-ideal model. In studying both the system impulse response to ideal components as well as the response to non-ideal components, the impact of the nonlinearities on cross-correlation signal detection and range-Doppler can be analyzed.
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Chapter 1

Introduction

1.1 Background Information

The main objective of this research is to study the impact of current hardware in the implementation of waveform diversity concepts. The use of various waveforms, or signals in both transmit and receive radar design for improving the overall system performance is referred to as waveform diversity [10]. Waveform diversity can encompass many aspects of the signal design problem, including frequency division multiplexing, pseudo-random phase coding, and pulse compression chirp rate diversity [11]. A couple system performance characteristics addressed in this waveform diversity study are reduction in side lobe levels of both the cross-correlation detection plots as well as in the range-Doppler response.

An end-to-end radar system simulated at the signal-level is desired such that the effect of hardware design choices can be analyzed. All choices of hardware components contribute in some way to the performance of the overall system. The effect of components such as amplifiers, digital-to-analog converters, mixers, filters, and analog-to-digital converters on a signal-level radar system is the main focus of this research. The signal-level simulation developed and used throughout the course of this research employs the use of binary phase-coded waveforms. The impact of the same hardware choices using other waveforms of interest such as complementary codes, though beyond the scope of this research, could certainly be studied in the future. The development of this signal-level simulation will aid in the future study of other such waveform diversity concepts.
In determining the impact of such hardware choices, multiple frequency channels as well as multiple pulses are considered. A requirement of the study is the capability to control the transmit waveform on each channel for each pulse in a processing interval. The ability to change the timing of the waveforms on each channel as well as for each pulse is also required. Also, the effect of such waveform parameters as bandwidth and frequency channel spacing will be considered.

A user friendly interface in Matlab that gives the ability to interchange hardware components is a secondary goal. The interface should allow the user to implement linear components as well as those with nonlinearities. Interchanging of component types as well as nonlinear parameters should be of ease to the user. The interface should also have the ability to determine the nonlinearity of the components through use of cross-correlation signal detection and range-Doppler response.

1.2 Technical Approach

A signal-level Matlab simulation of an end-to-end radar system is essential such that the impact of hardware design choices on the performance of waveform diversity techniques can be determined. An ideal simulation is needed such that when non-ideal components with nonlinear effects are interchanged with the ideal components, a performance comparison can be made. Ideal and non-ideal models for each functional block in the system architecture are developed. The models simulate the digital and analog performance of each step of the signal processing transmit and receive chains. The information used to construct the models is based on typical hardware specifications.

1.2.1 Simulation Environment

The end-to-end signal-level simulation is implemented in Matlab such that the component models are easy to manipulate and modify for future use. To make things even simpler, a separate module for each component model is generated. The general architecture of the transmit and receive chains chosen for the initial simulation development are shown in Figures 1.1 and 1.2. These figures do not represent the actual hardware components, but instead the signal flow through the system. Each of the actual hardware components can be modeled by choosing appropriate parameters based on the component specifications.
The general transmit signal flow is shown in Figure 1.1. The waveform generation block allows control over the waveform selection as well as the time of the pulses. The output waveforms from the generator block are digital. The waveforms are then digitally modulated into separate frequency channels and combined. The summed signal is passed through a digital-to-analog converter. The nonlinear DAC adds unwanted harmonics to the output signal and is therefore followed by an analog low-pass filter to remove these harmonics. The signal is then up-converted to the RF carrier frequency, and the output of the mixer is high-pass filtered to remove the low frequency signal. The signal is then amplified for transmission.

The emitted signal interacts with the target in the time delay network block. The time delay network block allows control over target parameters such as target range, range swath, and target velocity and may in the future include array processing effects. The target returns are captured by the receive architecture shown in Figure 1.2. The receive signal is amplified, down-converted from the RF frequency to the intermediate frequencies, and filtered. The signal is passed through an analog-to-digital converter. Finally, the signal is demodulated to the individual frequency channels and passed to matched filter blocks for detection and analysis.
1.2.2 Mathematical Background

The end-to-end signal-level simulation is first derived mathematically. The derivation proceeds through both the transmit and receive chains, analytically explaining the inputs and outputs of major blocks in the simulation. Figure 1.3 shows the mathematical architecture upon which the derivation is based. The transmitted signal can be an arbitrary waveform. The number of pulses is calculated by use of the pulse repetition frequency, $F_p$, and the integration time, $T_{coh}$, according to

$$N_{\text{pulses}} = T_{coh} F_p.$$ (1.1)

Once the waveform is generated, it is digitally modulated to the IF frequency. This waveform generation is repeated depending on the number of frequency channels used in the simulation.

A complex signal for each branch is generated. That is, there consists of an I-channel and a Q-channel, the sum of which is the transmitted complex signal, $S_Z(t)$ shown in

$$S_Z(t) = S_I(t) + jS_Q(t).$$ (1.2)

For both channels, a binary waveform is generated and digitally upsamples. Once upsampled, the next step in the derivation is the direct quadrature modulation block which modulates the complex signal to the intermediate frequency. Since the signal is complex, the I-channel is modulated by a cosine wave at the IF frequency while the Q-channel is modulated by a sine wave at the same frequency. The resulting signal is called $S_{IF}(t)$. The following equation

$$S_{IF}(t) = S_I(t) \cos(2\pi f_{IF}t) + S_Q(t) \sin(2\pi f_{IF}t)$$ (1.3)

shows this complex quadrature modulation.

Once the signal is digitally modulated to IF, it is converted to the analog domain by use of a digital-to-analog converter (DAC). If the non-ideal DAC is used, it generates unwanted harmonics.
in the output. In this case, a low pass filter is needed to remove the high frequency harmonic components of the DAC output. However, if an ideal DAC is assumed, there is no need for the low pass filter following the converter.

Now that the signal is in the analog domain, the next step in the derivation is the RF modulation. The IF modulated signal, $S_{IF}(t)$ is modulated to the RF frequency. The resulting signal is appropriately called $S_{RF}(t)$. In order that the signal is modulated to the chosen RF frequency, a cosine modulation of $S_{IF}(t)$ at a frequency of $(f_{RF} - f_{IF})$ is used. Equation 1.4 explains the analog RF modulation mathematically.

$$S_{RF}(t) = S_{IF}(t) \cos(2\pi(f_{RF} - f_{IF})t)$$

$$= [S_I(t) \cos(2\pi f_{IF}t) + S_Q(t) \sin(2\pi f_{IF}t)] \cos(2\pi(f_{RF} - f_{IF})t)$$

$$= S_I(t) \cos(2\pi f_{IF}t) \cos(2\pi(f_{RF} - f_{IF})t)$$

$$+ S_Q(t) \sin(2\pi f_{IF}t) \cos(2\pi(f_{RF} - f_{IF})t)$$

$$= S_I(t)(\cos(2\pi f_{RF}t) + \cos(2\pi(2f_{IF} - f_{RF})t))$$

$$+ S_Q(t)(\sin(2\pi f_{RF}t) + \sin(2\pi(2f_{IF} - f_{RF})t))$$

(1.4)

Once the modulation to the RF band is complete, the signal is high pass filtered in order to keep the part of the signal at the RF frequency and remove the low frequency components of the signal. Equation 1.5 illustrates the removal of the components at $(2f_{IF} - f_{RF})$ but allows the components at $f_{RF}$ to pass through.

$$S_{RF}(t) = \text{HPF}\{S_I(t)(\cos(2\pi f_{RF}t) + \cos(2\pi(2f_{IF} - f_{RF})t))$$

$$+ S_Q(t)(\sin(2\pi f_{RF}t) + \sin(2\pi(2f_{IF} - f_{RF})t))\}$$

$$S_{RF}(t) = S_I(t) \cos(2\pi f_{RF}t) + S_Q(t) \sin(2\pi f_{RF}t)$$

(1.5)

Once the signal is at the RF band and the correct frequency components are removed, the signal is propagated through the atmosphere to the target. The signal hits the target and is returned with a time delay, $t_0$. The calculation of the time delay is based upon target range and the speed of light. The time delay calculation is explained in more detail in Section 2.2.1 of this document. The target return, $S_{Rx}(t)$ is the RF signal, $S_{RF}(t)$ delayed by the time delay, represented symbolically by equation 1.6.
\[ S_{Rx}(t) = S_{RF}(t - t_0) \]
\[ = S_I(t - t_0) \cos(2\pi f_{RF}T) + S_Q(t - t_0) \sin(2\pi f_{RF}(t - t_0)) \tag{1.6} \]

On receive, the delayed signal is demodulated from the RF frequency. Since the analog modulation to the RF frequency on transmit required a cosine modulation to a frequency of \((f_{RF} - f_{IF})\), the received signal is demodulated by the same frequency to insure that the signal \(S_{Rx}(t)\) is at the correct IF frequency. Equation 1.7 shows this analog demodulation process mathematically. The equation also shows that the signal contains frequency components at both high and low frequencies.

\[ S_{IF_{Rx}}(t) = S_{Rx}(t) \cos(2\pi (f_{RF} - f_{IF})t) \]
\[ = S_I(t - t_0) \cos(2\pi f_{RF}(t - t_0)) \cos(2\pi (f_{RF} - f_{IF})t) \]
\[ + S_Q(t - t_0) \sin(2\pi f_{RF}) \cos(2\pi (f_{RF} - f_{IF})t) \]
\[ = S_I(t - t_0) [\cos(2\pi f_{IF} - 2\pi f_{RF}t_0) + \cos(2\pi (2f_{RF} - f_{IF}) - 2\pi f_{RF}t_0)] \]
\[ + S_Q(t - t_0) [\sin(2\pi (2f_{RF} - f_{IF}) - 2\pi f_{RF}t_0) \sin(2\pi f_{IF}t - 2\pi f_{RF}t_0)] \tag{1.7} \]

Therefore, a low pass filter is needed such that the portion of the signal spectrum at \((2f_{RF} - f_{IF})\) is filtered out while the frequency components at \(f_{IF}\) remain in the spectrum. Equation 1.8 shows the filtering process of the received signal. Once filtered, the analog received signal is converted back to the digital domain by means of an analog-to-digital converter.

\[ S_{IF_{Rx}}(t) = LPF\{S_I(t - t_0)[\cos(2\pi f_{IF} - 2\pi f_{RF}t_0) + \cos(2\pi (2f_{RF} - f_{IF}) - 2\pi f_{RF}t_0)] \}
\[ + S_Q(t - t_0)[\sin(2\pi (2f_{RF} - f_{IF}) - 2\pi f_{RF}t_0) \sin(2\pi f_{IF}t - 2\pi f_{RF}t_0)] \}
\[ = S_I(t - t_0) \cos(2\pi f_{IF}t - 2\pi f_{RF}t_0) + S_Q(t - t_0) \sin(2\pi f_{IF}t - 2\pi f_{RF}t_0) \tag{1.8} \]

Figure 1.3 shows that the received time delayed signal must be digitally demodulated into its corresponding complex signals at the IF band. Again, the complex signal is a sum of the I- and the Q-channel. The I-channel is demodulated to the IF frequency using a cosine wave while the Q-channel is demodulated using a sine wave. Equations 1.9 and 1.10 mathematically illustrate this digital demodulation process for both channels.
I channel:

\[ S_{IRx}(t) = S_{IFRx}(t) \cos(2\pi f_{IF} t) \]

\[ = S_I(t - t_0)\cos(2\pi f_{IF} t - 2\pi f_{RF} t_0)\cos(2\pi f_{IF} t) \]

\[ + S_Q(t - t_0)\sin(2\pi f_{IF} t - 2\pi f_{RF} t_0)\sin(2\pi f_{IF} t) \]

\[ = S_I(t - t_0)[\cos(2\pi f_{RF} t_0) + \cos(2\pi f_{IF} t - 2\pi f_{RF} t_0)] \]

\[ + S_Q(t - t_0)[\sin(2\pi f_{IF} t - 2\pi f_{RF} t_0) - \sin(2\pi f_{RF} t_0)] \]

(1.9)

Q channel:

\[ S_{QRx}(t) = S_{IFRx}(t) \sin(2\pi f_{IF} t) \]

\[ = S_I(t - t_0)\cos(2\pi f_{IF} t - 2\pi f_{RF} t_0)\sin(2\pi f_{IF} t) \]

\[ + S_Q(t - t_0)\sin(2\pi f_{IF} t - 2\pi f_{RF} t_0)\sin(2\pi f_{IF} t) \]

\[ = S_I(t - t_0)[\sin(2\pi f_{IF} t - 2\pi f_{RF} t_0) + \sin(2\pi f_{RF} t_0)] \]

\[ + S_Q(t - t_0)[\cos(2\pi f_{RF} t_0) + \cos(2\pi f_{IF} t - 2\pi f_{RF} t_0)] \]

(1.10)

Once the signals are demodulated accordingly, both the I- and Q-channel signals are low pass filtered in order to remove the spectrum of each channel at 2f_{IF}. After filtering out the high frequency components, the signal is at baseband. Equations 1.11 and 1.12 show the I- and Q-channel demodulation and the resulting digital signals. The I-channel signal, S_{IRx}(t) is a difference of the original transmitted I- and Q-channel signals, both multiplied by the time delay t_0. The Q-channel, S_{QRx}(t) is a sum of the original transmitted signals also multiplied by the same time delay.

I channel:

\[ S_{IRx}(t) = LPF \{ S_I(t - t_0)[\cos(2\pi f_{RF} t_0) + \cos(2\pi f_{IF} t - 2\pi f_{RF} t_0)] \}

\[ + S_Q(t - t_0)[\sin(2\pi f_{RF} t_0) - \sin(2\pi f_{IF} t - 2\pi f_{RF} t_0)] \} \]

\[ = S_I(t - t_0)\cos(2\pi f_{RF} t_0) - S_Q(t - t_0)\sin(2\pi f_{RF} t_0) \]

(1.11)

Q channel:

\[ S_{QRx}(t) = LPF \{ S_I(t - t_0)[\sin(2\pi f_{IF} t - 2\pi f_{RF} t_0) + \sin(2\pi f_{RF} t_0)] \}

\[ + S_Q(t - t_0)[\cos(2\pi f_{RF} t_0) + \cos(2\pi f_{IF} t - 2\pi f_{RF} t_0)] \} \]

\[ = S_I(t - t_0)\sin(2\pi f_{RF} t_0) + S_Q(t - t_0)\cos(2\pi f_{RF} t_0) \]

(1.12)
Therefore, the resulting received signal is

$$S_{Z_{Rx}}(t) = S_Z(t - t_0)e^{j2\pi f_{RF}t_0}.$$  \hspace{1cm} (1.13)

### 1.3 Matlab Simulation

#### 1.3.1 Transmit Signal Flow

The end-to-end radar simulation in Matlab starts by calculating the number of pulses needed with the given coherent integration time and pulse repetition frequency (PRF). The Doppler ambiguity is based upon the choice of the PRF. The pulse repetition frequency must be greater than the maximum anticipated Doppler frequency shift, or aliasing of the signal will occur and the Doppler measurement will become ambiguous. The governing equation used to choose the correct PRF is

$$F_p > \max(f_d) = \frac{2v_{max}}{\lambda},$$ \hspace{1cm} (1.14)

where $v_{max}$ is the maximum target velocity and $\lambda$ is the wavelength.

The target velocity resolution, $\Delta v$ is chosen to be $1\frac{m}{s}$ throughout the simulation. Knowing the signal wavelength, $\lambda$ and the chosen target velocity resolution, the coherent integration time, $T_{coh}$ can be calculated using

$$\Delta f_d = \frac{1}{T_{coh}} = \frac{2\Delta v}{\lambda}.$$ \hspace{1cm} (1.15)

The Doppler resolution, $\Delta v$ can also be calculated based upon the coherent integration time using the same equation. Again, Equation 1.1 calculates the number of pulses needed in the simulation based on the user defined values of the PRF and the integration time.

The next step in the simulation is to loop through the pulses and generate a waveform for each branch. The entire end-to-end simulation, aside from the range-Doppler plot, is run once for each pulse. The time of each pulse is governed by the pulse repetition frequency. The following equation,

$$\tau_k = k\frac{1}{PRF},$$ \hspace{1cm} (1.16)

calculates the time of each pulse based upon $k$, which is the pulse number. This pulse time is used in the time delay network to calculate the target range if the target is nonstationary. A digital binary waveform of a user defined length for each branch is generated. The waveform is quantized and
modulated up to the corresponding IF frequency of the respective branch with a digital mixer. The output signal of the digital mixer is the same as the input signal in amplitude, but not in frequency. The input signal was at baseband, while the output of the mixer is at the specified IF frequency. Once the signals are modulated to IF, the signals from each branch are summed. The next step in the simulation is the digital-to-analog converter. A DAC is, as its name suggests, a component used to convert a signal in the digital domain to its analog equivalent. The digital signal passes through the DAC, and the output signal is analog.

After the summed signal is converted to the analog domain, it is modulated up to the RF frequency from IF by an analog mixer. The frequency spectrum of the output signal is shifted from IF to the specified RF frequency. Once this modulation takes place, the signal must be high-pass filtered. The output signal of the mixer has peaks at \((2f_{IF} - f_{RF})\) and \(f_{RF}\). The peaks at the smaller frequency must be filtered out. Therefore, an analog filter is used to filter out the lower frequency signal. Finally, this signal is amplified for transmission.

### 1.3.2 Receive Signal Flow

The transmitted signal then interacts with the environment and the target. The signal hits the target and the return signal is captured. The target is simulated as a point with a scalar amplitude and can be either moving or stationary. The receive chain is simply the reverse of the transmit chain. Once the target return is captured, it is amplified again for detection.

The target return signal is still at the RF frequency. A frequency demodulation is done such that the signal is no longer at the RF frequency but now at the IF frequency. The demodulation from RF is performed in the same fashion as the modulation to the RF frequency. The analog mixer is used for this demodulation. The only difference between this demodulation from RF and the modulation to RF on transmit is the filter following the mixer. On transmit, the high frequency components of the signal were needed, but on receive, only the low frequency components are needed. Therefore, the signal is low-pass filtered and only the components of the signal at \(f_{IF}\) are passed through the filter while components at \((2f_{RF} - f_{IF})\) are filtered out.

The signal must now pass through an analog-to-digital converter. Just as the DAC converted the signal from the digital domain to the analog domain, the ADC converts the signal back from the
analog to the digital domain. The analog signal passes through the ADC and the output signal is digital.

A second demodulation on receive is needed to modulate the signal from the IF frequency to baseband. For each branch of the simulation, the digital output of the ADC is passed through a digital mixer with a carrier frequency equal to that of the modulation of that same branch. This modulation puts the receive signal for each branch back at baseband from the IF frequency. The frequency spectrum of the output of the digital mixer has two peaks, one at baseband and one at $2f_{IF}$. A low-pass filter is needed to allow the signal to pass through at baseband but filter out the peaks at the higher of the two frequencies, $2f_{IF}$. The output signal of the low-pass filter is the return signal used in the detection process.

### 1.3.3 Detection Process

The signal detection process consists of a matched filter for each branch of each pulse, and range-Doppler plots for each branch after the entire pulse train is sent and detected. The matched filter is the correlation of a known signal with an unknown signal. In this case the transmitted signal is known while the unknown signal is the received signal. The plot of the cross-correlation of these two signals will peak at the time delay of the returned signal. Because the transmitted signal was complex, that is, it consisted of an I- and a Q-channel for each branch, the cross-correlation is also complex.

A range-Doppler plot for each branch is generated. The range-Doppler values are generated simply by taking the Fourier Transform across the compressed pulses for each branch. The Doppler frequency is calculated using

$$f_d = \frac{2v_t}{\lambda}$$

(1.17)

where $f_d$ is the Doppler frequency shift, $v_t$ is the target velocity, and $\lambda$ is the wavelength of the signal. The range-Doppler plot should peak at the corresponding range and Doppler shift.

The 3dB main lobe width, 15dB main lobe width, peak to side lobe ratio, and integrated side lobe ratio are calculated. The 3dB resolution is measured at 3dB down from the peak of the main lobe. The 15dB resolution is measured similarly. The peak to side lobe ratio (PSLR) is defined as the ratio between the peak of the main lobe to that of the first side lobe. The PSLR is calculated
mathematically using

\[ PSLR = 20 \log_{10} \left| \frac{\text{max}(\text{sidelobe peak})}{\text{mainlobe peak}} \right|. \]  

(1.18)

The integrated side lobe ratio (ISLR) is the ratio between the peak power of the main lobe and the integrated power of several side lobes on both sides of the main lobe [4]. The following equation,

\[ ISLR = 10 \log_{10} \left| \frac{\sum |\text{sidelobe values}|}{\sum |\text{mainlobe values}|} \right|^2 \]  

(1.19)

mathematically demonstrates the calculation of the ISLR.

### 1.3.4 Matlab GUI

The Matlab simulation of the radar system is constructed by the use of separate modules for each component. A graphical user interface was also developed such that the parameters for each component are easily accessible and easily changed. Figure 1.4 is a screen snapshot of the graphical user interface. All buttons allow for the modification of certain parameters depending on which button is chosen. Each button is labeled with the name of the component which it controls. When a button is pushed, another window appears allowing the user to change component parameters.

For example, when the “Load Default Values” button is pushed, a separate window appears. Figure 1.5 is a snapshot of the window that appears after the button is selected. This window allows the user to choose a file in which the default values for the entire simulation are stored. Once the user selects the needed file, the GUI loads all the variables in that file and the pop-up window closes. Now all the default values for every parameter contained in the chosen file are loaded into the Matlab GUI handles structure for future use by the GUI.
Figure 1.4: Matlab Graphical User Interface
Figure 1.5: GUI: Load Default Values Window
Chapter 2

Ideal Simulation

2.1 Component Modules

For a baseline comparison of the cross-correlation and Doppler plots in the waveform diversity study, an ideal simulation is needed. The ideal simulation shows the cross-correlation and range-Doppler response of the radar system if the system is in a sense, perfect. Throughout this document, the terms “ideal” and “baseline” simulation are used interchangeably. Both refer to the simulation where all components are ideal. The ideal components do not have any nonlinearities that effect the output of the component. In order that the component models are easy to manipulate and modify, a separate module is constructed for each component. This allows for easy modification of any and all of the component parameters. Associated with each component module are default values for each of the parameters. Every parameter can be modified such that the component satisfies the system requirements.

2.1.1 Waveform Generator

Phase-coded waveforms were used to develop this simulation. The waveform generation module allows for control over parameters such as waveform selection and timing on a pulse-to-pulse and channel-to-channel basis. In execution of the ideal waveform generation module, the output code length, number of phases, chip rate, integration time, and pulse repetition frequency can all be altered. The chip rate is defined as the inverse of the length of the pulse. The units of chip rate
are Hertz. Both the integration time and the pulse repetition frequency are used in determining the number of pulses available for integration as shown in Equation 1.1. Table 2.1 shows the default parameters used in the ideal waveform generator module.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of phases</td>
<td>2</td>
</tr>
<tr>
<td>Code length</td>
<td>10</td>
</tr>
<tr>
<td>Chip rate</td>
<td>750 kHz</td>
</tr>
<tr>
<td>Integration Time</td>
<td>75 ms</td>
</tr>
<tr>
<td>Pulse Repetition Frequency</td>
<td>500 Hz</td>
</tr>
</tbody>
</table>

The simulation also has the ability to simulate multiple frequency branches. There is a block in the GUI that allows control over the number of branches used. In this block, the total number of branches can be specified by the user. The IF frequency corresponding to each branch is also a user input in this block.

### 2.1.2 Frequency Modulation

In the transmit chain of the end-to-end radar system simulation, two frequency modulations take place. The first is to the respective IF frequency of each branch and the other is a modulation of the summed IF signals to the RF frequency, illustrated by

\[
S_{IF, out}(t) = S_{I, in}(t) \cos(2\pi f_{IF}t) + S_{Q, in}(t) \sin(2\pi f_{IF}t) \tag{2.1}
\]

and

\[
S_{RF, out}(t) = S_{IF, in}(t) \cos(2\pi (f_{RF} - f_{IF})t), \tag{2.2}
\]

respectively. On receive, the signal is demodulated twice, once down from the RF to the IF frequency, and the other to baseband from the respective IF frequencies of that branch. An ideal mixer module is created for the analog modulation, or the modulation to the RF frequency. An ideal mixer is, mathematically, the signal multiplied by a sinusoidal waveform \cite{27}. The ideal mixer module allows for the user to change the carrier frequency of modulation. Table 2.2 shows the default parameters chosen for both the IF and RF modulations.
Table 2.2: Ideal Frequency Modulation

<table>
<thead>
<tr>
<th>Frequency Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Branch 1</td>
<td>200 MHz</td>
</tr>
<tr>
<td>Intermediate Frequency</td>
<td>Branch 2</td>
</tr>
<tr>
<td>Branch 3</td>
<td>400 MHz</td>
</tr>
<tr>
<td>RF Frequency</td>
<td>2 GHz</td>
</tr>
</tbody>
</table>

2.1.3 Amplifier

An amplifier is used to increase the power of a signal. The radar signal in the end-to-end simulation goes through an amplifier on both the transmit and receive chains. The signal gets amplified on transmit right before it is transmitted. The signal gets amplified a second time right after it is received. A module was created to simulate the amplifier. The ideal amplifier does not take noise into consideration in the output signal. The ideal amplifier simply scales the input signal [21]. The governing equation for an ideal amplifier is given by

\[ S_{out}(t) = aS_{in}(t) \]  

(2.3)

where \( a \) is a scalar quantity.

2.1.4 Converters

Digital-to-Analog Converter

An ideal digital-to-analog converter is simply modeled as an upsampler. To simulate the conversion from a digital signal to an analog signal, the signal is upsampled to a rate of ten times the Nyquist rate of the RF frequency. The Nyquist rate is defined as the value of two times the highest frequency component. Ten times the Nyquist rate of the RF frequency is used because, after the digital-to-analog conversion, the RF frequency is the carrier for the signal throughout the rest of the simulation. Therefore, to insure that the signal is heavily upsampled and aliasing due to nonlinearities does not occur, the analog sampling rate is chosen to be ten times the Nyquist of the carrier frequency. Since the analog sampling rate is dependent upon the RF frequency, the only user input to the ideal DAC
module is the RF frequency. The frequency modulation table, Table 2.2 shows the value chosen for the RF frequency.

**Analog-to-Digital Converter**

An analog-to-digital converter is the component that does just the opposite of the DAC. The ADC converts an analog signal to a digital signal. An ideal ADC is modeled just the opposite of the DAC also. It is modeled as a downsampler which converts analog sampling rate to the much lower digital sampling rate. Since the analog sampling rate is defined in the DAC module, and the digital rate is the original rate of sample preceding the digital-to-analog converter, there are no user inputs to the ideal ADC module.

**2.1.5 Time Delay Network**

The time delay network block consists of mainly the target properties. This block simulates the target interaction and signal return. The radar simulation has the capability to simulate a stationary point target as well as a moving point target. Therefore, the target properties specified by the user in the ideal time delay network module are the reference range, target range, range swath, target velocity, and signal amplitude. The reference range is the range to the center of the range swath, a range at which the time delay is known. The target range is the range at which the target lies. The range swath is the area on the ground that the radar can see. The target velocity is, as its name indicates, the velocity of the target. Table 2.3 shows the chosen target parameters for the ideal radar simulation. The time delay network block can easily be extended to simulate a phased array. The phased array allows for the study of beamforming or spatial filtering, etc.

**2.2 Software Testing**

**2.2.1 Simulation**

Before the ideal simulation is begun, the waveform generator module is used to input the pulse repetition frequency and the integration time required for the simulation. Once the user inputs these
Table 2.3: Ideal Time Delay Network

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target Range</td>
<td>9.75 km</td>
</tr>
<tr>
<td>Reference Range</td>
<td>10 km</td>
</tr>
<tr>
<td>Range Swath</td>
<td>1 km</td>
</tr>
<tr>
<td>Target Velocity</td>
<td>0 m/s</td>
</tr>
<tr>
<td>Signal Amplitude</td>
<td>1</td>
</tr>
</tbody>
</table>

determination, the number of pulses is calculated according to Equation 1.1. The integration time and pulse repetition frequency used in this ideal simulation are given in Table 2.1. The number of pulses needed based upon these values was found to be 38. Please note that the following description of the ideal simulation shows plots only from the first pulse generation. The plot of the detection cross-correlation is of only the first pulse also. Not until the Doppler shift is plotted are all the generated pulses used.

When the ideal simulation is begun, the number of frequency branches and their respective frequencies are chosen. Recall that the transmit architecture is shown in Figure 1.1. In this baseline simulation, three branches are used with the IF frequencies shown in Table 2.2. Once the number of branches and frequencies are chosen, a waveform for each branch is generated. The waveform generation module is used to generate a binary I- and Q-channel phase code for the three branches. The first plots in Figure 2.1 show the binary phase code for each of the three branches for the I-channel while the first plots in Figure 2.2 show the binary phase code for the Q-channel.

After the waveform generation, the binary signals get upsampled and modulated up to their respective IF brands. The intermediate frequencies for each branch, from Table 2.2 are 200 MHz, 300 MHz, and 400 MHz. The second and third plots in Figures 2.1 and 2.2 show the upsampled code and the IF modulated code for both the I- and the Q-channel, respectively.

The next step according to Figure 1.1 is a summation of the complex signals. The IF modulated I- and Q-channel for each branch are summed. Then the signals are summed over all of the branches such that only one signal is transmitted. The summed signal that is to propagate throughout the rest of the simulation is shown in Figure 2.3 while its frequency spectrum is shown in Figure 2.4.

The entire simulation to this point has been done digitally. The signal now needs to be co-
Figure 2.1: Ideal Simulation: Transmitted Waveforms: I-channel
Figure 2.2: Ideal Simulation: Transmitted Waveforms: Q-channel
Figure 2.3: Ideal Simulation: Summation of digital signals from all three branches
Figure 2.4: Ideal Simulation: Frequency spectrum of summed signal
verted from the digital domain to the analog domain by use of the ideal DAC module. The output signal from the ideal DAC should not differ much from the input signal because the output is simply an upsampled version of the input. The output signal is shown in Figure 2.5. It is also important to take note of the frequency spectrum of the output. The frequency spectrum of the DAC output shows peaks at the IF frequencies of each branch which is consistent with the simulation development. The frequency spectrum of the DAC output signal is shown in Figure 2.6. A nonlinear DAC generates many harmonics that will cause problems with the output signal and output frequency spectrum. A low pass analog filter is then needed to rid the signal of these unwanted frequency peaks. The ideal DAC, which is used in this simulation, does not generate harmonics and therefore no low pass filter following the DAC is implemented.

![Analog signal from DAC](image)

**Figure 2.5: Ideal Simulation: DAC output signal**

Now that the signal is in the analog domain, the ideal analog mixer module can be used to modulate the signal to the RF frequency. The RF frequency chosen in this ideal simulation is 2 GHz. The output signal from the mixer is shown in Figure 2.7. The math behind this simulation...
Figure 2.6: Ideal Simulation: DAC output frequency spectrum
demonstrates that the frequency spectrum of a signal after passing through a mixer should show two peaks, as seen in Equation 1.4. In the case of this simulation, the peaks should appear at $f_{RF}$ and $(f_{RF} - 2f_{IF})$. Since the simulation is using three branches, there should be three peaks centered around the two above mentioned frequencies. The first three peaks should be at 1.3, 1.4, and 1.5 GHz while the second set of peaks should be at 1.9, 2.0, and 2.1 GHz. This can be confirmed in the plot of the frequency spectrum shown in Figure 2.8.

![Figure 2.7: Ideal Simulation: RF mixer output signal](image)

Since the modulation is to the RF frequency, the only frequency peaks needed are the peaks at the RF frequency, or the peaks around 2 GHz. A high pass filter is implemented in order to get rid of all of the low frequency components in the spectrum. The ideal high pass filter was designed using the “Filter, Design, and Analysis Tool” in Matlab. Figure 2.9 shows the high pass filter spectrum with a stop band edge frequency of 1.3 GHz and a pass band edge frequency of 1.8 GHz. The resulting frequency spectrum of the high pass filtered signal is shown in Figure 2.10.
Figure 2.8: Ideal Simulation: RF mixer output frequency spectrum
Figure 2.9: Ideal Simulation: High pass filter spectrum of filter following RF mixer
Figure 2.10: Ideal Simulation: High pass filter output signal spectrum following RF mixer
In reference again to the system architecture in Figure 1.1, the final step before the signal is transmitted is an amplifier. Since this simulation is using all ideal components, there is need to change the signal amplitude to suit the specifications of the component following the amplifier. Therefore, there is no need for an amplifier in this baseline simulation.

After the signal passes through the entire transmit system architecture, it is delayed upon return. The time delay of the signal is based upon the range and velocity of the point target. These parameters are chosen by the user, specified in the target delay module, and for this simulation, have the values given in Table 2.3. The actual target range is calculated using

\[ R(\tau) = R_0 + v_t \tau_k \] (2.4)

where \( R_0 \) is the user input target range, \( v_t \) is the user input target velocity, and \( \tau_k \) is the time at which the signal pulse is transmitted.

The following equation

\[ t_0(\tau) = \frac{2R(\tau)}{c} \] (2.5)

is used to calculate the time delay of the signal where \( R(\tau) \) is the range previously calculated and \( c \) is the speed of light, which is 300,000,000 \( \text{m/s} \). The target used in this simulation is a stationary point target. Since the target velocity is zero, the actual target range is simply the target range chosen by the user. Using the speed of light and the target range, the time delay for this simulation is calculated to be 65\( \mu \text{s} \). Figure 2.11 is a plot of both the transmitted signal and the delayed signal at the center of the range swath.

After the signal is received, an amplifier is again needed to change the received signal amplitude. Again, this is an ideal simulation, so there is no need to change the amplitude for the components following the amplifier. Therefore, the amplifier is simply a placeholder in this baseline simulation.

The signal upon receive is still at RF frequency. The received signal should be converted to baseband. The first demodulation to take place, according to the receive architecture in Figure 1.2 is the modulation down from the RF frequency. The analog mixer module is used for the demodulation process. The demodulated signal is shown in Figure 2.12 while the frequency spectrum of the signal is shown in Figure 2.13. According to the mathematical derivation and Equation 1.7, the frequency spectrum of the signal should have peaks at \( (2f_{RF} - f_{IF}) \) and \( f_{IF} \). Since this ideal simulation
Figure 2.11: Ideal Simulation: Time Delayed Signal
is using three frequency channels, there should be three peaks at the higher frequencies, that is $2f_{RF}$ minus each one of the IF frequencies, and three peaks at each of the IF frequencies. The plot of the frequency spectrum is consistent with the mathematical theory. Figure 2.13 shows the entire frequency spectrum of the RF demodulated signal, with emphasis on both the low and high frequency peaks. The peaks are at the expected frequencies.

![Figure 2.12: Ideal Simulation: RF demodulation output signal](image)

Due to the fact that RF demodulation brought the signal to the IF frequency range, the only peaks to be concerned with are the low frequency peaks. Therefore, a low pass filter is needed to suppress the high frequency signal. The ideal low pass filter does not take into consideration noise or losses inside the component. The ideal filter was again designed using the “Filter, Design, and Analysis Tool” in Matlab. The low pass filter has a pass band edge frequency of 500 MHz and a stop band edge frequency of 800 MHz, as seen in the filter frequency spectrum in Figure 2.14. The resulting frequency spectrum of the output of the low pass filter is shown in Figure 2.15.
Figure 2.13: Ideal Simulation: RF demodulation output frequency spectrum
Figure 2.14: Ideal Simulation: Low pass filter spectrum of filter following RF demodulation on receive
Figure 2.15: Ideal Simulation: Low pass filter output frequency spectrum following RF demodulation on receive
After the demodulation to IF and low pass filtering, the signal is still in the analog domain. In order to compare the received signal with the transmitted signal, both signals need to be in the same domain. The next step in the simulation is to convert the received signal from the analog domain to the digital domain using an ideal ADC module. As with the digital-to-analog conversion, the output of the ADC should not differ much from the input to the component because the ideal converter is simply modeled as a downsampler. Figure 2.16 shows the output of the ADC in the time domain. The important aspect of the output is the frequency spectrum of the signal, shown in Figure 2.17. The spectrum should show signal components at each of the three IF frequencies. It is clear from the plot of the frequency spectrum that the IF frequencies of each branch are 200 MHz, 300 MHz, and 400 MHz.

![Figure 2.16: Ideal Simulation: ADC output signal](image)

A second demodulation is needed to get the signal from the IF frequency to complex baseband. The I- and Q-channels are each IF band are demodulated separately. Figures 2.18 through 2.21 show both the received signal and its corresponding frequency spectrum for both complex channels.
Figure 2.17: Ideal Simulation: ADC output frequency spectrum
of branch one. Figures 2.22 through 2.25 and Figures 2.26 through 2.29 show the same plots for branches two and three, respectively.

![Digital signal from mixer](image)

**Figure 2.18: Ideal Simulation: IF demodulated I-channel signal: Branch 1**

In examination of the plots of the signal frequency spectrum for both complex channels, the modulation process generated unwanted harmonics in the frequency spectrum of the complex output signal for every branch. A low pass filter is needed to filter out these unwanted frequency peaks in the spectrums. The received signal should also be at baseband. Figure 2.30 shows the frequency response of the low pass filter used for the complex channels in every branch. The signal spectrum after filtering remains the same for all channels in that there is a single peak at baseband. The filtered signal spectrum for the I-channel of the first branch can be seen in Figure 2.31. Note that all unwanted harmonics were filtered out leaving only the peak at baseband.

The above described simulation is similar for stationary targets as well as moving targets. Please note that the results section of this chapter, immediately following, addresses the stationary
Figure 2.19: Ideal Simulation: IF demodulated I-channel frequency spectrum: Branch 1
Figure 2.20: Ideal Simulation: IF demodulated Q-channel signal: Branch 1
Figure 2.21: Ideal Simulation: IF demodulated Q-channel frequency spectrum: Branch 1
Figure 2.22: Ideal Simulation: IF demodulated I-channel signal: Branch 2
Figure 2.23: Ideal Simulation: IF demodulated I-channel frequency spectrum: Branch 2
Figure 2.24: Ideal Simulation: IF demodulated Q-channel signal: Branch 2
Figure 2.25: Ideal Simulation: IF demodulated Q-channel frequency spectrum: Branch 2
Figure 2.26: Ideal Simulation: IF demodulated I-channel signal: Branch 3
Figure 2.27: Ideal Simulation: IF demodulated I-channel frequency spectrum: Branch 3
Figure 2.28: Ideal Simulation: IF demodulated Q-channel signal: Branch 3
Figure 2.29: Ideal Simulation: IF demodulated Q-channel frequency spectrum: Branch 3
Figure 2.30: Ideal Simulation: Baseband low pass filter frequency spectrum used for all branches and channels
Figure 2.31: Ideal Simulation: Received signal spectrum after low pass filter to remove harmonics
target detection as well as moving target detection. The difference between stationary and moving targets is noticeable in the analysis of the range-Doppler plots since a moving target causes Doppler shift.

2.2.2 Results

Cross-correlation

Once the baseband signal is received, a cross-correlation of the transmitted pulse with the received pulse can be performed. Since the signals are complex, the cross-correlation is complex also. The cross-correlation plot should peak at the time delay of the signal. Looking back, the theoretical time delay for this ideal simulation was calculated to be $65\mu s$. Figure 2.32 shows the cross-correlation plot for each of the three branches. Within the figure contains the time delay value extracted from the peak of the cross-correlation plots. The simulated time delays are $64.887\mu s$, $64.881\mu s$, and $64.883\mu s$ for pulse number one of branches one, two, and three, respectively.

Doppler

Once all of the 38 pulses are generated, received, and detected, a range-Doppler plot can be produced. The range-Doppler plot should show a peak value at the range of the target and the correct Doppler frequency shift. Figure 2.33 shows the plots for each of the three branches. Since the target is stationary, there should be no Doppler shift. The range was specified by the user in the target delay module, and for this simulation had the value of 9.75 km. It can be seen that in each of the range-Doppler plots, the peak is at 0 Hz Doppler shift and a range of 9.75 km, as predicted.

After the range-Doppler plots are generated, calculations of the side lobe levels of each of the plots are performed. A table of the calculated values for both the range and the Doppler shift are shown in Table 2.4.

The ideal simulation was run again, but this time with a moving target. The target was given a positive velocity of $10\text{m/s}$, input into the target delay module. Range-Doppler plots for the moving target were generated along with the main lobe and side lobe calculations. The peak of the range-Doppler plot with a moving target of specified velocity should now be at 133 Hz, according to Equation 1.17 where the frequency used to calculate the wavelength is RF, or 2 GHz. This is true
Figure 2.32: Ideal Simulation: Cross-correlation
Figure 2.33: Ideal Simulation: Range-Doppler plots for zero Doppler shift
Table 2.4: Ideal Doppler calculations: Stationary Target

<table>
<thead>
<tr>
<th>Branch</th>
<th>Parameter</th>
<th>Doppler</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>3 dB width</td>
<td>1.1875</td>
<td>0.5333</td>
</tr>
<tr>
<td>1</td>
<td>15 dB width</td>
<td>2.4375</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-30.3006</td>
<td>-15.9568</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-22.2435</td>
<td>-12.3300</td>
</tr>
<tr>
<td></td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td>2</td>
<td>15 dB width</td>
<td>2.500</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-28.1754</td>
<td>-14.3422</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-22.2375</td>
<td>-10.3466</td>
</tr>
<tr>
<td></td>
<td>3 dB width</td>
<td>1.2188</td>
<td>0.5333</td>
</tr>
<tr>
<td>3</td>
<td>15 dB width</td>
<td>2.4375</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-26.6499</td>
<td>-13.7983</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-21.2740</td>
<td>-10.7666</td>
</tr>
</tbody>
</table>

of all three range-Doppler plots shown in Figure 2.34. Table 2.5 shows the calculated main lobe and side lobe levels.

The ideal simulation was run a third time, but the target is now moving with a velocity of $-10 \text{m/s}$. Range-Doppler plots for the moving target were generated along with the main lobe and side lobe calculations. The peak of the range-Doppler plot with a moving target of specified velocity should now be at -133 Hz. This is true of all three range-Doppler plots shown in Figure 2.34. Table 2.6 shows the calculated main lobe and side lobe levels.
Figure 2.34: Ideal Simulation: Range-Doppler plots for positive Doppler shift
Figure 2.35: Ideal Simulation: Range-Doppler plots for negative Doppler shift
<table>
<thead>
<tr>
<th>Branch</th>
<th>Parameter</th>
<th>Doppler</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3 dB width</td>
<td>1.2188</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.4375</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-21.9151</td>
<td>-15.5333</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-17.3063</td>
<td>-12.2900</td>
</tr>
<tr>
<td>2</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.5313</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-25.0058</td>
<td>-13.8884</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-20.5662</td>
<td>-10.3357</td>
</tr>
<tr>
<td>3</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.500</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-26.2576</td>
<td>-12.5132</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-19.6275</td>
<td>-10.4047</td>
</tr>
</tbody>
</table>
### Table 2.6: Ideal Doppler calculations: Negative Velocity Target

<table>
<thead>
<tr>
<th>Branch</th>
<th>Parameter</th>
<th>Doppler</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.4375</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-23.5590</td>
<td>-15.3563</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-17.9694</td>
<td>-11.8270</td>
</tr>
<tr>
<td>2</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.5313</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-27.5469</td>
<td>-13.5717</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-20.2527</td>
<td>-9.4997</td>
</tr>
<tr>
<td>3</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.500</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-25.9022</td>
<td>-13.1631</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-18.7387</td>
<td>-10.6403</td>
</tr>
</tbody>
</table>
Chapter 3

Non-ideal Simulation

3.1 Component Modules

All hardware components such as amplifiers, filters, analog-to-digital converters, digital-to-analog converters, and mixers are nonlinear to some degree. The nonlinearity gives rise to unwanted spectral components in the output. There are different ways to model this nonlinear effect, but the most useful is as a polynomial. A polynomial model is simple yet efficient, and analytical solutions are possible [15]. When modeling a nonlinear component with a polynomial, third and fifth order polynomials are acceptable since higher order terms are usually negligible.

A common way to represent the nonlinearity of a component is two-tone intermodulation distortion (IMD). Intermodulation distortion is characterized in the output of a device by the appearance of frequencies that are linear combinations of the fundamental frequencies and all harmonics in the input signal. For example, two sine waves closely spaced in frequency are applied to the input of a device. Distinct spectral peaks at predictable frequencies arise due to the nonlinear effects of the component as seen in Figure 3.1. The amplitude of these peaks relative to the two fundamental peaks is a measure of the component nonlinearity [12].

Two-tone IMD has a recognizable spectral component pattern. The primary spectral components can be grouped into four frequency bands; DC, fundamental, $2^{nd}$ harmonic, and $3^{rd}$ harmonic. Figure 3.1 [34] shows the fundamental peaks and the nonlinear components. The input tones are at $f_1$ and $f_2$ while the $2^{nd}$ order nonlinearities lie at DC, $(f_2 - f_1)$, $2f_1$, $(f_1 + f_2)$, and $2f_2$. The $3^{rd}$
order nonlinearities lie at \((2f_1 - f_2), f_1, f_2, (2f_2 - f_1), 3f_1, (2f_1 + f_2), (f_1 + 2f_2), \) and \(3f_2\), though the components at the fundamental frequency are smaller in amplitude than the input tones. The components in the fundamental band are of interest because they are “in-band”. The components not “in-band” can usually be filtered out. The “in-band” components are generated by the 3rd order nonlinearities, and therefore, the odd order polynomial coefficients in the nonlinear model are of special interest [15].

![Figure 3.1: Two-tone Intermodulation Distortion](image)

Another way that nonlinearities in components are defined is by use of intercept points and compression points. When the power input to a component is increased, the nonlinear spectral parameters of the component increase faster in amplitude than the fundamentals. The 2nd order polynomial slope increases at a rate of 2 output dB per input dB while the 3rd order polynomial slope increases at a rate of 3 output dB per input dB. Therefore there is a point at which the output power of the nonlinear spectral components is equal to the output power of the fundamentals. That point is called the intercept point [12]. Figure 3.2 [34] shows this intercept point for the 3rd order case, specified either by input power as \(iIP_3\), or output power as \(oIP_3\). The roll off of both the fundamental as well as the 3rd order nonlinear spectral component is due to compression.

The component can be additionally characterized by the output 1 dB compression point, or \(oCP_1\). Compression is defined as what happens when the actual output power falls below the ideal output power. The output power could also reach its maximum, which results in saturation. The nonlinear component will experience compression and eventually saturation as the input power to the component is increased [21]. The compression point is the output power value that falls 1 dB below the ideal output power [15, 12, 21]. This is illustrated in figure 3.3 [34].

Most components also contain noise which causes nonlinearities. There are three sources of noise in a component; flicker, thermal, and shot noise. Flicker, or \(\frac{1}{f}\) noise is believed to be caused
Figure 3.2: Third Order Nonlinear Intercept Point

Figure 3.3: Nonlinear Compression Point
by defects and contaminants in semiconductor crystals, though the origin is a subject of current research. Flicker noise typically occurs at low frequencies or audio band and has a spectrum that varies inversely with frequency [20].

Thermal noise is generated within the resistors of the component and is proportional to the temperature of the system. Thermal noise has a spectrum that is white, or flat [16, 20] as does shot noise. The discrete fluctuations of current through the component results in shot noise. Shot noise is therefore proportional to the level of current through the device [20]. For most components, one of the white spectrum noise sources will dominate. As a result, the noise figure can be characterized by a single noise power level and modeled with a white spectrum.

3.1.1 Frequency Modulation

Oscillator

The main cause of nonlinearity in an oscillator is phase noise. Phase noise is defined as the ratio of the noise in a 1 Hz bandwidth at a specified frequency offset relative to the oscillator signal amplitude at the carrier frequency [28]. The phase noise spectrum can be very complicated. The noise can arise from a number of different sources and it also depends on the type of device. Environmental factors such as vibration, mechanical shock, and temperature can cause noise. The oscillator module in Matlab uses a known single sideband spectrum, a known root mean squared (RMS) timing jitter, or a peak-to-peak timing jitter to model the phase noise.

If a single sideband spectrum is known, the phase noise frequencies and phase noise powers while the RMS and peak-to-peak jitter parameters are left un-
known or open. If the spectrum is not known, the jitter must be specified. Table 3.1 shows the values chosen for the phase noise parameters in order that the white noise Gaussian spectrum can be shaped.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phase noise frequency</td>
<td>[ 1e3 10e3 100e3 400e3 1e6 3e6 10e6 ] Hz</td>
</tr>
<tr>
<td>Phase noise power</td>
<td>[ -84, -100, -96, -102, -109, -115, -122 ] dBc/Hz</td>
</tr>
<tr>
<td>RMS jitter</td>
<td>[ ] s</td>
</tr>
<tr>
<td>Peak-to-peak jitter</td>
<td>[ ] s</td>
</tr>
</tbody>
</table>

**Mixer**

Mixers are characterized by several different parameters. The most useful are output intercept points, the output compression point, and noise figure as described above. Conversion loss is also a contributor to the nonlinearity of the mixer. The difference in amplitude of the IF signal to the RF signal output if performing an upconversion or the difference between the RF signal to the IF signal output if performing a downconversion is defined as the conversion loss [30].

A module to simulate a nonlinear mixer was created in Matlab. The LO input to the mixer is generated by the nonlinear oscillator. The mixer module takes as user inputs intercept points, the compression point, the conversion loss, the noise figure, and isolation figures. Isolation figures are used so that either the LO signal or the analog input signal does not cause spurious products in the output of the mixer. The parameter values used in the nonlinear mixer model are shown in Table 3.2. These values are taken from the specification sheet for a Hittite double-balanced mixer [35].

### 3.1.2 Amplifier

Amplifiers can be characterized by a few different parameters. Similar to mixers, the most frequent parameters used to characterize an amplifier are the output intercept points and the output compression point. Amplifiers also contain noise. Using the noise figure together with the output intercept points and the compression point, along with the understanding of how to model intermodulation
Table 3.2: Non-ideal Mixer

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conversion Loss</td>
<td>9 dB</td>
</tr>
<tr>
<td>Output 2\textsuperscript{nd} Order Intercept Point</td>
<td>46 dBm</td>
</tr>
<tr>
<td>Output 3\textsuperscript{rd} Order Intercept Point</td>
<td>8 dBm</td>
</tr>
<tr>
<td>Output 1 dB Compression Point</td>
<td>1 dBm</td>
</tr>
<tr>
<td>Noise Figure</td>
<td>9 dB</td>
</tr>
<tr>
<td>Input Analog Signal Isolation</td>
<td>17 dB</td>
</tr>
<tr>
<td>Lo Signal Isolation</td>
<td>40 dB</td>
</tr>
</tbody>
</table>

distortion in an amplifier, a Matlab module of a non-ideal amplifier can be generated. The module takes as user inputs the gain of the amplifier, the output 2\textsuperscript{nd} order intercept point, the output 3\textsuperscript{rd} order intercept point, the output 1 dB compression point, and the noise figure. The amplifier module also allows the user to choose whether to model the component as a third or a fifth order polynomial, and whether or not to model noise in the component. Depending on the compression and saturation points of the amplifier, the use of a third order model might not be sufficient.

Specific amplifiers are chosen for the diversity study with certain output power intercept points, noise figures, and compression points, which are user inputs to the non-ideal amplifier model. Table 3.3 shows the values of the parameters for the amplifiers used in the diversity study. The same amplifier is used on transmit and receive in the ideal simulation and is made by Hittite [36].

3.1.3 Converters

The nonlinearities of analog-to-digital converters are similar to those of digital-to-analog converters and they can be modeled using a 3\textsuperscript{rd} order polynomial. The converter nonlinearity is specified in two different ways, one being integral nonlinearity (INL) and the other as spurious free dynamic range (SFDR).

Integral nonlinearity is the deviation of the input/output response curve from the ideal linear response. There are a couple of ways in which the INL can be described. The first is with an actual measured data curve. If the data curve is provided, it can be fit with a 3\textsuperscript{rd} order polynomial.
Table 3.3: Non-ideal Amplifier

<table>
<thead>
<tr>
<th>Location</th>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Gain</td>
<td>43 dB</td>
</tr>
<tr>
<td>Transmit</td>
<td>Output 2\textsuperscript{nd} Order Intercept Point</td>
<td>56 dBm</td>
</tr>
<tr>
<td>Receive</td>
<td>Output 3\textsuperscript{rd} Order Intercept Point</td>
<td>59 dBm</td>
</tr>
<tr>
<td></td>
<td>Output 1dB Compression Point</td>
<td>43 dBm</td>
</tr>
<tr>
<td></td>
<td>Noise Figure</td>
<td>8.5 dB</td>
</tr>
<tr>
<td></td>
<td>Gain</td>
<td>13 dB</td>
</tr>
<tr>
<td>Receive</td>
<td>Output 2\textsuperscript{nd} Order Intercept Point</td>
<td>37 dBm</td>
</tr>
<tr>
<td></td>
<td>Output 3\textsuperscript{rd} Order Intercept Point</td>
<td>37 dBm</td>
</tr>
<tr>
<td></td>
<td>Output 1dB Compression Point</td>
<td>22 dBm</td>
</tr>
<tr>
<td></td>
<td>Noise Figure</td>
<td>1.6 dB</td>
</tr>
</tbody>
</table>

curve. The second is by a statistical maximum standard deviation from the ideal. If the curve is not provided, a third order shape is assumed and scaled to fit the maximum standard deviation.

The dynamic range of the device output which is free of spurious spectral components, either due to nonlinearities or noise is called the spurious free dynamic range. The SFDR is usually given in decibels for a particular frequency and input level [13]. The shape is again assumed to be 3\textsuperscript{rd} order, and scaled to fit the specified SFDR performance.

**Digital-to-Analog Converter**

The digital-to-analog converter module has many user defined inputs. Among them are the resolution, output voltage constraints, maximum expected offset error, maximum expected full scale gain error, and settling time. Offset error indicates how well the actual transfer function matches the ideal transfer function at a certain point. For DACs that means the analog output response to an input of all zeros [6]. Full scale gain error is defined as the difference between the actual and ideal output voltage range [6]. Settling time is defined as the amount of time required for the DAC output to settle, within a specific tolerance, to a certain level [19]. Also as user inputs are the integral nonlinearity and the noise spectral density. The default inputs are shown in table 3.4, modeling a
16-bit, 600Msps, high-dynamic-performance DAC with LVDS inputs [37].

**Table 3.4: Non-ideal Digital-to-Analog Converter**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DAC Resolution</td>
<td>16 bits</td>
</tr>
<tr>
<td>Output Minimum Voltage</td>
<td>-0.1 V</td>
</tr>
<tr>
<td>Output Maximum Voltage</td>
<td>0.1 V</td>
</tr>
<tr>
<td>Maximum Expected Offset Error</td>
<td>0.02</td>
</tr>
<tr>
<td>Maximum Expected Full Scale Gain Error</td>
<td>4</td>
</tr>
<tr>
<td>Settling Time</td>
<td>1e-9 s</td>
</tr>
<tr>
<td>Integral Nonlinearity</td>
<td>3.8</td>
</tr>
<tr>
<td>Noise Spectral Density</td>
<td>-155 \text{dBFS} / Hz</td>
</tr>
</tbody>
</table>

**Analog-to-Digital Converter**

The analog-to-digital converter module in the simulation has, as user defined inputs, a resolution, input voltage constraints, maximum expected offset error and maximum expected full scale gain error. The offset error of an ADC is the voltage difference until the first transition occurs. The ADC also takes the integral nonlinearity as an input, as does the DAC. Table 3.5 shows the default values for the ADC parameters, based on specs of an actual \( \pm 5 \) V, 600 Msps, 8-bit ADC [38].

**Table 3.5: Non-ideal Analog-to-Digital Converter**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADC Resolution</td>
<td>8 bits</td>
</tr>
<tr>
<td>Input Minimum Voltage</td>
<td>-0.02 V</td>
</tr>
<tr>
<td>Input Maximum Voltage</td>
<td>0.02 V</td>
</tr>
<tr>
<td>Maximum Expected Offset Error</td>
<td>0</td>
</tr>
<tr>
<td>Maximum Expected Full Scale Gain Error</td>
<td>0</td>
</tr>
<tr>
<td>Integral Nonlinearity</td>
<td>0.25</td>
</tr>
</tbody>
</table>
3.1.4 Analog Filter

Analog filters are modeled as infinite impulse response (IIR) filters. The most efficient are of Butterworth, Chebyshev, or Elliptical design [22]. Analog filters have nonlinear properties as do most hardware components. The nonlinearities in analog filters are insertion loss, nonlinear response curves, and noise. Insertion loss is defined as attenuation of the pass band signal due to losses within the component. Insertion loss is modeled with a random scale factor within component specification.

Analog filters have a nonlinear response curve that could potentially produce intermodulation. This type of response curve typically has a small effect on the nonlinearity of the component, and can therefore be easily controlled with a small input amplitude. However, the nonlinearity contributed by the nonlinear response curve is not normally modeled because the curve parameters are not typically given in the component specification sheet. Analog filters also contain noise that effects the nonlinearity of the component. Again, however, the noise is not typically modeled because the values are not normally given on the specification sheet.

A Matlab module of an analog filter is designed. The module takes in as user inputs the type of filter, being Butterworth, Chebyshev Type I, Chebyshev Type II, or Elliptical. The pass-band and stop-band frequencies and attenuations can also be specified by the user. The last user input is the insertion loss. Table 3.6 shows the default type of filter and the default value of the insertion loss for each of the analog filters on transmit and receive, both high-pass and low-pass. The pass-band and stop-band frequencies for the two filters on transmit can be changed to suit the simulation needs. The pass-band and stop-band frequencies of the filter on receive, however, are based upon the IF frequencies that were previously specified. Therefore, the edge frequencies of the low-pass filter on receive can not be changed by the user. It is also important to note that the analog filter module can not have a transition width of less than 500 MHz.

3.1.5 Time Delay Network

The time delay network block values did not change much between the ideal values and the non-ideal values. The target properties all remained the same, but the signal amplitude scale value changed. The ideal simulation used a signal amplitude scale value of one. The signal amplitude
Table 3.6: Non-Ideal Analog Filters

<table>
<thead>
<tr>
<th>Function</th>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transmit LPF</td>
<td>Type</td>
<td>Butterworth</td>
</tr>
<tr>
<td></td>
<td>Insertion Loss</td>
<td>0.4</td>
</tr>
<tr>
<td>Transmit HPF</td>
<td>Type</td>
<td>Butterworth</td>
</tr>
<tr>
<td></td>
<td>Insertion Loss</td>
<td>0.6</td>
</tr>
<tr>
<td>Receive LPF</td>
<td>Type</td>
<td>Butterworth</td>
</tr>
<tr>
<td></td>
<td>Insertion Loss</td>
<td>0.4</td>
</tr>
</tbody>
</table>

scale was changed to a value of 0.1 for use in the non-ideal simulation. Table 3.7 shows the non-ideal target parameters.

Table 3.7: Non-ideal Time Delay Network

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target Range</td>
<td>9.75 km</td>
</tr>
<tr>
<td>Reference Range</td>
<td>10 km</td>
</tr>
<tr>
<td>Range Swath</td>
<td>1 km</td>
</tr>
<tr>
<td>Target Velocity</td>
<td>0 m/s</td>
</tr>
<tr>
<td>Signal Amplitude</td>
<td>0.1</td>
</tr>
</tbody>
</table>

3.2 Software Testing

3.2.1 Simulation

The simulation process with an entire nonlinear end-to-end radar system follows exactly the process described in detail in the Software Testing: Simulation section in Chapter 2. The same transmit and receive architecture is followed for this nonlinear simulation, given in Figures 1.1 and 1.2. The difference between this simulation and the baseline simulation is the fact that the component
models used, the converters, filters, mixers, and amplifiers have added nonlinearities to the modeling. Therefore, some detail and plots may not be in this simulation due to the fact that the ideal simulation generated the same plot.

For simplicity, this nonlinear simulation uses many of the same parameters as the baseline simulation pending that the module in question does not include nonlinear affects. The waveform generation, digital upsample, IF modulation, and digital summation processes are the same as in the baseline simulation. So to recap, 38 pulses are generated, three frequency channels are used, and the respective IF frequencies are 200 MHz, 300 MHz, and 400 MHz, respectively. The pulse repetition frequency and the integration time used to calculate the number of pulses are given in Table 2.1.

Now the signal encounters the first nonlinear component, the digital-to-analog converter. The DAC output signal is shown in Figure 3.4 along with its corresponding frequency spectrum in Figure 3.5. It is known that the nonlinearity of the DAC adds unwanted harmonics to the signal. These harmonics decrease in amplitude as the frequency increases. The harmonics can be seen in the frequency spectrum of the output signal.

A low pass filter is needed such the harmonics do not affect the DAC output signal. The nonlinear parameter of the Butterworth low pass filter used is given in Table 3.6. The pass band edge frequency of the filter is 250 MHz while the stop band edge frequency is 750 MHz. These edge frequencies meet the transition width requirement of the filter model and they also allow the correct frequency components to pass through, that is, the components at the IF frequencies. The low pass filtered output and frequency spectrum are shown in Figures 3.6 and 3.7, respectively.

The signal is now in the analog domain, and the nonlinear mixer module can be used to modulate the signal up to the RF frequency. The RF frequency is the same as in the ideal simulation which is 2 GHz. The signal is shown in Figure 3.8 and, by inspection, is noticeably noisier than the modulated signal in the baseline simulation shown in Figure 2.7. The frequency spectrum of the signal should show two peaks as explained mathematically by Equation 1.4. The peaks should appear at $f_{RF}$ and $(f_{RF} - 2f_{IF})$, or in this case 2 GHz and 1.4 GHz. The frequency spectrum of the signal at RF, shown in Figure 3.9 shows this to be true.

The frequency components of the signal which are needed are the components at 2 GHz. Therefore, a high pass filter is needed. The nonlinear analog filter module is used to filter the signal. The nonlinear parameters of this high pass filter are given in Table 3.6. The stop band and pass band
Figure 3.4: Non-ideal Simulation: Nonlinear DAC output signal
Figure 3.5: Non-ideal Simulation: Nonlinear DAC output signal spectrum
Figure 3.6: Non-ideal Simulation: Low pass filtered signal after filtering out unwanted DAC harmonics
Figure 3.7: Non-ideal Simulation: Low pass filtered signal frequency spectrum after filtering out unwanted DAC harmonics
Figure 3.8: Non-ideal Simulation: RF mixer output signal
Figure 3.9: Non-ideal Simulation: RF mixer output signal frequency spectrum
edge frequencies are 199.5 and 2 GHz, respectively, satisfying the analog filter module transition width requirement. The signal frequency spectrum now only consists of the high frequency components as seen in Figure 3.11.

Figure 3.10: Non-ideal Simulation: High pass filter output signal following RF mixer

In reference to the system architecture in Figure 1.1, the next step in the transmit chain is an amplifier. The nonlinear amplifier parameters are given in Table 3.3. This amplifier is modeled with a third order polynomial. Noise is also included in the amplifier model. After the signal is amplified, the signal is time delayed. The same time delay network parameters are used for this nonlinear simulation as were used in the baseline simulation. Those parameters are given in Table 2.3. The same equations are used to calculate the actual target range as well as the time delay. Since the parameters remain the same, the calculated time delay is 65μs.

After the signal is received, a power amplifier is again needed. The amplifier model used in the receive chain is the same as the model in the transmit chain. A third order polynomial along with
Figure 3.11: Non-ideal Simulation: High pass filter output signal frequency spectrum following RF mixer
noise are used to model the receive amplifier. The signal, though delayed and amplified is still at RF frequency. According to the receive architecture, Figure 1.2, the first demodulation to take place is from RF to IF frequency. The analog mixer module is used for this demodulation process. The demodulated signal is shown in Figure 3.12. The frequency spectrum of the demodulated signal should again show two peaks as seen mathematically in Equation 1.7. This mixing process should produce peaks at $(2f_{RF} - f_{IF})$ and $f_{IF}$. Since this nonlinear simulation is using three frequency channels, there should be three peaks at the three high frequencies, $2f_{RF}$ minus each one of the IF frequencies, and three peaks at each of the IF frequencies. This is verified in the plot of the signal spectrum, Figure 3.13.

![Figure 3.12: Non-ideal Simulation: RF demodulated signal](image)

The required components of the signal are the low frequency components. Therefore, a low pass filter is needed. The analog filter module is again used with the nonlinear parameter value given in Table 3.6. Note that the average IF frequency is 300 MHz. Due to the transition width requirement of the analog filter module, the pass band edge frequency is 200 MHz higher than the
average IF frequency, or 500 MHz while the stop band edge frequency is 700 MHz higher than the average, or 1 GHz. The resulting filtered signal and frequency spectrum are shown in Figures 3.14 and Figure 3.15.

The signal now needs to be converted from the analog domain to the digital domain. The conversion is done by use of the nonlinear analog-to-digital converter module. The nonlinear parameters used in this simulation are given in Table 3.5. It is worth noting that the minimum and maximum input voltages change based upon the input signal to the ADC. If input voltages are not correct, the signal is not scaled properly, and the output of the component will be clipped. Figure 3.16 shows the output signal from the ADC module and Figure 3.17 shows the signals frequency spectrum. The frequency spectrum peaks at all three of the IF frequencies, as it should.

In looking at the receive architecture, the last step in the nonlinear simulation is the digital IF demodulation and filter to keep the frequency components of the received signal at baseband. These two processes are the same as in the ideal simulation, and therefore are not included in this simulation description. After the demodulation and filtering processes, the complex received signal is at baseband and the detection process can begin.

The above described nonlinear simulation is similar for stationary targets as well as moving
Figure 3.14: Non-ideal Simulation: Low pass filter output signal following RF demodulation
Figure 3.15: Non-ideal Simulation: Low pass filter output signal frequency spectrum following RF demodulation
Figure 3.16: Non-ideal Simulation: Nonlinear ADC output signal
Figure 3.17: Non-ideal Simulation: Nonlinear ADC output signal frequency spectrum
targets. Please note that the results section of this chapter, immediately following, addresses the stationary target detection as well as moving target detection. The difference between stationary and moving targets is noticeable in the analysis of the range-Doppler plots since a moving target causes Doppler shift.

### 3.2.2 Results

#### Cross-correlation

The expected results of the detection process of this nonlinear simulation are that of the baseline simulation. If detection is still possible, the chosen nonlinear components are working properly as is the simulation code. The first plots to compare are the cross-correlation plots. The cross-correlation should peak at the time delay of the signal. The theoretical time delay was calculated to be $65\mu s$ and it can be seen in all three plots of Figure 3.18 that the peak is approximately that. The simulated time delays are $64.888\mu s$, $64.890\mu s$, and $64.886\mu s$ for pulse number one of branches one, two, and three, respectively. It can be concluded from the experimental time delays that the nonlinear simulation is working as expected.

#### Doppler

The target in the above described nonlinear simulation was a stationary target. Therefore, there should be no Doppler shift. Once all of the 38 pulses are received, a range-Doppler plot can be produced. The plot should peak at the value of the range as well as the correct Doppler shift. The range was a user input to the target delay module and had a value of 9.75 km for this simulation. The range-Doppler plots shown in Figure 3.19 all show the correct values of range and Doppler shift.

After the range-Doppler plots are generated, calculations of the side lobe levels of each of the plots are performed. The 3 dB main lobe width, 15 dB main lobe width, peak to side lobe ratio, and integrated side lobe ratio are calculated. A table of the calculated values for both the range and Doppler are shown in Table 3.8.

This nonlinear simulation was run a second time, but this time with a moving target. The target was given a positive velocity of $10\frac{m}{s}$, input into the target delay module. Range-Doppler
Figure 3.18: Non-ideal Simulation: Cross-correlation
Figure 3.19: Non-ideal Simulation: Range-Doppler plots for zero Doppler shift
Table 3.8: Non-ideal Doppler Calculations: Stationary Target

<table>
<thead>
<tr>
<th>Branch</th>
<th>Parameter</th>
<th>Doppler</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3 dB width</td>
<td>1.1875</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.4375</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-29.5896</td>
<td>-15.6758</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-22.0888</td>
<td>-12.4556</td>
</tr>
<tr>
<td>2</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.5625</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-30.4604</td>
<td>-14.8372</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-22.2228</td>
<td>-10.6950</td>
</tr>
<tr>
<td>3</td>
<td>3 dB width</td>
<td>1.2188</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.4375</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-29.2518</td>
<td>-11.5552</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-20.8486</td>
<td>-8.8810</td>
</tr>
</tbody>
</table>

plots for the moving target were generated along with the main lobe and side lobe calculations. The peak of the range-Doppler plot with a moving target of specified velocity should now be at 133 Hz, according to Equation 1.17 where the frequency used to calculate the wavelength is RF, or 2 GHz. This is true of all three range-Doppler plots shown in Figure 3.20 which further proves that the simulation is working as to be expected. Table 3.9 shows the calculated main lobe and side lobe levels.

The nonlinear simulation was run again but with the target moving at a velocity of \(-10\frac{m}{s}\). Range-Doppler plots for the moving target were generated along with the main lobe and side lobe calculations. The peak of the range-Doppler plot with a moving target of specified velocity should now be at -133 Hz. This is true of all three range-Doppler plots shown in Figure 3.21. Table 3.10 shows the calculated main lobe and side lobe levels.
Figure 3.20: Non-ideal Simulation: Range-Doppler plots for positive Doppler shift
<table>
<thead>
<tr>
<th>Frequency shift (Hz)</th>
<th>Range (km)</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.3</td>
<td>9.4</td>
</tr>
<tr>
<td></td>
<td>−250</td>
</tr>
</tbody>
</table>

(a) Branch 1  
(b) Branch 2  
(c) Branch 3

Figure 3.21: Non-ideal Simulation: Range-Doppler plots for negative Doppler shift
Table 3.9: Non-ideal Doppler calculations: Positive Velocity Target

<table>
<thead>
<tr>
<th>Branch</th>
<th>Parameter</th>
<th>Doppler</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3 dB width</td>
<td>1.2188</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.4063</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-19.5184</td>
<td>-15.5866</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-15.2741</td>
<td>-12.3392</td>
</tr>
<tr>
<td>2</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.5625</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-26.9544</td>
<td>-14.0454</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-19.6786</td>
<td>-10.4284</td>
</tr>
<tr>
<td>3</td>
<td>3 dB width</td>
<td>1.2188</td>
<td>0.5333</td>
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<tr>
<td></td>
<td>15 dB width</td>
<td>2.500</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
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<td>-13.4313</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-16.1785</td>
<td>-9.6804</td>
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</table>
Table 3.10: Non-ideal Doppler calculations: Negative Velocity Target

<table>
<thead>
<tr>
<th>Branch</th>
<th>Parameter</th>
<th>Doppler</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3 dB width</td>
<td>1.2188</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.4063</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-20.5285</td>
<td>-15.2137</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-15.5700</td>
<td>-12.2056</td>
</tr>
<tr>
<td>2</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.5625</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-28.7281</td>
<td>-14.7704</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-21.1511</td>
<td>-10.4869</td>
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<tr>
<td>3</td>
<td>3 dB width</td>
<td>1.2500</td>
<td>0.5333</td>
</tr>
<tr>
<td></td>
<td>15 dB width</td>
<td>2.500</td>
<td>0.9333</td>
</tr>
<tr>
<td></td>
<td>Peak to Sidelobe Ratio</td>
<td>-22.8385</td>
<td>-12.6818</td>
</tr>
<tr>
<td></td>
<td>Integrated Sidelobe Ratio</td>
<td>-16.6495</td>
<td>-9.3740</td>
</tr>
</tbody>
</table>
Chapter 4

Component Parameter Study

The Matlab GUI is designed such that every nonlinear parameter for each component can be modified. The GUI is especially useful in determining upper and lower bounds on those nonlinear parameters. Each nonlinear component has certain limitations associated with it. In the following sections, several parameters to each nonlinear component are changed, one at a time. The entire end-to-end simulation is then run. An analysis of the range-Doppler side lobe calculations are performed, along with component saturation and compression limitations. Please note that all of the simulations run in this study are using three frequency branches of 200, 300, and 400 MHz, a stationary target, and for the side lobe calculations, the average over the three branches was used.

4.1 Digital-to-Analog Converter

Four nonlinear parameters of the digital-to-analog converter were tested. The first three parameters tested were the offset error, the full scale gain error, and the integral nonlinearity, none of which produced any unwanted effects in the range-Doppler side lobe calculations. The offset error ranged in value from 0.02 to 1 while the corresponding peak to side lobe ratio (PSLR) and integrated side lobe ratio (ISLR) averaged a change of only 0.2328 and 0.105 dB, respectively. The full scale gain error ranged from 0.1 to 25, with average changes in PSLR and ISLR being 0.3102 and 0.3150 dB, respectively. The integral nonlinearity values ranged from 1 to 30, only changing the PSLR by 1.1792 dB and the ISLR by 0.3507 dB, on average. The changes in these three nonlinear DAC parameters also did not effect the cross-correlation of the signal in any way.
The final DAC parameter to be tested was the noise spectral density. Unlike the other three nonlinear parameters, changes in the noise spectral density did alter the range-Doppler calculations as well as the signal detection. Figure 4.1 shows the plots of the range PSLR and ISLR versus the noise spectral density. Figure 4.2 shows the changes in the Doppler PSLR and ISLR due to changes in the noise spectral density of the nonlinear DAC. Between -100 and -95 dBFS, the side lobe ratios significantly increase. However, signal detection at a noise spectral density level of -90 dBFS can still take place.

Figure 4.1: DAC: Changes in range PSLR and ISLR due to changing the noise spectral density

False detection of the received signal happens when the noise spectral density of the DAC is approximately -80 dBFS. In Figure 4.3, it can be seen that the third branch produces a false target detection with a time delay of \(68.359 \mu s\) instead of the actual time delay value of \(65 \mu s\). It is determined that large enough positive changes in the noise spectral density of the DAC will cause increased side lobe ratios in the range-Doppler plots as well as possible false target detection.

4.2 Mixer

The first nonlinear parameter of the analog mixer that was tested was the conversion loss. The mixer used to ensure that the nonlinear simulation was working properly had a conversion loss of 9 dB. This value was decreased by 1 dB until the loss was at 5 dB, in which case the mixer
was in compression on receive with an input compression point of 6 dB. The conversion loss was then increased from the initial starting value in increments of approximately 5 dB. Around 20 dB, the range side lobe calculations noticeably changed while the Doppler did not change much at all. Figure 4.4 shows both the range PSLR and the range ISLR plots. The PSLR ranges in value from -30 dB to approximately -24 dB, at conversion losses of 6 and 25 dB, respectively. The ISLR ranges from -22 dB to -16 dB over the same range of conversion loss.

Though the range side lobe calculations significantly changed, it can be seen in Figure 4.5 that the Doppler side lobe calculations did not. The PSLR and ISLR do appear to increase with the increased value in mixer conversion loss, however the increase is on the order of only 1 dB.

As the conversion loss was increased, the mixer did not saturate or compress though the range-Doppler side lobe levels changed. However, problems did arise in the cross-correlation detection. Figure 4.6 shows the cross correlation plots for all three branches with a mixer conversion loss of 30 dB. The side lobes are clearly larger than the side lobe levels in the baseline and nonlinear simulations discussed in the previous chapters. To further investigate the overall effect of the increase in conversion loss on the side lobe levels of the cross-correlation, the parameter was increased again to 40 dB. Figure 4.7 shows the resulting cross-correlations. The cross-correlation plot of branch 2 produces a false target detection. The side lobe levels are greater than that of the main lobe causing a false target detection. Therefore, a conversion loss value of less than 6 dB causes compression.
Figure 4.3: Cross-correlation plots with the DAC noise spectral density = -80 dB

Figure 4.4: Mixer: Changes in range PSLR and ISLR due to changing the conversion loss

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Figure 4.5: Mixer: Changes in Doppler PSLR and ISLR due to changing the conversion loss within the mixer, while a conversion loss of greater than approximately 30 dB will cause false target detection in the signal cross-correlation.

The second parameter of the mixer that was tested was the noise figure. The noise figure did not seem to have very much effect on the range-Doppler side lobe calculations nor the cross-correlation. The simulation was run for noise figures ranging from 0 to 40 dB. The PSLR values fell anywhere between -30.0737 and -29.6552 dB while the ISLR values fell between -22.3816 and -21.5939 dB, both ratios having a range of values less than 1 dB. Other nonlinear mixer parameters that could be tested in the future include both the output $2^{nd}$ and $3^{rd}$ order intercept points and the output compression point as well as the isolation values of the analog in and the LO in signal.

### 4.3 Amplifier

Three nonlinear parameters were tested in the non-ideal amplifier. The three parameters tested were the amplifier gain, output compression point, and the noise figure. Since both the non-ideal amplifier on transmit and receive use the same nonlinear Matlab model, only the transmit amplifier was tested. The amplifier on receive could be tested in the same way, if needed. It is found that none of the three parameters had any effect on either the cross-correlation or the range-Doppler metrics. Though as the gain to the amplifier model was increased, the amplifier approached saturation. Detection was
Figure 4.6: Cross-correlation plots with the mixer conversion loss = 30 dB
Figure 4.7: Cross-correlation plots with the mixer conversion loss = 40 dB
still possible, however, and the side lobe calculations only differed by 1dB for gains from 0dB to 60dB when saturation occurred. The output compression point simply showed the point at which the non-ideal amplifier was in compression, with no noticeable changes in the PSLR and ISLR values of the range-Doppler plots. As seen in the mixer also, the noise figure did not seem to have any effect on either the cross-correlation or the side lobe calculations.

4.4 Analog-to-Digital Converter

Finally, the nonlinear parameters of the analog-to-digital converter were tested. The parameters are the offset error, the full scale gain error, and the integral nonlinearity; three of the same parameters as in the nonlinear DAC. The ADC, however, does not have a nonlinear noise spectral density parameter as the DAC did. Since the functionality of ADCs and DACs is very similar, a theory that the three aforementioned parameters will not affect the range-Doppler side lobe calculations or the cross-correlation plots is proposed. After running the simulation with several different values of each of the three parameters, the proposed theory is validated. Neither the offset error, the full scale gain error, nor the integral nonlinearity of the ADC affect either the side lobe calculations or the signal detection. The PSLR and ISLR range of values for all three nonlinear parameters is less than 1 dB.

Please note, though, that in order for the output of the ADC to not get clipped, the correct minimum and maximum input voltages need to be put into the ADC module. These maximum and minimum values are found by initially running the simulation and plotting the output of the mixer low pass filter on receive, the signal that enters the ADC. Observe the amplitude of the low pass filter output. Those positive and negative amplitudes are now the correct maximum and minimum input voltage parameters.
Chapter 5

Conclusion

5.1 Matlab Software Development

A user friendly graphical interface in Matlab was successfully developed that simulates both the transmit and receive signal flow of a radar system. The interface allows multiple transmit frequency channels and the ability to control the waveform for each channel. The interface also allows the user the capability to control the timing of the waveform as well as the pulse frequency. An important aspect of the Matlab GUI is the fact that hardware components can easily be interchanged. The interface also allows for the use of either an ideal or non-ideal component. If a non-ideal component is chosen, the parameters of the component which contribute to its nonlinearities can be altered according to the hardware specifications.

An end-to-end baseline radar system was simulated using only ideal, or linear components. The goal of the baseline simulation was to ensure that ideal hardware components would adequately detect a time delayed signal as well as produce ambiguity plots with small peak to side lobe and integrated side lobe ratios. Using the interface described above, an end-to-end baseline simulation was successfully implemented. The ideal simulation did have the ability to accurately detect a time delayed signal using cross-correlation. The simulation also produced range-Doppler plots with sufficient side lobe ratios.

After the baseline simulation was complete, an entire nonlinear system simulation was implemented. The nonlinear simulation used only nonlinear hardware component models. A similar
goal was required of the non-ideal simulation. Again, using the Matlab GUI and hardware component modules with the ability to change the nonlinearities of each component, an end-to-end radar system was simulated. The initial goal of the nonlinear simulation was met in that a time delayed signal was accurately detected, and acceptable peak to side lobe and integrated side lobe ratios of the range-Doppler plots were produced.

5.2 Nonlinear Parameter Study

Once the baseline as well as the nonlinear simulations were working properly, the usefulness of the interface was tested. The interface allows for interchanging of nonlinear hardware components such that the detection properties of each individual component can be tested. Nonlinear parameters of the DAC, analog mixer, ADC, and amplifier were changed, and the detection process and range-Doppler ambiguity of each was analyzed.

It was found that the only nonlinear parameter tested that affected the performance of the digital-to-analog converter was the noise spectral density. As the noise spectral density was changed, the side lobe calculations for both range and Doppler changed significantly and there was false target detection. The mixer was tested next and the only nonlinear parameter that caused any performance changes was the conversion loss. Changing the conversion loss caused false target detection and the range side lobe calculations changed significantly while the Doppler did not. Finally, the analog-to-digital converter and amplifier were tested. Since the ADC does not have a noise spectral density parameter, none of the nonlinear parameters tested contributed to any degradation in performance of the component. The same was true of the amplifier. The three nonlinear amplifier parameters tested did not affect the detection or the side lobe calculations of the range-Doppler plots.

5.3 Future Work

As described in the beginning of this document, the waveforms used in the performance study were binary phase-coded waveforms. In order to further study the impact of current hardware in the future, other waveforms of interest can be implemented and studied. Such waveforms include Barker codes and complementary codes. In using such waveforms as complementary codes, further differences between the non-ideal and the ideal simulation should be apparent. Since complementary
codes are intended to suppress side lobe levels to very small values, the noise contributions due to
the nonlinearities in the component models should be more noticeable.
Bibliography


[35] “GaAs MMIC SMT Double-balanced Mixer Specification Sheet”, Hittite Microwave Corporation

[36] “SMT PHEMT Low Noise Amplifier Specification Sheet”, Hittite Microwave Corporation

[37] “Digital-to-Analog Converter Specification Sheet”, Maxim

[38] “Analog-to-Digital Converter Specification Sheet”, Maxim
