# LAB1 Report

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In lab 1 We explored the basic configuration of  $e^2$  studio, ultrasound module and the basic signal processing flow of the wireless transmitted signal.

## 1 Hardware configuration

# 2 Error Source Analysis

We inspect the relative distance and try to determine the model of the error.

| Pair (cm)           | $n_{ m theory}^*$    | $n_{ m meas}$         | $\Delta n_{\mathrm{theory}}^*$ | $\Delta n_{\rm meas}$ | $\frac{\Delta n_{\mathrm{meas}}}{\Delta n_{\mathrm{theory}}^*}$ | Relative Error (%) |
|---------------------|----------------------|-----------------------|--------------------------------|-----------------------|---|--------------------|
| $20\rightarrow40$   | $184.97 {	o} 369.94$ | $231 \rightarrow 412$ | 184.97                         | 181                   | 0.979   | -2.1               |
| $40 \rightarrow 60$ | $369.94 {	o} 554.91$ | $412 \rightarrow 598$ | 184.97                         | 186                   | 1.006   | +0.6               |
| $60 \rightarrow 80$ | $554.91 {	o} 739.88$ | $598 \rightarrow 773$ | 184.97                         | 175                   | 0.946   | -5.4               |
| 80→100              | $739.88 {	o} 924.86$ | $773 \to 953$         | 184.97                         | 180                   | 0.973   | -2.7               |

Table 1: Relative Distance Comparison (Measured vs. Theoretical)

We retrieve back  $n_{\rm meas}$  by aligning the starting sample by the maximum Tx data. We plot out the

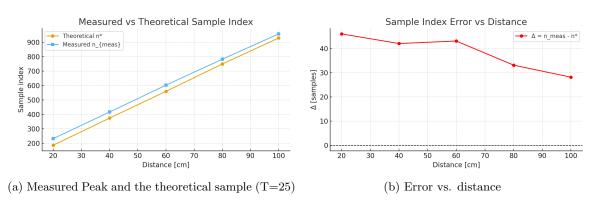


Figure 1: Comparing robust and non-robust design for linear precoder

$$\Delta = a + bn_{\text{meas}}$$

With linear regression we found  $a \simeq 53.14, b \simeq -0.02469,$  and we therefore perform the correction on the measurements

# 3 Signal Processing

some discussion below:

| True Distance (cm) | $n_{\rm meas}$ | $\hat{n}$ (Corrected) | $n^*$ (Theory) | Error After Correction (cm) |
|--------------------|----------------|-----------------------|----------------|-----------------------------|
| 20                 | 231            | 183.56                | 184.97         | -0.15                       |
| 40                 | 412            | 369.04                | 369.94         | -0.10                       |
| 60                 | 598            | 559.39                | 554.91         | +0.48                       |
| 80                 | 773            | 738.65                | 739.88         | -0.13                       |
| 100                | 953            | 922.23                | 924.86         | -0.28                       |

Table 2: Corrected Sample Index and Residual Distance Error (Linear Model)

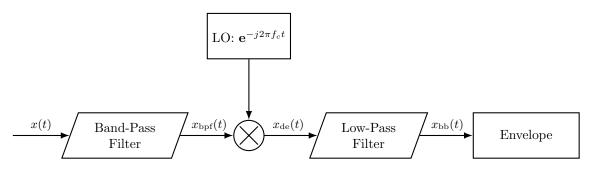


Figure 2: block diagram: signal  $\rightarrow$  BPF  $\rightarrow$  demod (mixer)  $\rightarrow$  LPF  $\rightarrow$  envelope.

• Correct "Detection" threshold. The physically correct measure of the time of flight (TOF) would be the *wavefront* of the received signal.

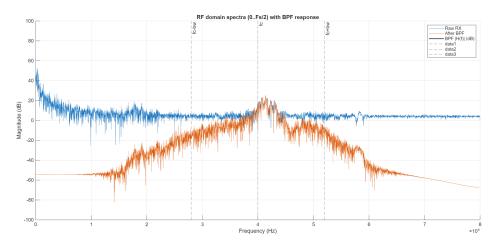


Figure 3: bpf

### **Bandpass Filtering**

Given the sampled received signal x(t), we first apply a bandpass filter to isolate the signal components near the carrier frequency. The bandpass-filtered signal is

$$x_{\rm bpf}(t) = x(t) * h_{\rm bpf}(t) \tag{1}$$

where \* denotes convolution and  $h_{\rm bpf}(t)$  is the impulse response of the bandpass filter.

Filter Design. We choose a 6<sup>th</sup>-order IIR Butterworth bandpass filter with half-power frequencies

$$f_{\text{bp},1} = f_c - 2f_w, \qquad f_{\text{bp},2} = f_c + 2f_w,$$

where the carrier frequency  $f_c=40~\mathrm{kHz}$  and the signal bandwidth is

$$f_w = \frac{1}{T_{\rm burst}} \approx 5 \text{ kHz}.$$

To ensure the filter design remains within valid frequency bounds, we compute:

$$\begin{split} \mathtt{bp\_bw} &= \max \bigl( 2f_w, \ 12 \ \mathrm{kHz} \bigr), \\ \mathtt{bp\_f1} &= \max \bigl( 10, \ f_c - \mathtt{bp\_bw} \bigr), \\ \mathtt{bp\_f2} &= \min \bigl( \frac{F_s}{2} - 10, \ f_c + \mathtt{bp\_bw} \bigr), \end{split}$$

where  $F_s$  is the sampling frequency.

#### MATLAB Implementation. The filter is implemented and applied using MATLAB as follows:

```
bp_bw = max(2*fw, 12e3); % Bandwidth selection
bp_f1 = max(10, fc - bp_bw); % Lower cutoff frequency
bp_f2 = min(Fs/2-10, fc + bp_bw); % Upper cutoff frequency

dbp = designfilt('bandpassiir','FilterOrder',6, ...
    'HalfPowerFrequency1', bp_f1, ...
    'HalfPowerFrequency2', bp_f2, ...
    'SampleRate', Fs);

rx_bp = filtfilt(dbp, rx_dc); % Zero-phase filtering
```

This produces the zero-phase bandpass-filtered signal  $x_{\text{bpf}}(t)$ .

#### Demodulation

After bandpass filtering, we demodulate the signal by multiplying it with a complex exponential at the carrier frequency  $f_c$ :

$$x_{\rm de}(t) = x_{\rm bpf}(t) \cdot e^{-j2\pi f_c t} \tag{2}$$

In the frequency domain, this shifts the bandpass signal to baseband, centering its spectrum around 0 Hz.

```
w0 = 2*pi*fc/Fs; % Normalized carrier frequency (rad/sample)
lo = exp(-1j*w0*n); % Complex exponential for downconversion
bb = rx_bp .* lo; % Complex baseband signal
```

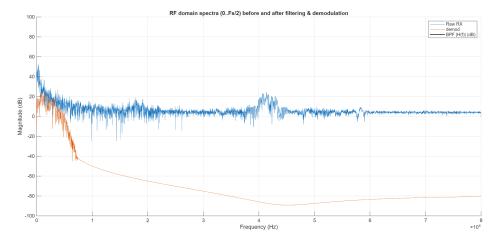


Figure 4: demod

#### Low-Pass Filtering

The downconverted signal still contains high-frequency components due to the product term. We pass  $x_{de}(t)$  through a low-pass filter to retain only the baseband component.

**Filter Design.** We use a FIR low-pass filter with passband edge at  $f_{\rm LP} = f_{\rm c}$  and stopband starting at  $1.6 f_{\rm LP}$ :

This yields the complex baseband signal  $x_{\rm bb}(t)$  with high-frequency components removed.

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### **Envelope Detection**

Since the original signal is carried by  $f_c$ , it can be expressed as

$$x(t) = A\cos(2\pi f_c t + \phi).$$

After demodulation, we have

$$\text{Re}\{x_{\text{de}}(t)\} = x(t)\cos(2\pi f_c t) = \frac{A}{2}[\cos(\phi) + \cos(4\pi f_c t + \phi)].$$

Thus, the envelope amplitude is halved. We compensate for this loss by multiplying by 2 when extracting the magnitude of the analytic signal:

```
env = 2 * abs(bb_f);
```

This produces the envelope of the baseband signal, scaled back to its original amplitude.

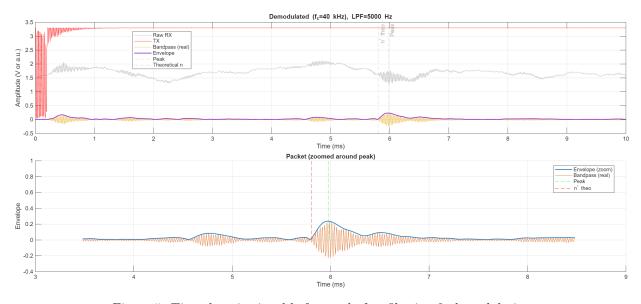


Figure 5: Time domain signal before and after filtering & demodulation

### References

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