LAB 1 – MANUAL

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Objective:

In this lab, you will learn how signals and systems are classified in communication systems. You will explore how different types of signals behave and how common system properties (linearity, time-invariance, causality, and stability) are evaluated.

Lab Outline:

1. Classification of Signals
    - Continuous vs Discrete
    - Deterministic vs Random
    - Even vs Odd
    - Periodic vs Aperiodic
    - Energy vs Power
    - Time Shift & Time Scaling
2. Classification of Systems
    - Linearity
    - Time-Invariance
    - Causality

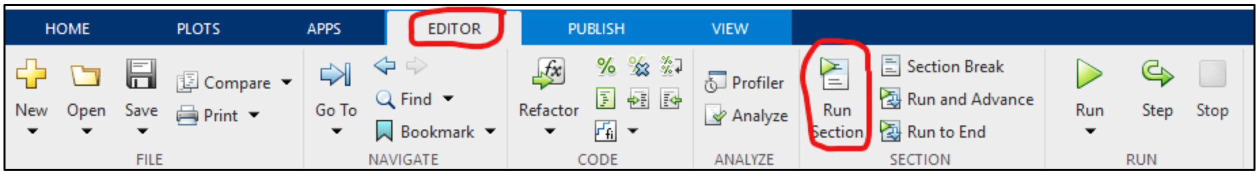
Procedure:

Step 1 – Launch MATLAB

Open '**MATLAB R2024b**' on your workstation

Step 2 – Load the Lab Script

Load the file **Lab1.m** into MATLAB. To run a block of code, use **Run Section** in the Editor panel (or press CTRL + ENTER).



Part 1 – Classification of Signals

Signals represent information-bearing waveforms transmitted from a source to a receiver. In communication systems, it is essential to classify signals based on how they vary in time, energy and randomness.

(1) Pre-Lab Setup

Run the section labeled **Pre-Lab Setup**

This initializes:

- Base sample rate fs
- Sampling interval Ts
- Total simulation time T
- Random seed for reproducibility

Explain briefly what **fs** and **Ts** represent and why reproducibility is useful in simulation.

Fs represents the sampling rate and Ts represents the sampling period. Reproducibility allows for consistent results across repeated experiments.

(2) Continuous-Time vs Discrete-Time (Sec 2.1)

Run the section labeled **Continuous-Time vs Discrete-Time.**

Observe how a 500 Hz cosine looks when sampled at:

- fs = 8 kHz (well above Nyquist)
- fs = 600 Hz (near Nyquist)

Question: Describe what you observe. What happens when **fs < 2f0?** (example, set fs=100 Hz).

Setting sampling rate well above the Nyquist rate accurately reconstructs the original signal(8KHz) with 17 samples per cycle. Setting sampling rate below the Nyquist rate (100 Hz) shows 1 sample every 6 cycles and does not accurately reconstruct the original signal.

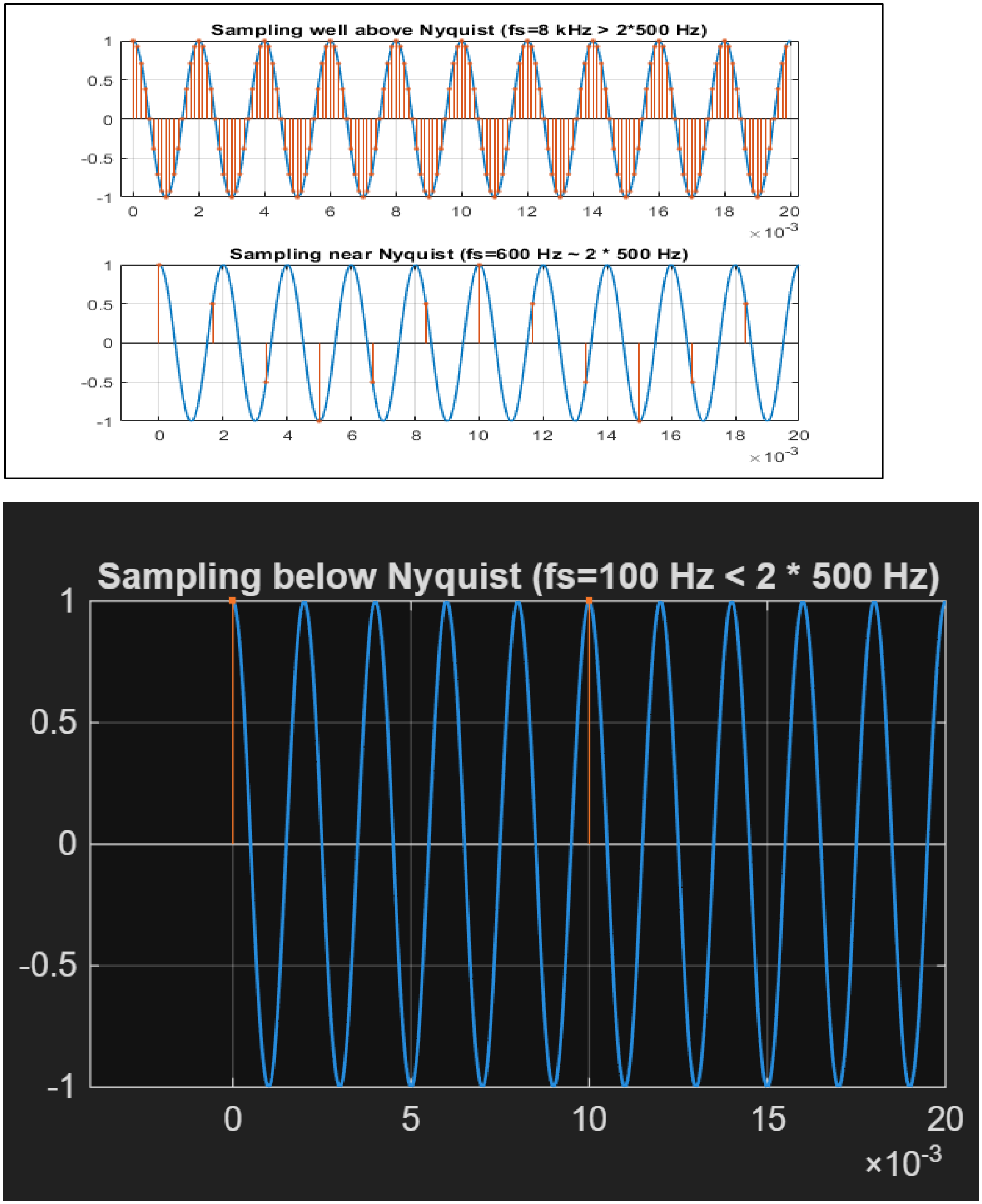


Figure 1. Continuous vs Discrete

(3) Deterministic vs Random (Sec 2.2)

Run the section **Deterministic vs Random.** The deterministic signal is fully predictable (cosine), while the random sequence varies unpredictably.

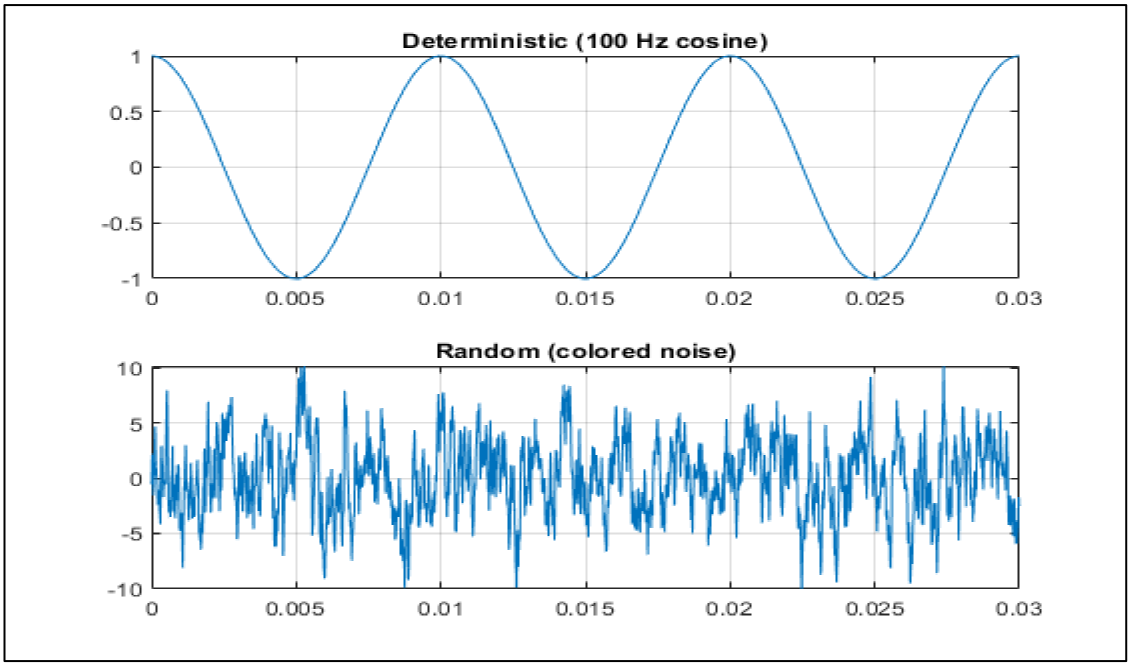
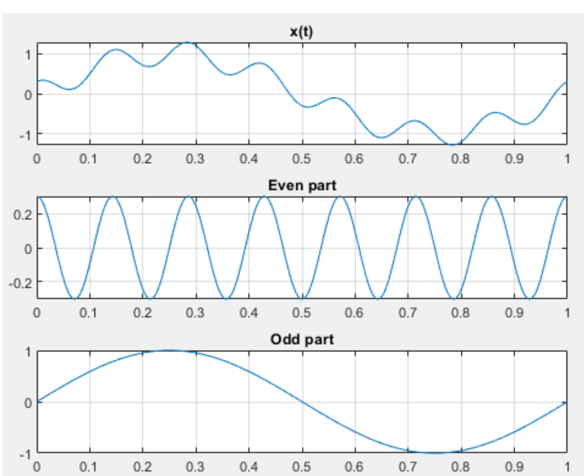


Figure 2. Deterministic vs Random

(4) Even/Odd Decomposition (Sec 2.3)

Run this section and observe how a signal can be decomposed into:

- An even component xe(t)
  - If a signal is composed of a linear combination of sine and cosine waves, then the even part is the cosine wave.
- An odd component x0(t)
  - The odd part is the sine wave.



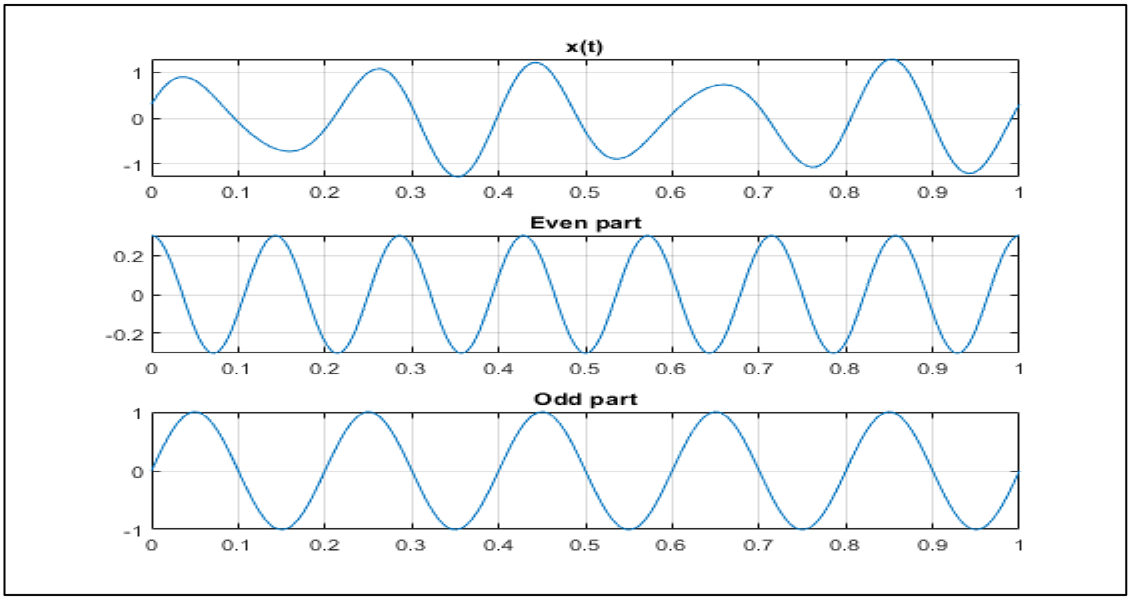


Figure 3. Even vs Odd Decomposition

(5) Periodic vs Aperiodic (Sec 2.4)

Run this section to compare signals that repeat over time vs signals localized in time.

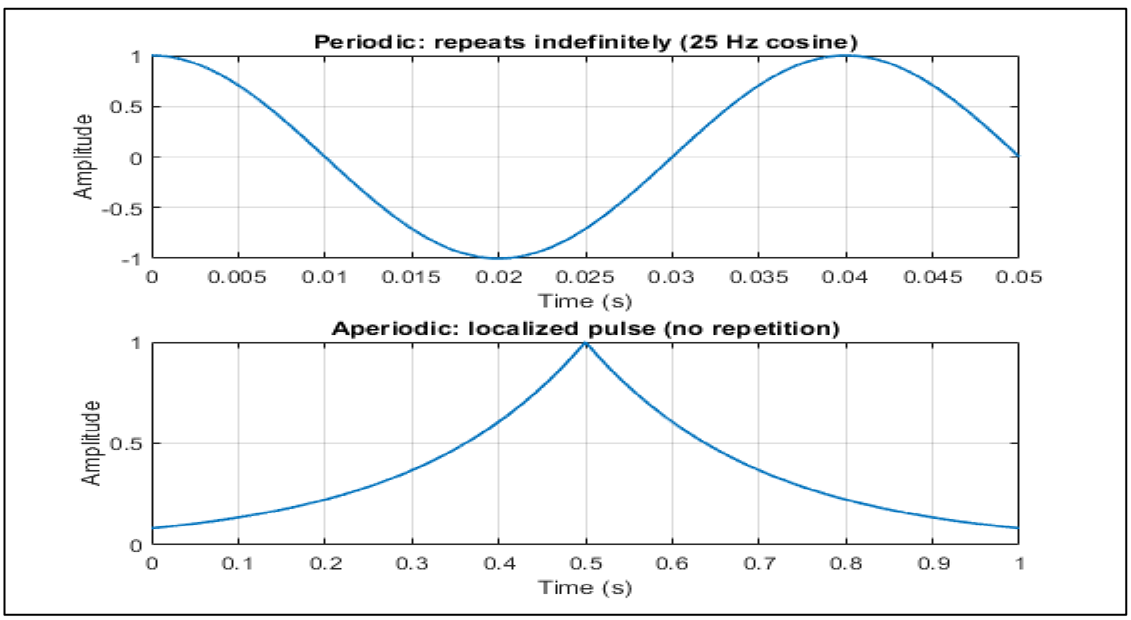
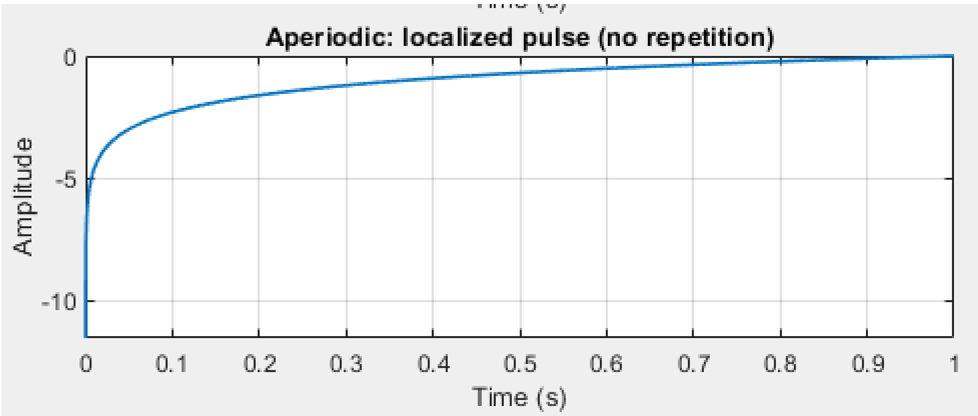


Figure 4. Periodic vs Aperiodic

Questions: Which subplot is periodic and how can you tell? Estimate the period from the plot and compute: T0, f0 = 1/T0, and ω0 = 2πf0

The top subplot is periodic since it is a sinusoidal wave, and the sine wave repeats every 0.04 seconds: f_0= 25 Hz, T_0= 0.04 s, ω_0=50π. The bottom subplot does not repeat and is not periodic.



(6) Energy vs Power Signals (Sec 2.5)

Run this section and observe how:

- Decaying pulses are **energy signals**
- Persistent sinusoids are **power signals**

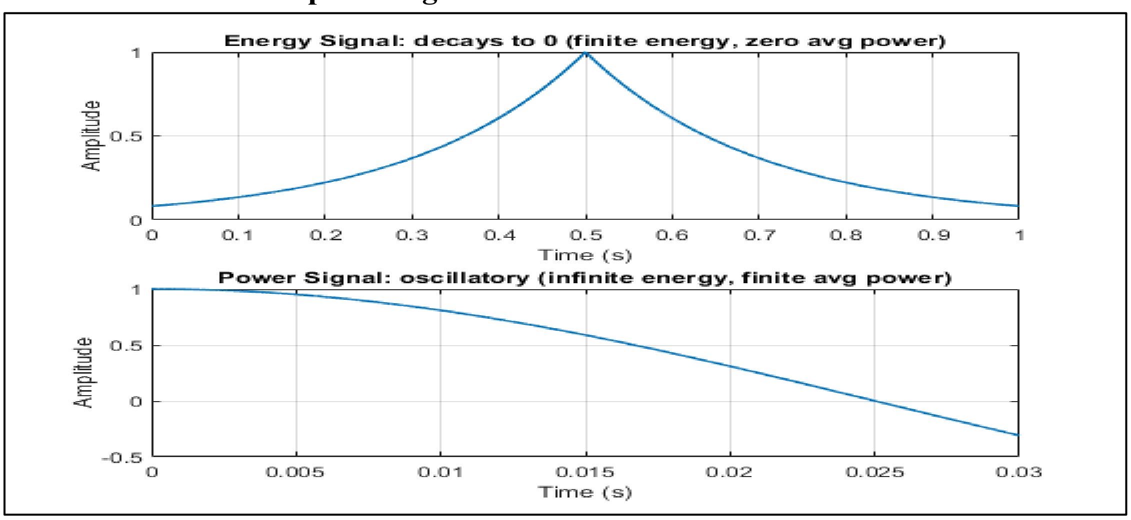


Figure 5. Energy vs Power Signal

Questions: Which signal has finite total energy and zero average power? Do energy signals typically look aperiodic? Why? Can a signal be both an energy and a power signal? Explain

The top subplot has finite total energy and zero average power. If the energy signal has finite energy, then the total energy will converge to zero. Thus, the signal will be aperiodic and not continue to repeat as time goes to infinity. A signal cannot be a power and energy signal since the signal cannot have finite and infinite energy at the same time.

(7) Time Shift & Time Scaling (Sec 2.6)

Run this section to visualize how shifting and scaling affect waveforms.

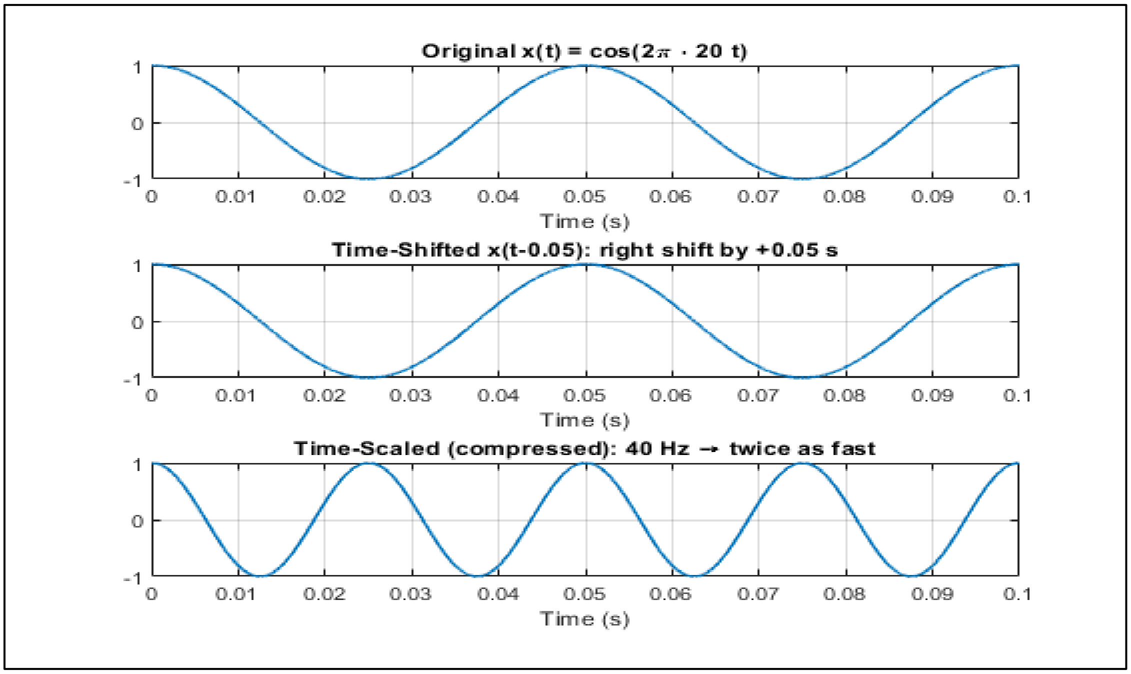


Figure 6. Time Shift & Scaling

Does x(t-0.05) shift left or right? Why? What is the period of the original and scaled signals? Which operation increases frequency and which affects phase? In communications, what does time compression correspond to?

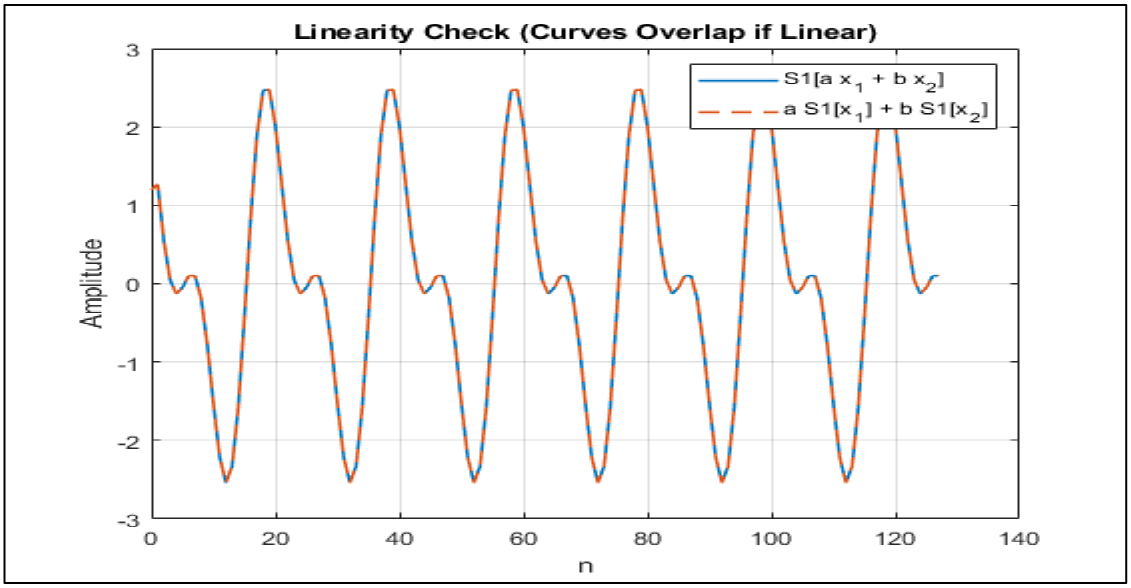
It shifts to the right because x(t-c) delays the time. The period of the original signal is 0.05 seconds and 0.025 for the scaled signal. Time-scaling will increase frequency, and time-shifting affects phase. In communications, time-compression indicates that the period is lower and sends more data in a shorter interval.

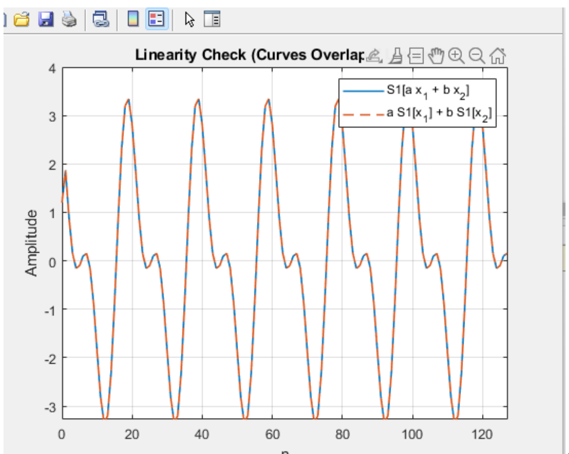
Part 2 – Classification of Systems

In this section, we examine how systems respond to inputs and how to test basic properties. We investigate how systems respond to different inputs and how to evaluate common system properties. Many communication system blocks (filters, mixers, equalizers) are modeled as linear time-invariant (LTI) systems.

(1) Linearity Test

Run the section titled **Linearity & Time-Invariance**





After we changed the scaling factor to 1, it still overlapped.

Figure 7. Linearity Test

From your observation; do the curves in the plot (lhs and rhs) overlap? What does that imply about linearity? Does shifting before vs after the system produce the same result? Based on these observations, how would you classify S1?

Yes they overlap. If the two curves overlap, then it implies that a signal is linear and follows the property of linearity. Since the signal is linear, shifting before and after produces the same result. Therefore, S1 is the changer for making sure that the input would match the output. If they are both the same, then they're linear.

(2) Time-Invariance Test

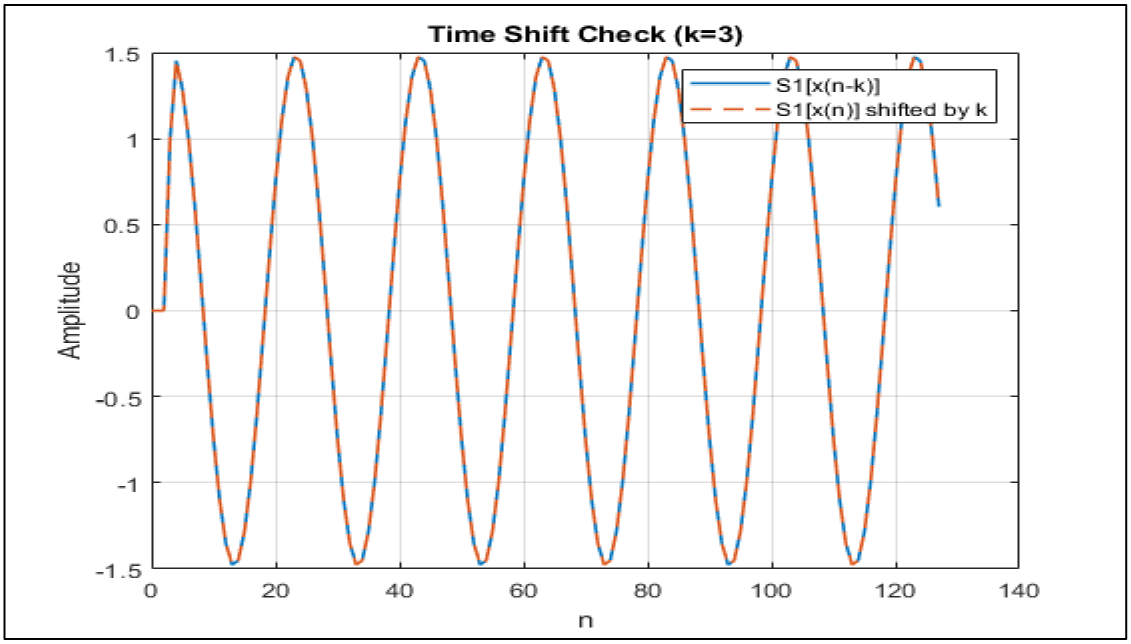
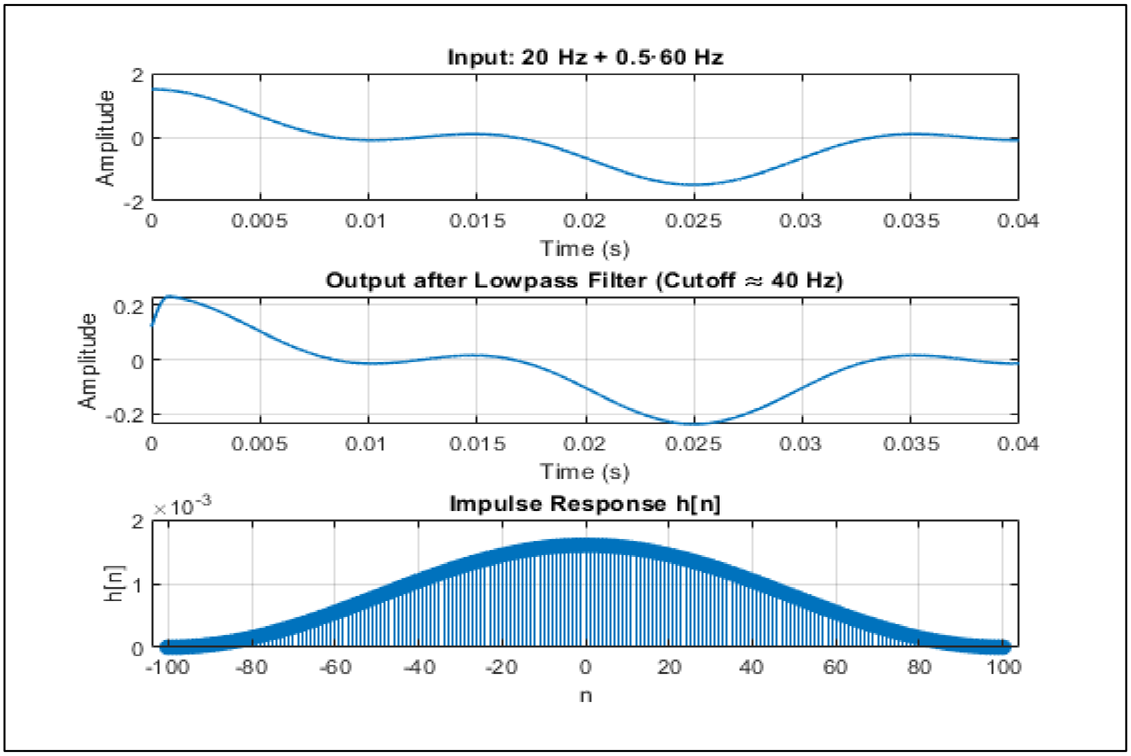


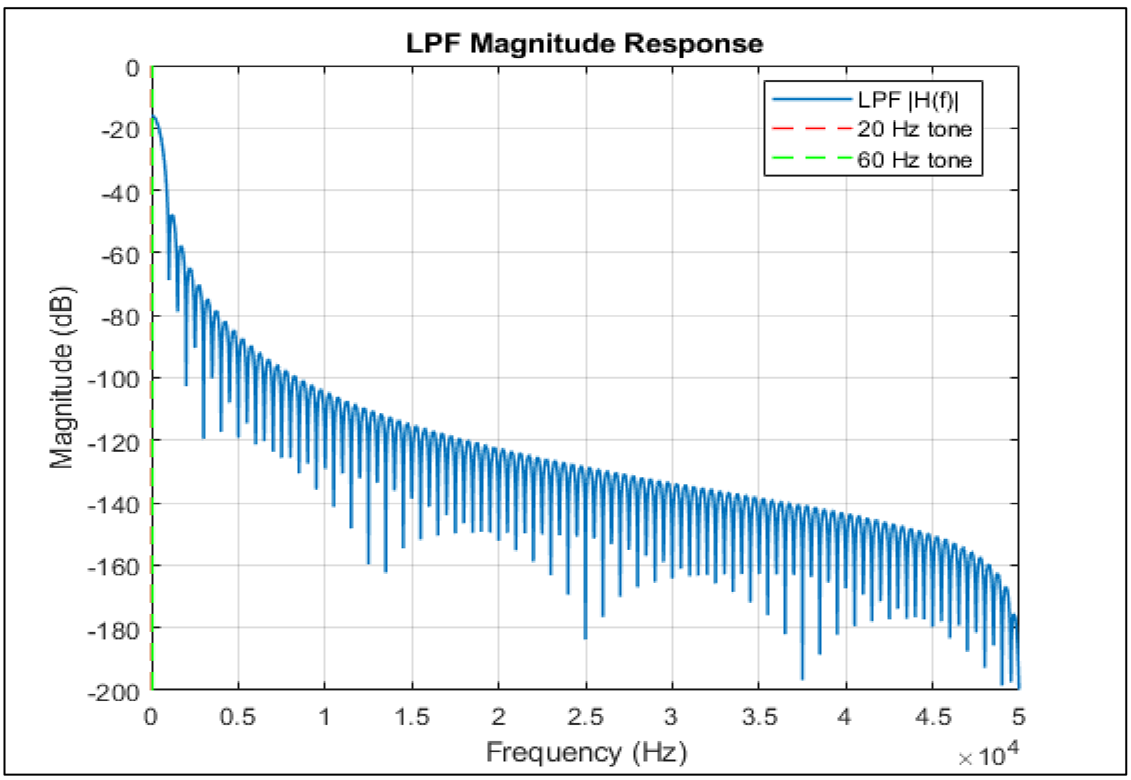
Figure 8. Time-Invariance Test

When the input is delayed by k samples, does the output also delay by k? Do the shifted waveforms align?

Delaying a linear signal by k samples delays both the input and output waveform by the same amount. The time shifted by 2 when it changed from 3 to 5. They still align.

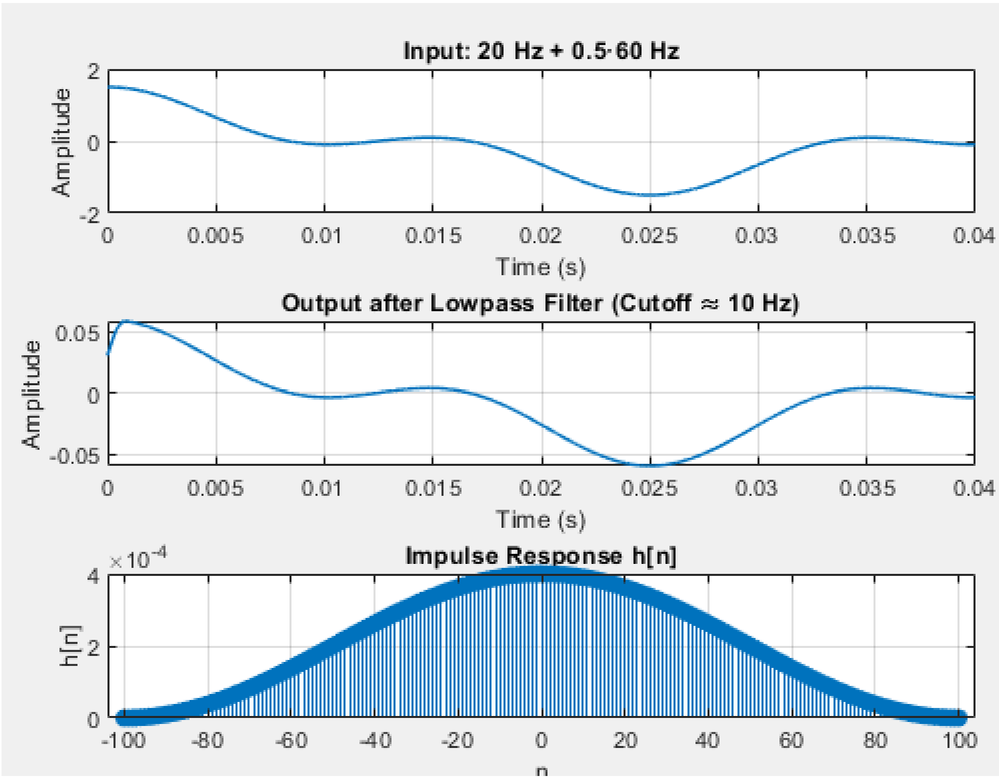
(3) Convolution Demo

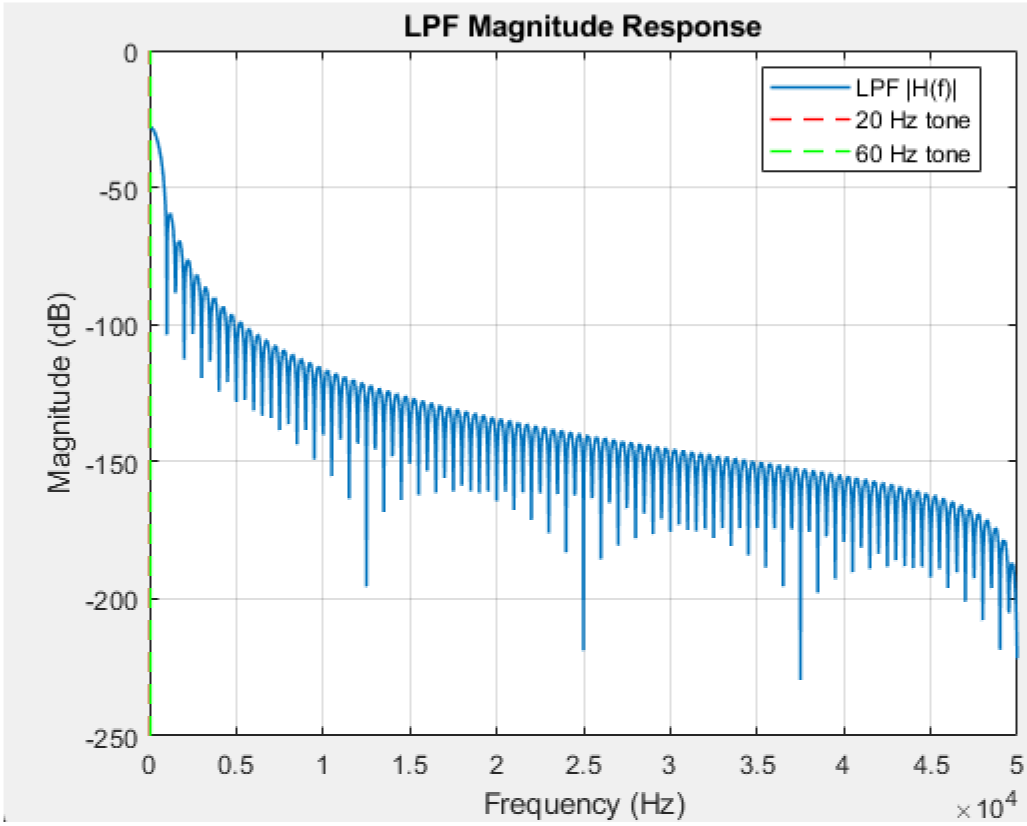




Which tone is attenuated more by the LPF and why? What happens if you move the cutoff below 20 Hz? What if you make the filter order smaller?

The 60 Hz tone is attenuated by the LPF since it is greater than the cutoff frequency of 40 Hz. If we move the cutoff to 10 Hz, then the magnitude of the LPF increases. Decreasing the filter order would increase the magnitude of the spikes within the LPF.





Part 3 — Signal Representation & Capacity

In this part of the lab, we introduce **complex baseband representation (I/Q)** and examine the effects of frequency conversion used in practical communication receivers. We also introduce **Shannon capacity** to build physical intuition for how SNR and bandwidth constrain data rate.

(1) Complex Baseband & Frequency Conversion

Run Section **4.1 (I/Q Baseband and Upconversion)** and **4.2 (Downconversion)** in the MATLAB file.

A 1 kHz baseband tone is mapped into complex I/Q components and then upconverted to a 20 kHz RF carrier. Downconversion with a local oscillator (LO) and lowpass filtering recovers the original I/Q signals.

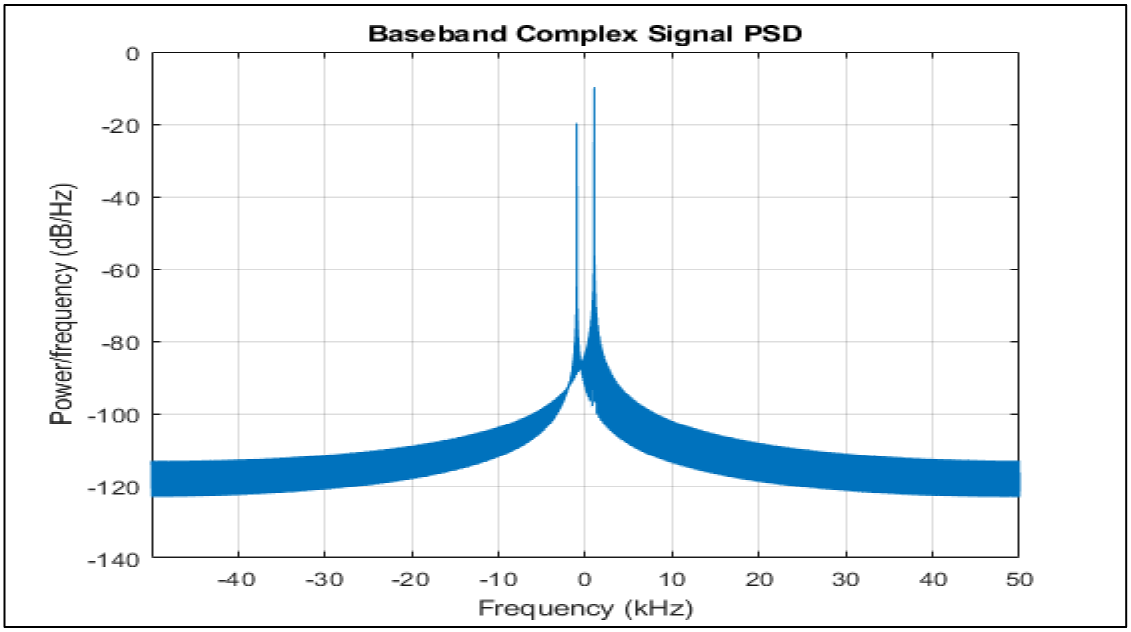


Figure 9. I(t): True vs Recovered

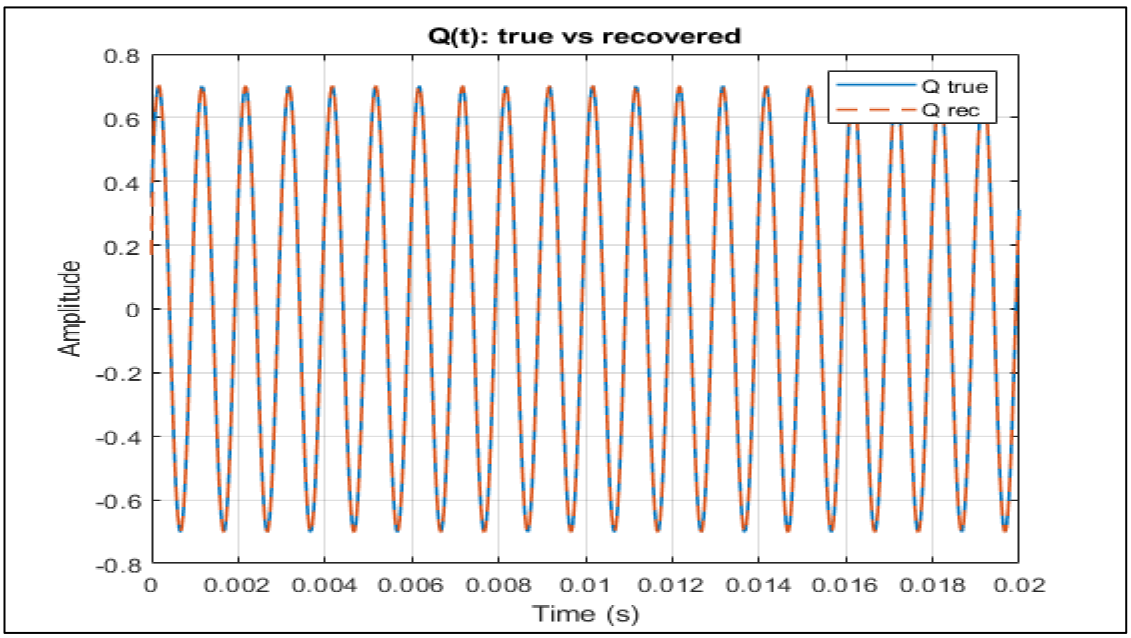


Figure 10. Q(t): True vs Recovered

Questions:

• How well does the recovered baseband match the original?

They are close to overlapping with each other.

• What do the RMSE values indicate?

It indicates an average error between the reconstructed and original signal and since it is arbitrary close to zero, the reconstruction of the signal is very accurate.

• Why can we apply a lowpass filter after mixing down to baseband?

The LPF helps remove the 20 kHz radio frequency to align the recovered signal with the original I/O signal.

(2) LO Phase Error Demonstration

Run Section **4.3 (LO Phase Error)**.

A small LO phase error rotates the recovered complex signal in the I/Q plane.

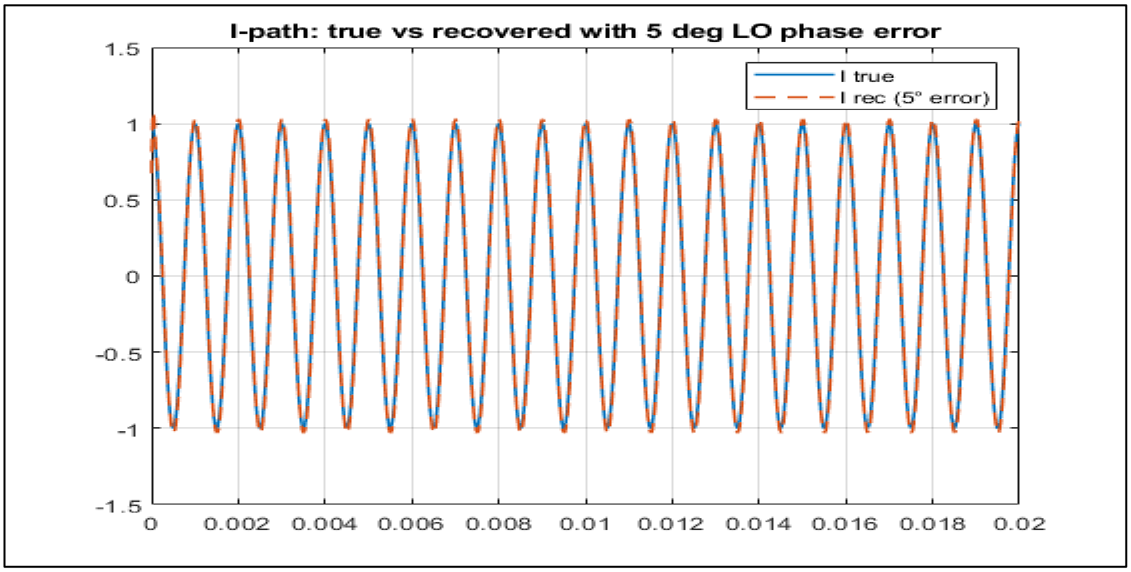


Figure 11. LO Phase Error Effect

Questions:

• How does a phase offset in the LO affect the recovered (I(t)) and (Q(t))?

A phase offset in the LO recovers the original I(t) and Q(t).

• In a constellation view, what would this error correspond to?

It shifted the original signal to the right and down.

(1) Capacity vs SNR

Run Section **5.1 (Capacity vs SNR)**.

Here we explore Shannon's capacity formula:

Capacity is plotted versus SNR for a fixed bandwidth.

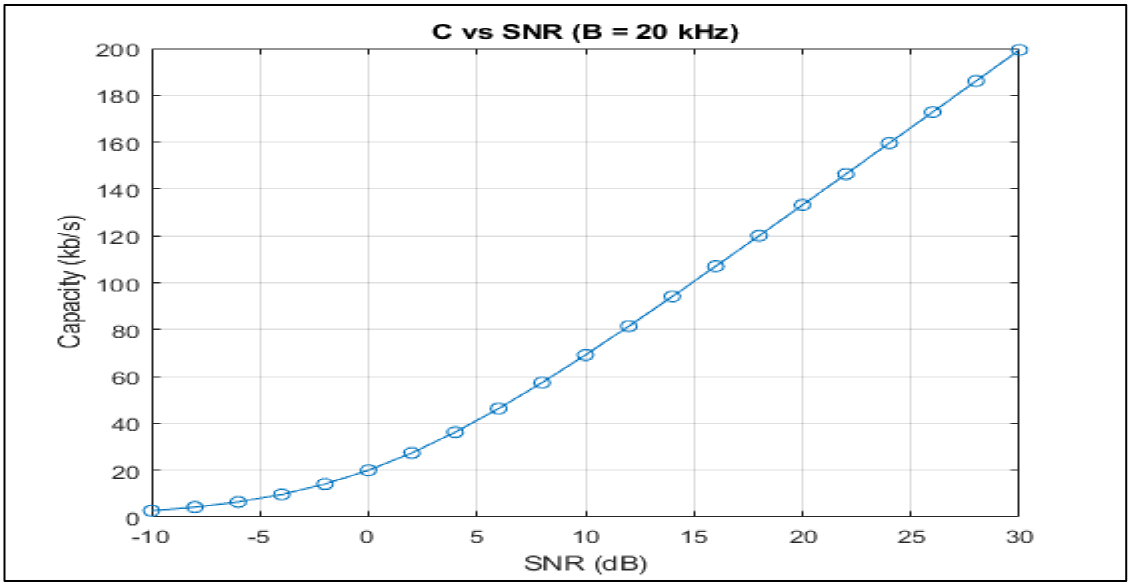


Figure 12. Capacity vs SNR

Questions:

• At very low SNR, which increases capacity more: +3 dB SNR or doubling bandwidth (B)?

Doubling the bandwidth would increase capacity more.

• How does this relate to systems operating in noise-limited regimes?

Increasing bandwidth has a greater effect than increasing signal strength.

(Hint: at low SNR, increases in B and SNR both give nearly linear gain)

(2) Capacity Heatmap: Bandwidth vs SNR

Run Section **5.2 (Capacity Heatmap)**.

The heatmap shows how data rate depends jointly on both **SNR** and **bandwidth**, revealing a region where capacity scales almost linearly with B.

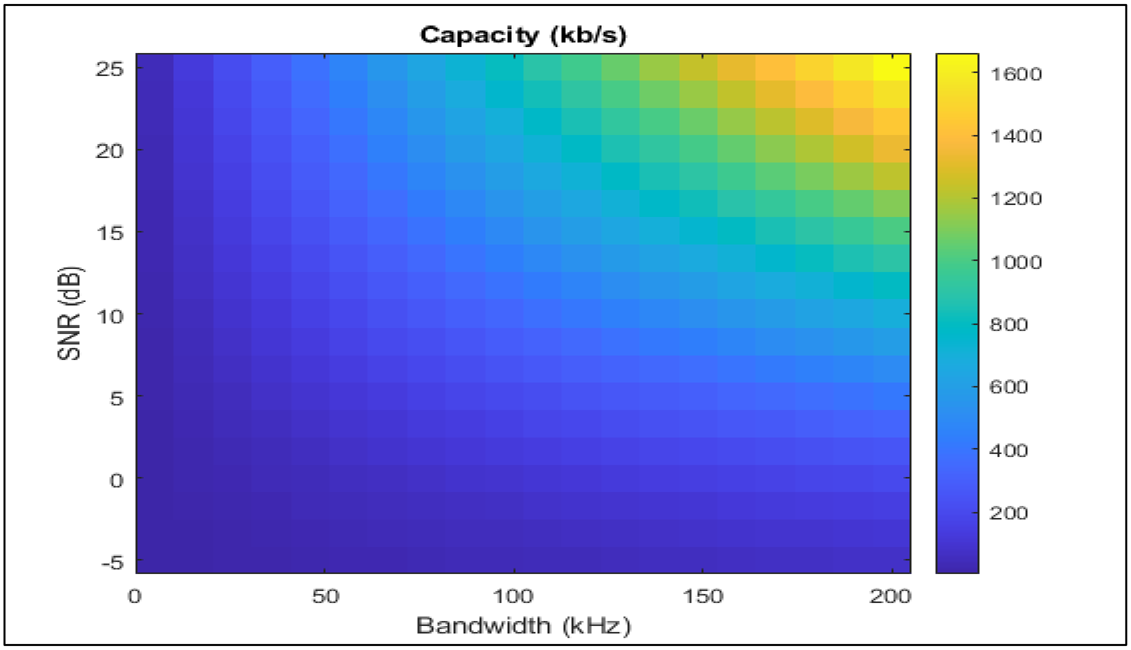


Figure 13. Capacity Heatmap

Questions:

• Why can capacity scale almost linearly with (B) at low SNR?

A low SNR implies that the signal and noise is indistinguishable and S is approximately equal to B. According to Shannon's formula, S scales approximately linear to B.

• How does this explain why ultra-wideband and spread-spectrum systems can operate with low spectral efficiency?

The capacity of the system becomes nearly proportional to bandwidth, therefore, even at weak signals the systems still functions.

• What engineering tradeoffs limit bandwidth in practical systems?

Cost and Complexity.