

Third Year

First Term

**Super-heterodyne Receiver  
Communication Project report**

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| --- | --- | --- |
| Name | Sec | ID |
| Yousef Khaled Omar Mahmoud | 4 | 9220984 |

Presented by

**Instructors: Dr.** Ahmed Hesham

**Eng.** Said Kamel

# Abstract:

In this project, we aimed to simulate a multi-stage communication system, focusing on the transmission and reception of audio signals through various stages, including baseband, RF (Radio Frequency), IF (Intermediate Frequency), and AM (Amplitude Modulation). Our system processes audio signals by modulating them into AM DSB-SC (Double Sideband Suppressed Carrier) format.

We implemented a radio simulator device with MATLAB. we focused on emulating the receiver and its filtering process, using software to handle the audio files, and explored different signal processing techniques.

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# Introduction

The primary goal of this project is to simulate the fundamental components of an analog AM communication system using MATLAB. This involves the creation of an AM modulator to encode audio signals for transmission and a super-heterodyne receiver to recover the transmitted signal. The project uses provided audio signals as input messages, simulating a real-world scenario of radio transmission and reception.

# System Design and Implementation:

An AM communication system consists of two main stages:

* The modulator: encodes the message signal onto a higher-frequency carrier for efficient wireless transmission.
* The receiver: utilizes a super-heterodyne architecture, involving mixing, filtering, and amplification, to recover the original message.

## System Implementation:

In *fig[1]*, We describe how we implement the system in MATLAB.

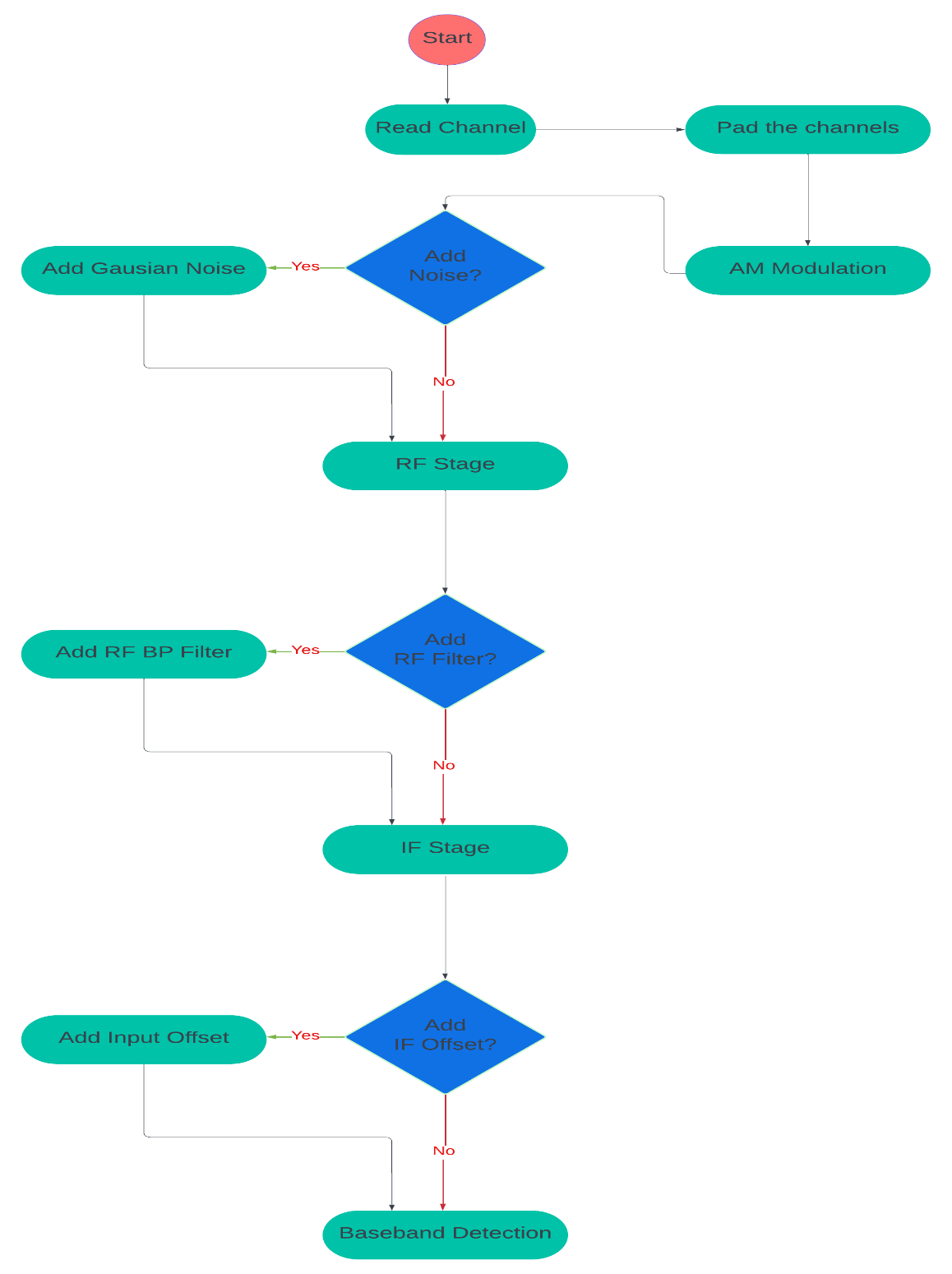


Figure Simulation Flow Chart

## System Components:

### AM Modulator:

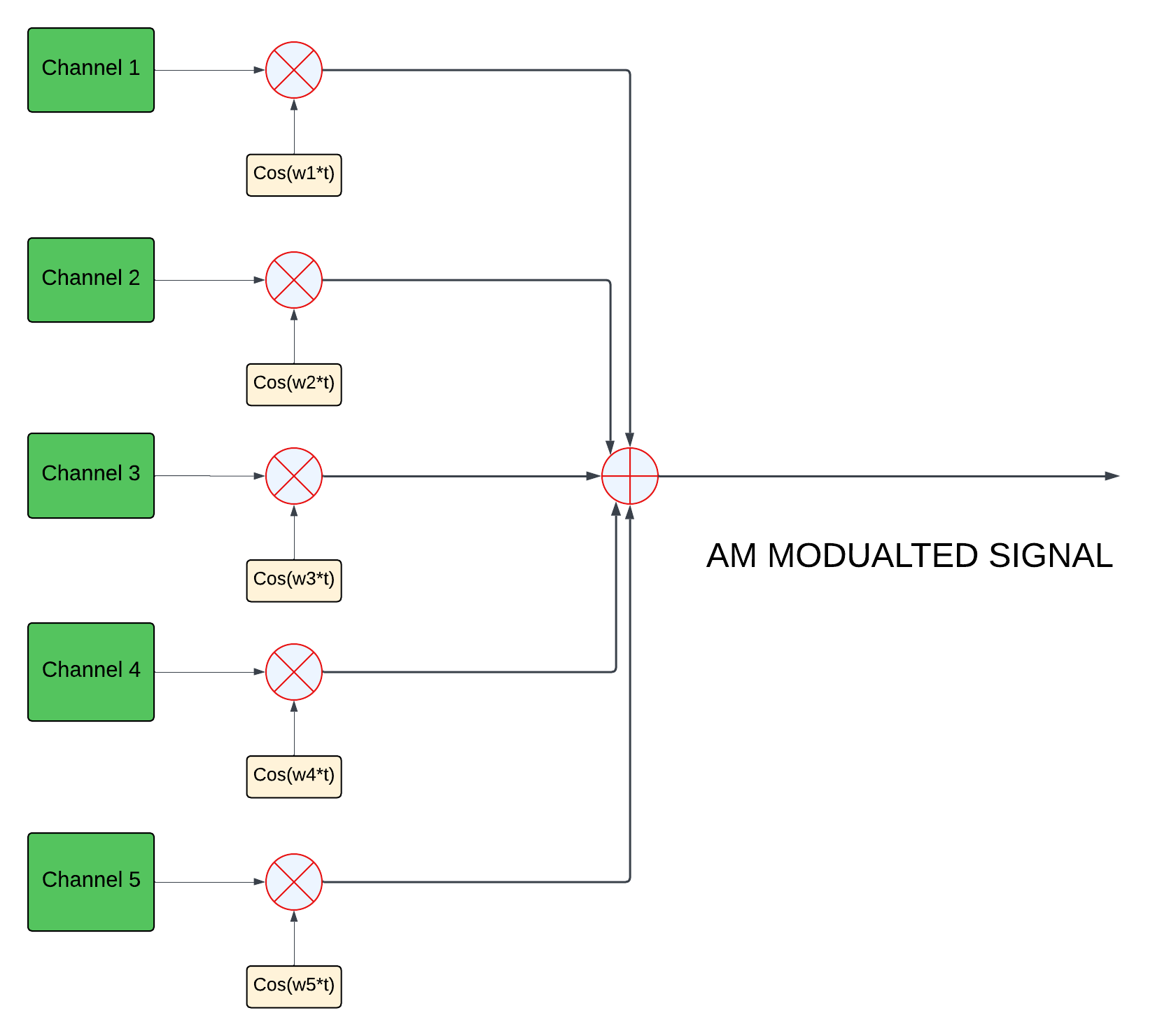


Figure AM Modulation Double Sideband Suppressed Carrier

Equation 1 Frequency Division Multiplexing

* *Where: Mn(t) is the n-th channel's message signal, Wn=100+nΔF, with ΔF=50 kHz and n as the channel index  
   (n=0 corresponds to the first signal modulated at 100 kHz)*

As shown in *fig[2]*,This project presents a MATLAB simulation of a multi-channel communication system using Frequency Division Multiplexing (FDM). The system processes five audio signals, each encoded using Amplitude Modulation (AM) onto carrier waves with unique frequencies.

The modulated signals are summed to form the FDM signal, defined mathematically in *Equ[1]*, and transmitted over RF channels and processed at the receiver.

### RF Stage:

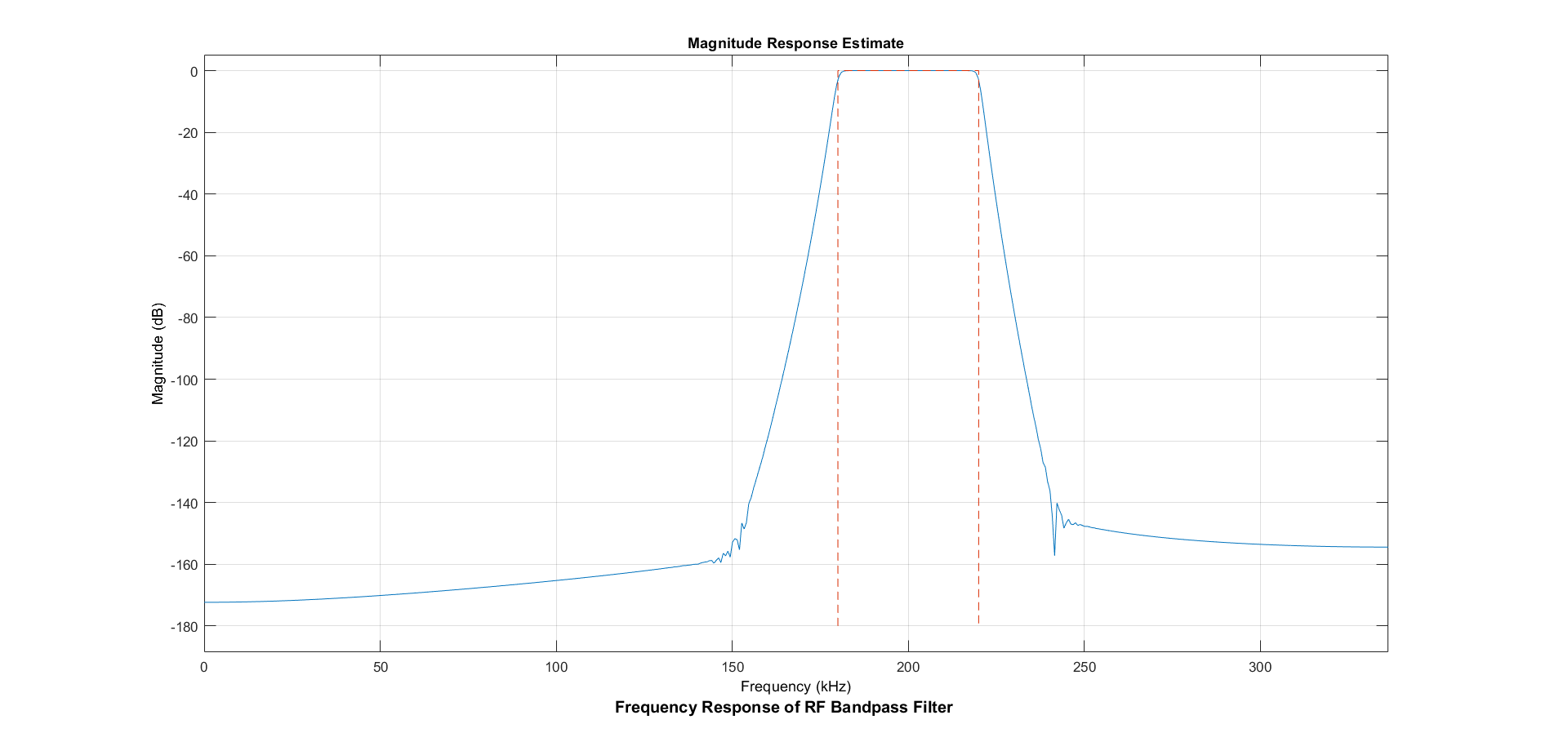


Figure Receiver Filter Frequency Response

As shown in *fig[3]*, RF filter is a band pass filter centered on the desired station’s carrier frequency ωn.

* **The purpose of the RF stage**: is to reject the intermediate frequency (IF) image and other unwanted signals outside the passband to extract the desired station.

### Mixer:

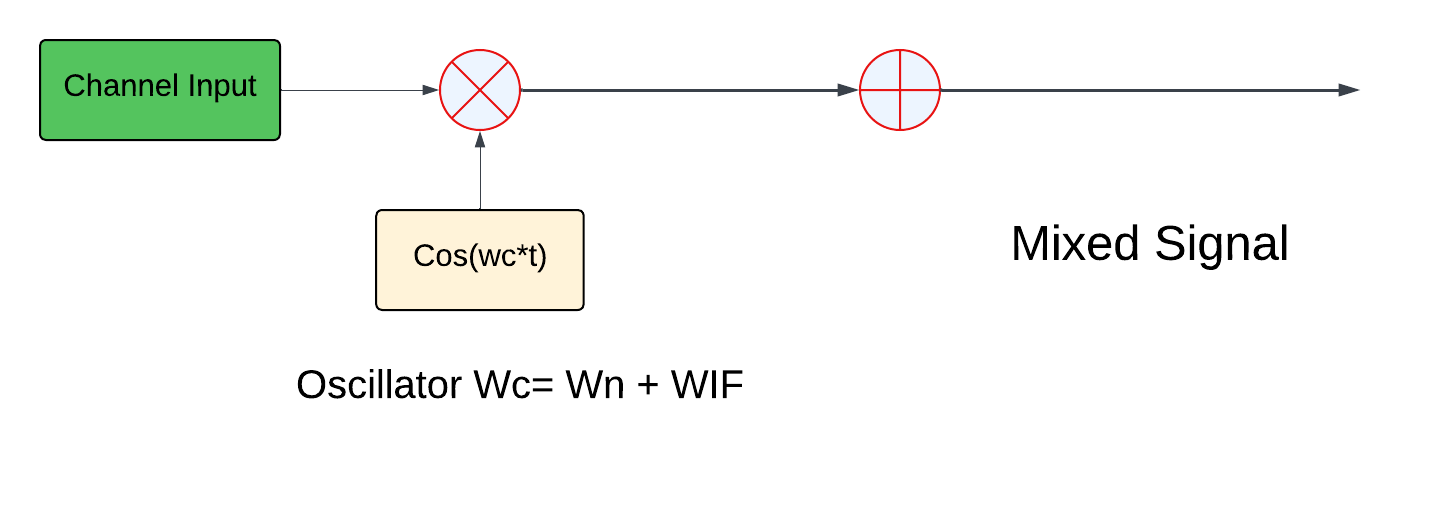


Figure Mixer with Oscillator Block Diagram

As shown in *fig[4]*, The Oscillator generates a carrier signal for mixing, shifting the desired signal to the Intermediate Frequency (IF) band.

Equation 2 Oscillator Frequency

* *Where: Wc is the oscillator frequency, Wn is the channel frequency and WIF is the intermediate frequency*

According to *Equ[2]*,The oscillator frequency is set to get the desired channel to IF stage.

### IF Stage:

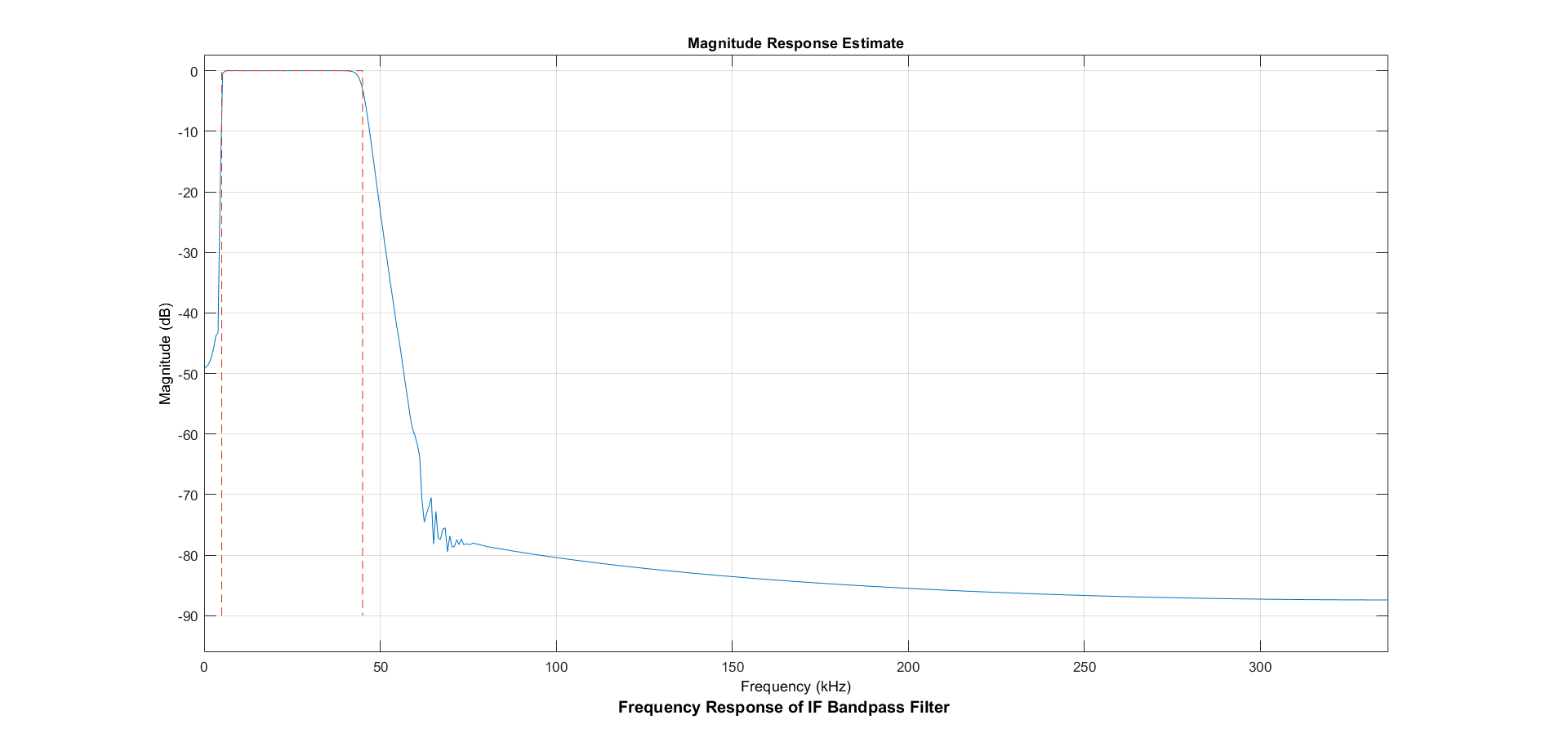


Figure Intermediate Filter Frequency Response

As shown in *fig[5]*, IF filter is a band pass filter centered on intermediate frequency (WIF).

* **The purpose of the IF stage**: filters the IF signal for further processing. By shifting the signal to a fixed WIF, it simplifies the filter design and improves selectivity.

### Baseband Detection:

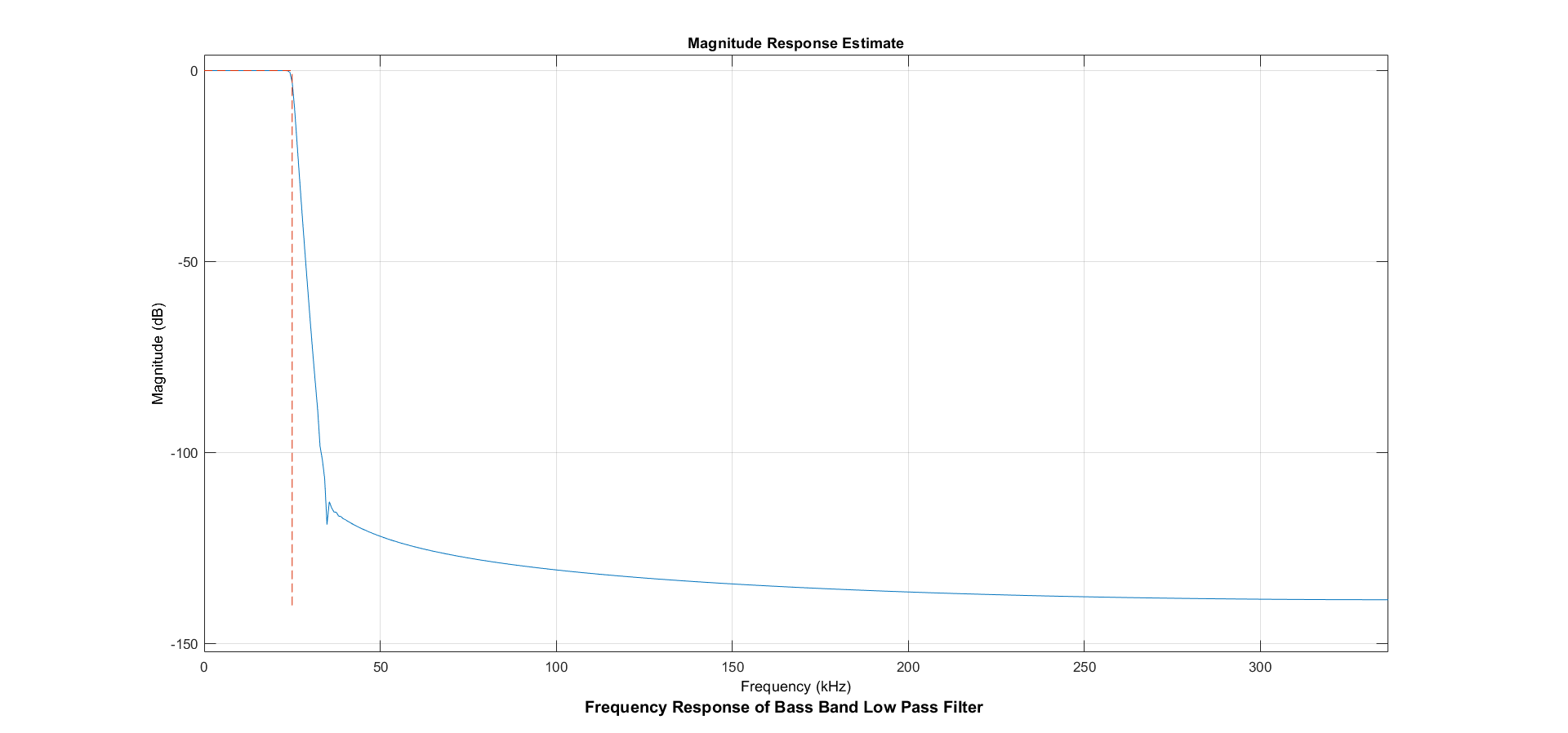


Figure Baseband Filter Frequency Response

As shown in *fig[6]*, The baseband filter a Low-Pass Filter (LPF) that removes high-frequency components after mixing the IF signal with the carrier, centered at a frequency little higher than the channel band width which is 10 kHz.

* **The purpose of the Baseband Detector:** recovering the original message by demodulating the IF signal.

The IF stage is crucial because it allows the receiver to use fixed components (filters and amplifiers) irrespective of the received carrier frequency.

# Simulation Results

In this project, we implemented and analyzed various simulations to evaluate the performance and reliability of our communication system under different conditions. The simulations included the following scenarios:

### Normal Simulation:

The system operates without any external disturbances, providing a baseline for performance analysis.

### Noise:

Additive noise was introduced to simulate real-world interference, demonstrating the system's ability to process and recover the original signal under noisy conditions.

### No RF Filter:

The impact of bypassing the RF stage was analyzed to highlight its role in filtering and isolating the desired signal.

### Frequency Offset (200 Hz):

A small offset was applied to the carrier frequency, simulating minor synchronization errors and testing the system's tolerance.

### Frequency Offset (1200 Hz):

A larger offset was introduced, representing more severe desynchronization, to analyze the limits of the system's error-handling capabilities

## Normal Simulation:

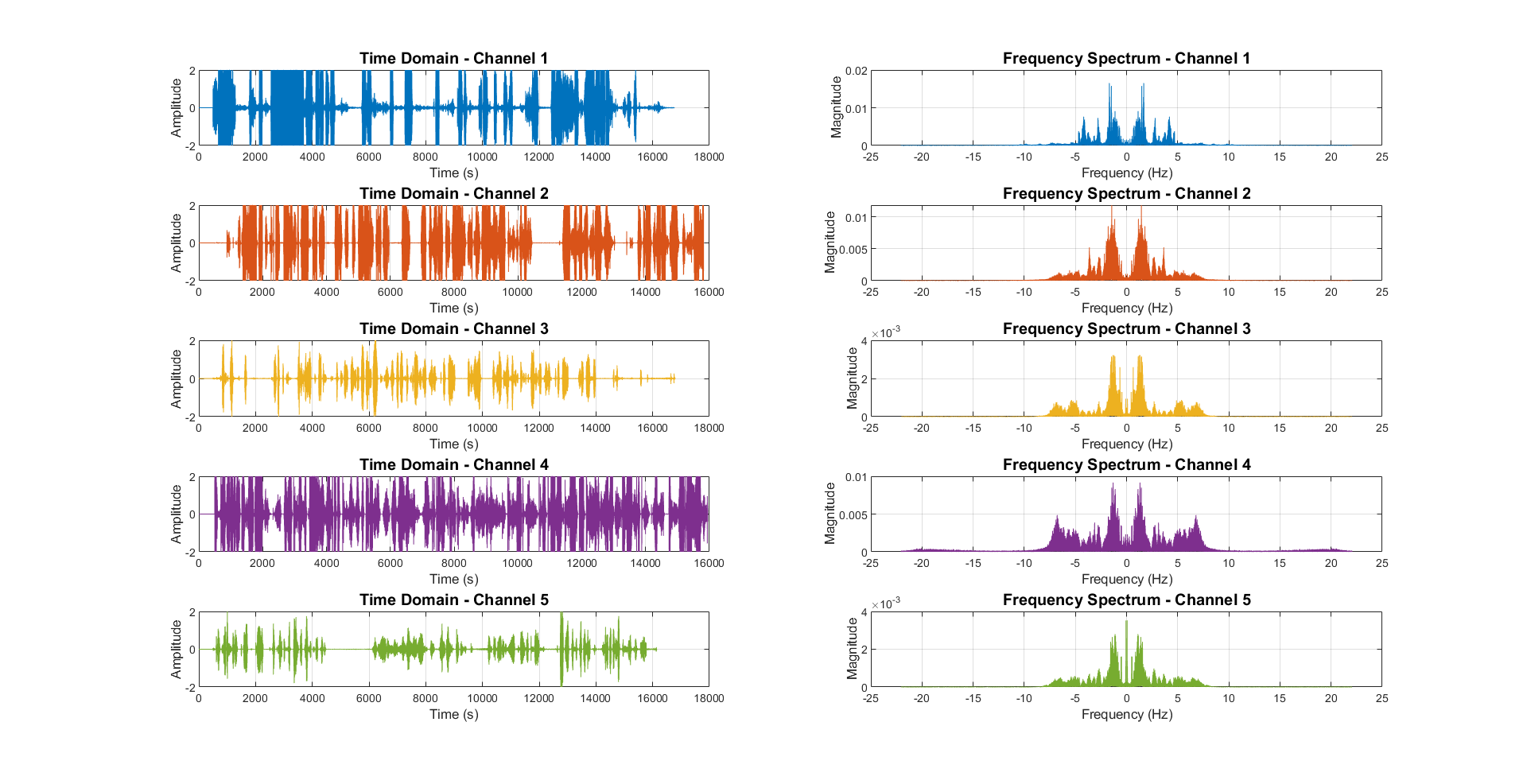
We’re going to run the simulation without any problems (Ideal Case) and going to choose channel 3.

Figure Channels plot in time and frequecny domain

As shown in *fig[6]*, All channels have a Band width approximately equal to 10 kHz.

Their length wasn’t equal so we need to pad them

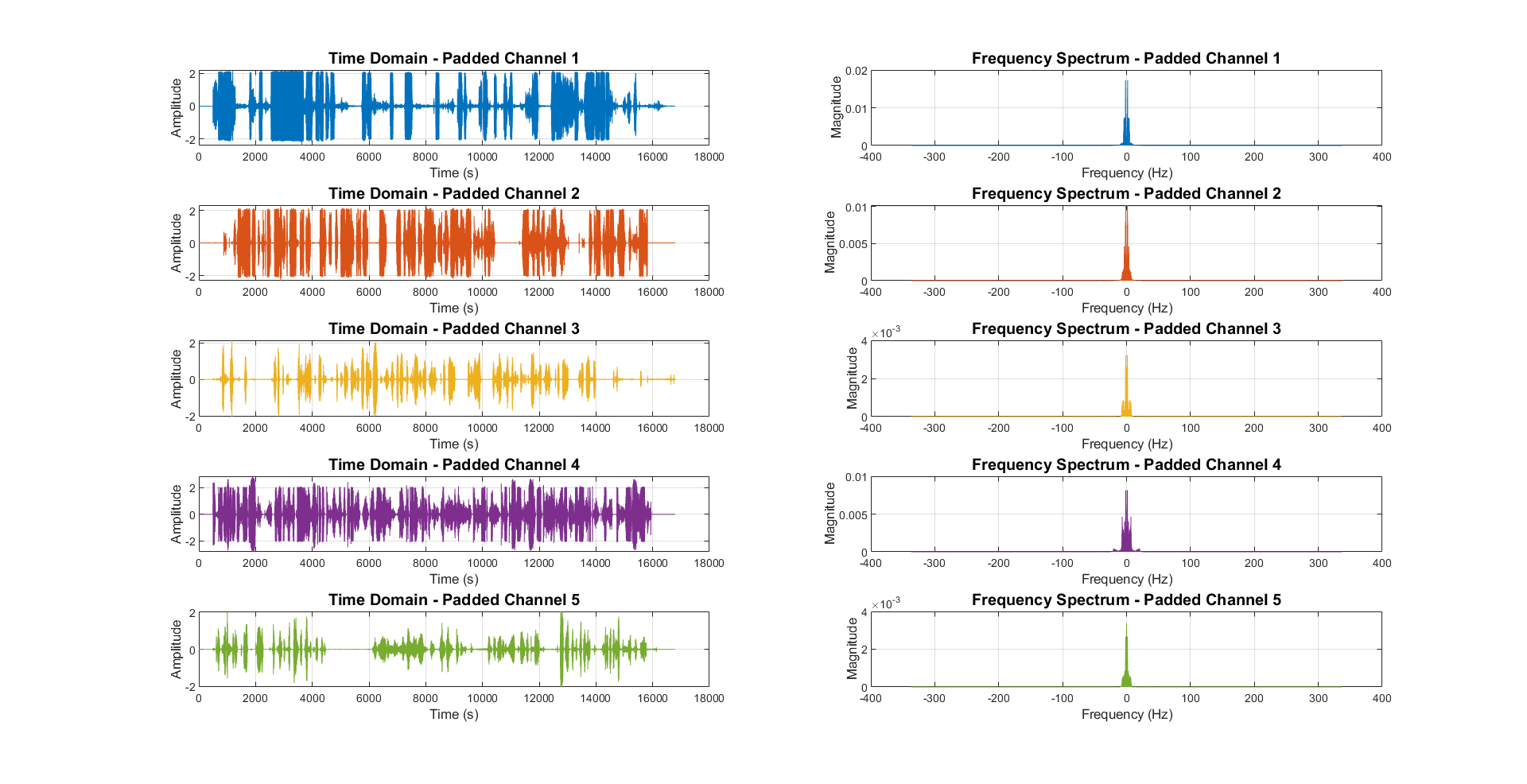


Figure Padded Channels

As shown in *fig[7]*, The channels are now equal in length, so they are padded.

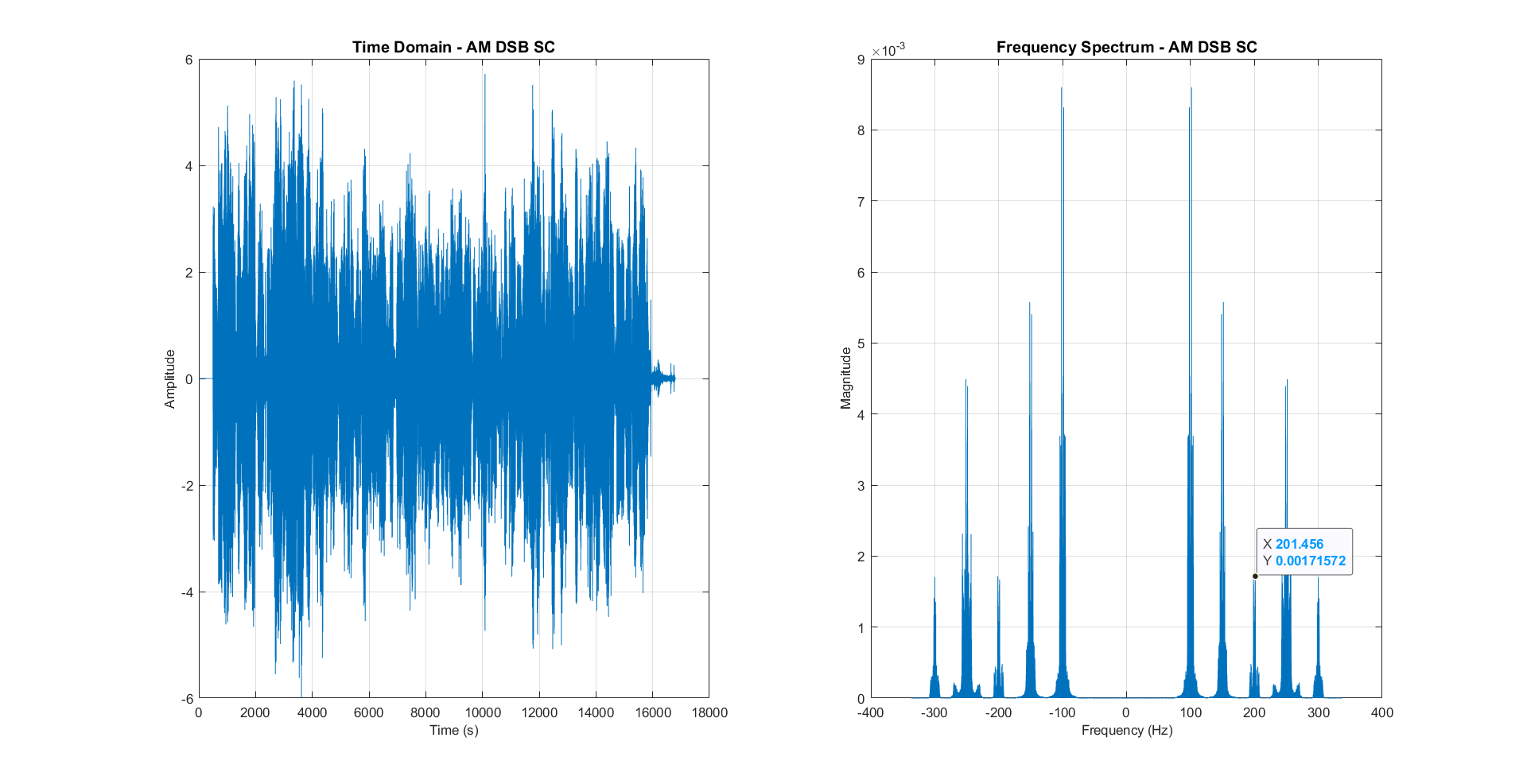


Figure frequency division multiplexing

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APPENDIX:

Our codes are here:

<https://drive.google.com/drive/u/0/folders/1Y5geGvLuV8uRkiI-eebuQKIvkGOPMNaM>

Or scan QR Code:

