Report 5 :  88%
Prelab 5 : 100%
Prelab 6 : 100%

Constructor University Bremen

Natural Science Laboratory

Signals And Systems

Fall Semester 2024

# Lab Experiment 5- AM Modulation

Author: Wanzia Nambule

Instructor: Pagel Uwe

Experiment conducted by: Wanzia Nambule, Dahyun Ko
Place of execution: Teaching Lab EE
Bench 12

Date of execution: 23 October, 2024

## INTRODUCTION

### Objective

The objective of this experiment was to study and analyze different analog modulation techniques, specifically amplitude modulation (AM) and frequency modulation (FM), and to observe their effects in both the time and frequency domains. We investigated the properties of double-sideband (DSB), double-sideband suppressed carrier (DSB-SC), and single-sideband (SSB) amplitude modulation, examining their frequency spectra and exploring various demodulation techniques. Through practical applications, we used an oscilloscope to demonstrate the influence of amplitude modulation parameters, and we built a complete AM system using a function generator as the modulator and an envelope detector as the demodulator. In the second part of the experiment, we analyzed the effects of frequency modulation parameters on bandwidth, utilizing the oscilloscope as a spectrum analyzer and constructing a simple slope detector for FM demodulation.

## THEORY

Modulation is the process of embedding an information-carrying signal within a secondary carrier signal, while demodulation is the retrieval of this embedded information. In amplitude modulation (AM), the transmitted signal modifies the amplitude of the carrier signal. The most common form, sinusoidal amplitude modulation, involves the information signal altering the amplitude of a sinusoidal carrier wave. Modulation serves several purposes, such as reducing signal attenuation during transmission, determining the required antenna size for effective propagation, and enabling simultaneous transmission of multiple distinct signals.

## PRELAB

### Problem 1: Single frequency Amplitude Modulation

1. Deriving the index modulation expression according to the manual description:

   $modulation\ index\ = m$

   $carrier\ amplitude\ = A_c$

   $signal\ frequency\ = f_m$

   $carrier\ frequency\ = f_c$

   $carrier\ after\ modulation\ = A_c cos(2\pi f_c t)$

   $carrier\ after\ modulation\ = A_c[1\ +\ mcos(2\pi f_m t)\ \times\ cos(2\pi f_c t)$

   $A_{max} = A_c[1\ +\ m]$

$A_{min} = A_c[1 - m]$

$m = \frac{A_m}{A_c}$

$A_{max} + A_{min} = A_c[1 + m] + A_c[1 - m] = 2A_c$

$A_{max} - A_{min} = A_c[1 + m] - A_c[1 - m] = 2mA_c$

$m = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}$

2.Deriving the ratio of the total sideband power to the total power expression according to the manual description:

$s(t) = A_c[1 + \mu cos(2\pi f_m t)] \times cos(2\pi f_c t)$

$s(t) = A_c cos(2\pi f_c t) + A_c \frac{\mu}{2} cos2\pi(f_c + f_m)t + A_c \frac{\mu}{2} cos2\pi(f_c - f_m)t$

$resistor\ power = p = \frac{V^2 rms}{R}$

$RMS\ Voltage\ carrier = \frac{A_c}{\sqrt{2}}$

$voltage\ of\ upper\ side\ band = \frac{A_c \mu}{2\sqrt{2}}$

$voltage\ of\ lower\ side\ band = \frac{A_c \mu}{2\sqrt{2}}$

$P_s = P_{USB} + P_{LSB}$

$P_c = \frac{A_c^2}{2R}$

$P_{USB} = P_{LSB} = \frac{m^2 P_c}{4}$

$P_s = P_{USB} + P_{LSB} = 2 \times \frac{m^2 P_c}{4} = \frac{m^2 P_c}{2}$

$P_T = P_C + P_S = P_C + \frac{m^2 P_c}{2}$

$r_p = \frac{P_S}{P_T} = \frac{\frac{m^2 P_c}{2}}{P_C + \frac{m^2 P_c}{2}} = \frac{m^2}{2+m^2}$

3. Calculating the ratio $r_p$ assuming a modulation index of 100%:

$$r_p = \frac{m^2}{2+m^2} = \frac{1}{3}$$

4. Calculating the ratio $r_p$ of the carrier $V_C(t) = 5cos(20000\pi t)$, modulated by signal

$V_m(t) = 2 + cos(2000\pi t)$

$$s(t) = [A_c + V_{off}]cos(2000\pi t) + \frac{A_{cm}}{2}cos(22000\pi t) + \frac{A_{cm}}{2}cos(18000\pi t)$$

$$P_S = \frac{V_s^2}{R} = \frac{m^2A_c^2}{2R}$$

$$P_c = \frac{V^2}{R} = \frac{(A_C+V_{off})^2}{R}$$

$$P_T = \frac{m^2A_c^2}{2R} + \frac{(A_C+V_{off})^2}{R}$$

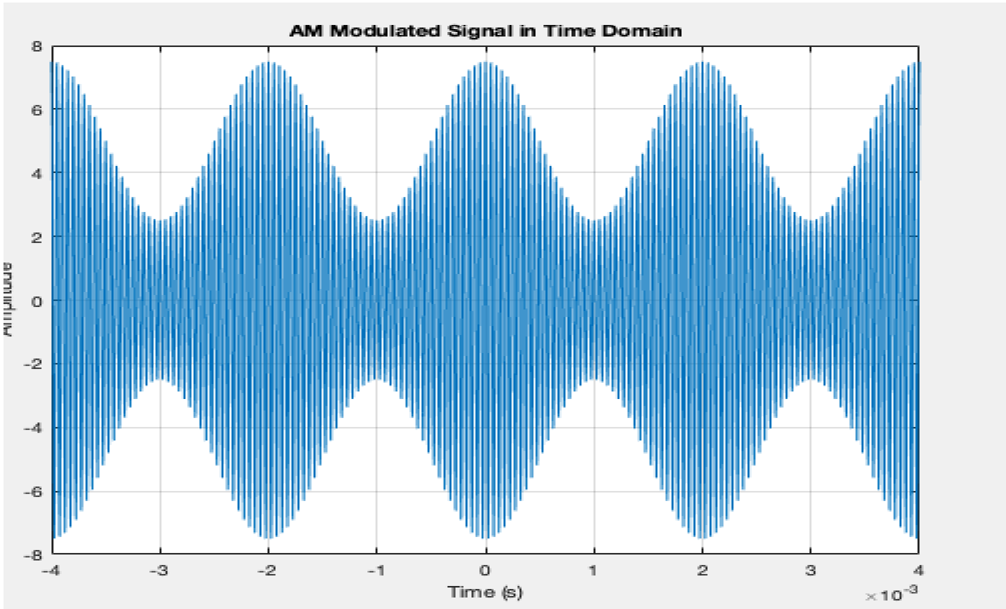
$$r_p = \frac{P_S}{P_T} = \frac{\frac{m^2A_c^2}{2R}}{\frac{m^2A_c^2}{2R} + \frac{(A_C+V_{off})^2}{R}} = \frac{\frac{(\frac{1}{5})^2 5^2}{2R}}{\frac{(\frac{1}{5})^2 5^2}{2R} + \frac{(5+2)^2}{R}} = \frac{1}{99}$$

We adjust the input to maximize the sideband-to-total power ratio by eliminating or reducing the constant component (DC offset) from the message signal.

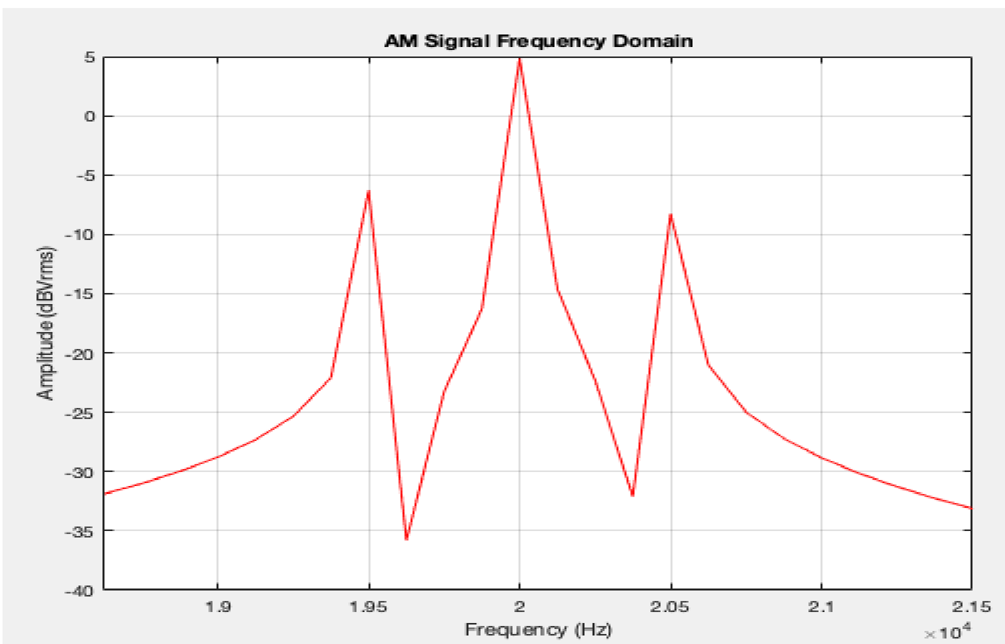
**Problem 2: Amplitude Demodulation**

1. Simulating the demodulation of an AM signal using a MATLAB script. The carrier signal is a sinusoid with a frequency of 20 kHz and an amplitude of 5 V, while the modulation signal is a sinusoid with a frequency of 500 Hz and a modulation index of 50%."

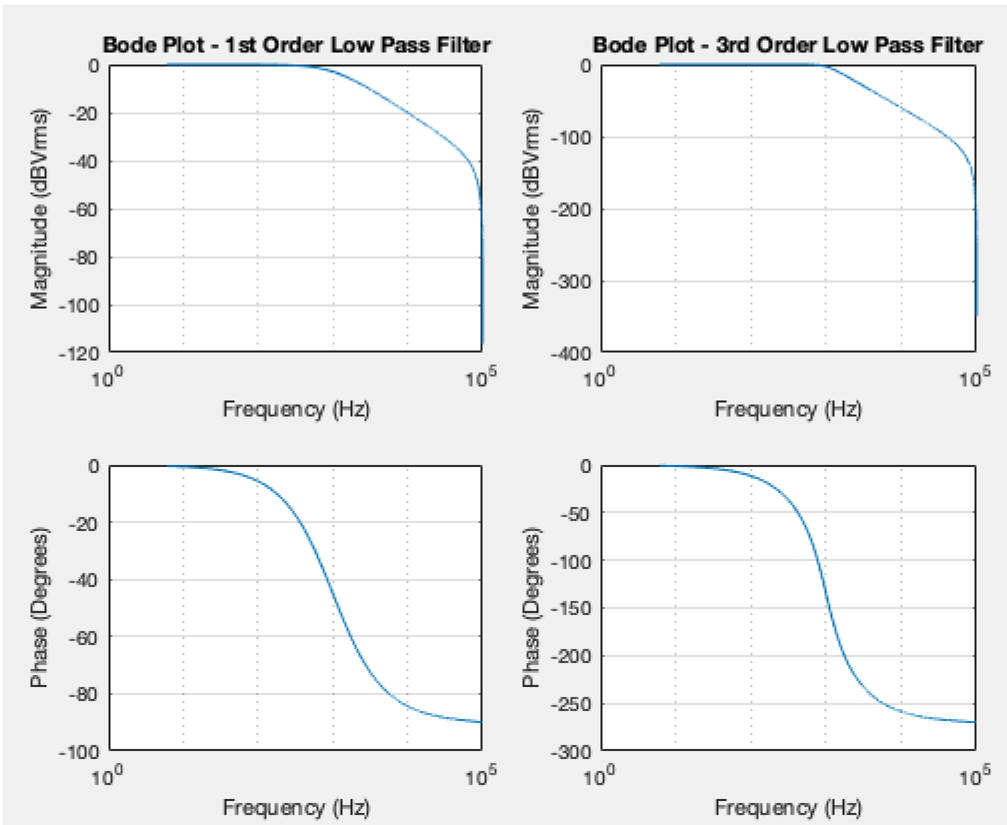
Plot showing modulated signal in time domain:



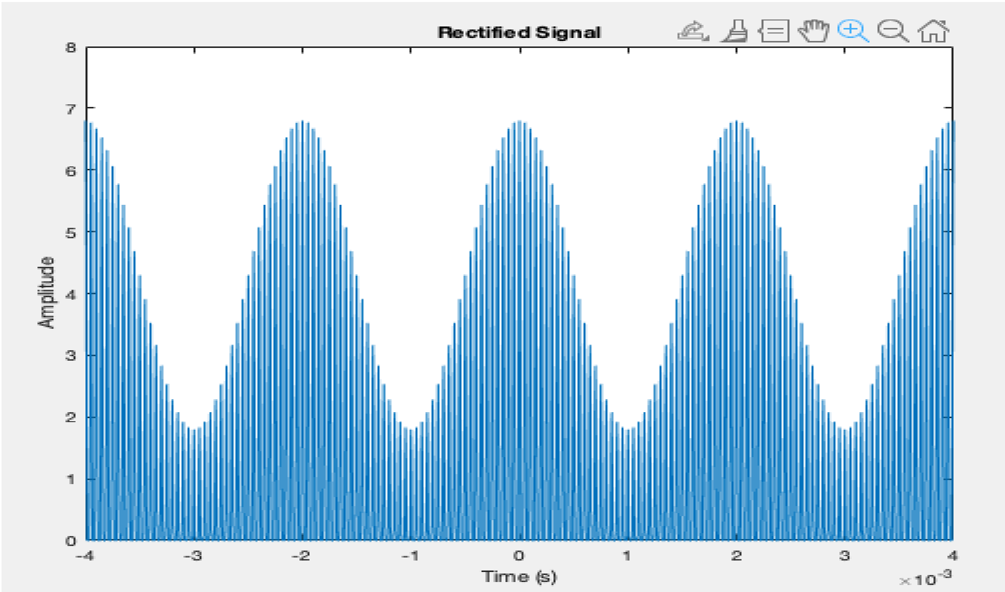

Plot showing modulated signal in frequency domain:

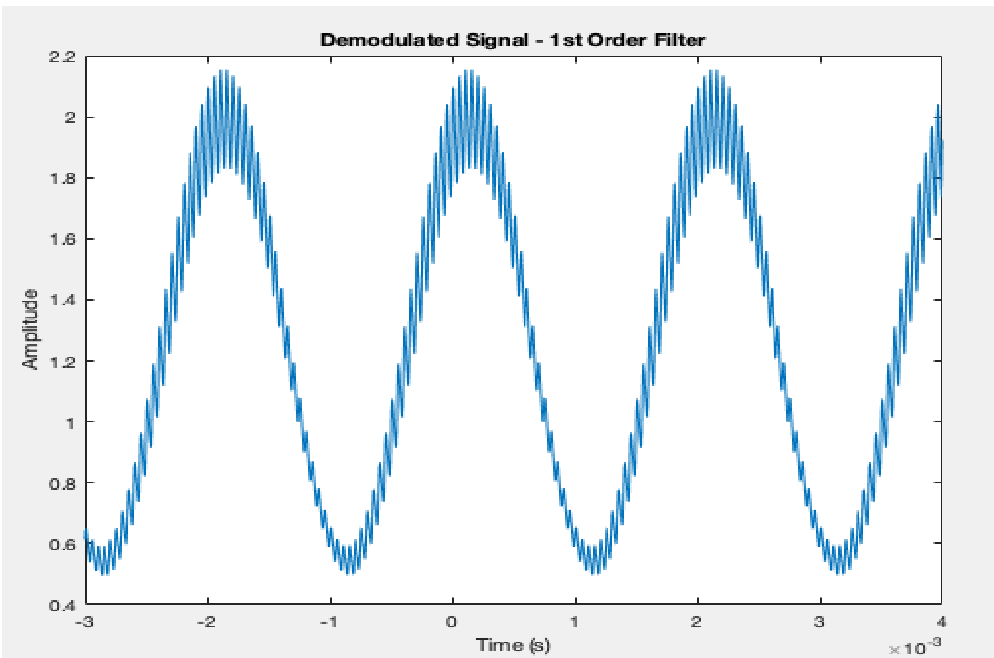


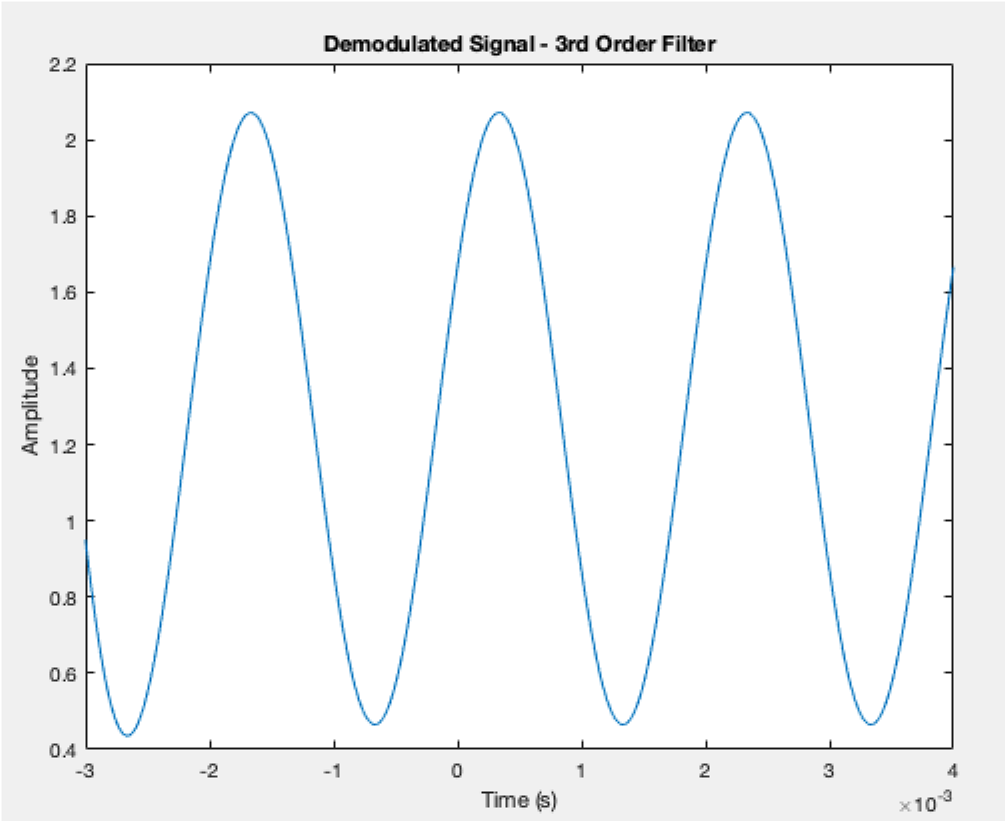
We designed a first-order and a third-order Butterworth low-pass filter, each with a cutoff frequency of 1 kHz, to aid in demodulating the AM signal. To confirm their functionality, we generated Bode plots over a frequency range of 100 Hz to 100 kHz.




The Bode diagrams in MATLAB show that both the first-order and third-order filters maintain a similar magnitude response up to the cutoff frequency of 1 kHz, with their phase responses also closely aligned. Beyond the cutoff, the third-order filter exhibits a sharper roll-off compared to the first-order filter, confirming the expected filter behaviors for clearer signal demodulation.


Rectified Signal
Amplitude
Time (s)
8
7
6
5
4
3
2
1
0
-4
-3
-2
-1
0
1
2
3
4
$\times 10^{-3}$


Demodulated Signal - 1st Order Filter
Amplitude
Time (s)
2.2
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
-3
-2
-1
0
1
2
3
4
$\times 10^{-3}$




The two Bode plots show that both filters are in phase, but the first-order low-pass filter exhibits more ripples and fluctuations compared to the third-order filter. This indicates that the third-order filter is the better choice for eliminating unwanted frequency components and achieving a cleaner demodulated signal.

Below is the code that was used to to create the plots above:

```
>> % Sampling and Signal Parameters
fs = 200000;            % Sampling rate
fn = fs / 2;            % Nyquist rate
t = -0.004: 1/fs: 0.004; % Time vector

% Modulate signal
m = 5 .* (1 + 0.5 * cos(2 * pi * 500 * t)) .* cos(2 * pi * 20000 * t);

% Plot Modulated Signal in Time Domain
figure(1);
plot(t, m);
grid on;
xlabel('Time (s)');
ylabel('m(t)');
title('AM Modulated Signal in Time Domain');

% Fourier Transform of the Modulated Signal
n = length(m);
M = abs(fft(m) / n); % Normalize FFT output
M = M(1:((n+1)/2)); % Take half the FFT for single-sided spectrum
d = 20 * log10(M / sqrt(2)); % Convert to dBVrms
f = linspace(0, fn, length(M)); % Frequency vector

% Plot Frequency Spectrum
figure(2);
plot(f, d, 'r', 'LineWidth', 1);
xlim([18625 21500]);
xlabel('Frequency (Hz)');
ylabel('Amplitude (dBVrms)');
title('Frequency Spectrum of AM Modulated Signal');
grid on;

% Design Butterworth Filters (1st and 3rd Order)
fc = 1000; % Cutoff frequency
[b1, a1] = butter(1, fc / fn, 'low');  % First-order Butterworth filter
[b3, a3] = butter(3, fc / fn, 'low');  % Third-order Butterworth filter

% Frequency Response for the First-Order Filter
[H1, f1] = freqz(b1, a1, 2^14, fs);
```

```matlab
% Frequency Response for the Third-Order Filter
[H3, f3] = freqz(b3, a3, 2^14, fs);

% Plot Bode Diagram for First and Third Order Filters
figure(3);

% Bode Plot - First Order Filter
subplot(2,2,1);
semilogx(f1, 20 * log10(abs(H1)));
grid on;
xlabel('Frequency (Hz)');
ylabel('Magnitude (dBVrms)');
title('Bode Plot - 1st Order Low Pass Filter');

subplot(2,2,3);
semilogx(f1, unwrap(angle(H1)) * (180/pi));
grid on;
xlabel('Frequency (Hz)');
ylabel('Phase (Degrees)');

% Bode Plot - Third Order Filter
subplot(2,2,2);
semilogx(f3, 20 * log10(abs(H3)));
grid on;
xlabel('Frequency (Hz)');
ylabel('Magnitude (dBVrms)');
title('Bode Plot - 3rd Order Low Pass Filter');

subplot(2,2,4);
semilogx(f3, unwrap(angle(H3)) * (180/pi));
grid on;
xlabel('Frequency (Hz)');
ylabel('Phase (Degrees)');

% Rectify the Signal
rm = m - 0.7;             % Simulate diode drop (0.7V)
r = (rm + abs(rm)) / 2;   % Full-wave rectification
figure(4);
plot(t, r);
ylim([0 8]);
xlabel('Time (s)');
```

```matlab
ylabel('Voltage (V)');
title('Rectified Signal');

% Filter the Rectified Signal with First-Order Filter
demod1 = filter(b1, a1, r);
figure(5);
plot(t, demod1);
xlim([-0.003 0.004]);
xlabel('Time (s)');
ylabel('Voltage (V)');
title('Demodulated Signal using 1st Order Filter');

% Filter the Rectified Signal with Third-Order Filter
demod3 = filter(b3, a3, r);
figure(6);
plot(t, demod3);
xlim([-0.003 0.004]);
xlabel('Time (s)');
ylabel('Voltage (V)');
title('Demodulated Signal using 3rd Order Filter');
```

# EXECUTION AND RESULTS

## Experimental Setup

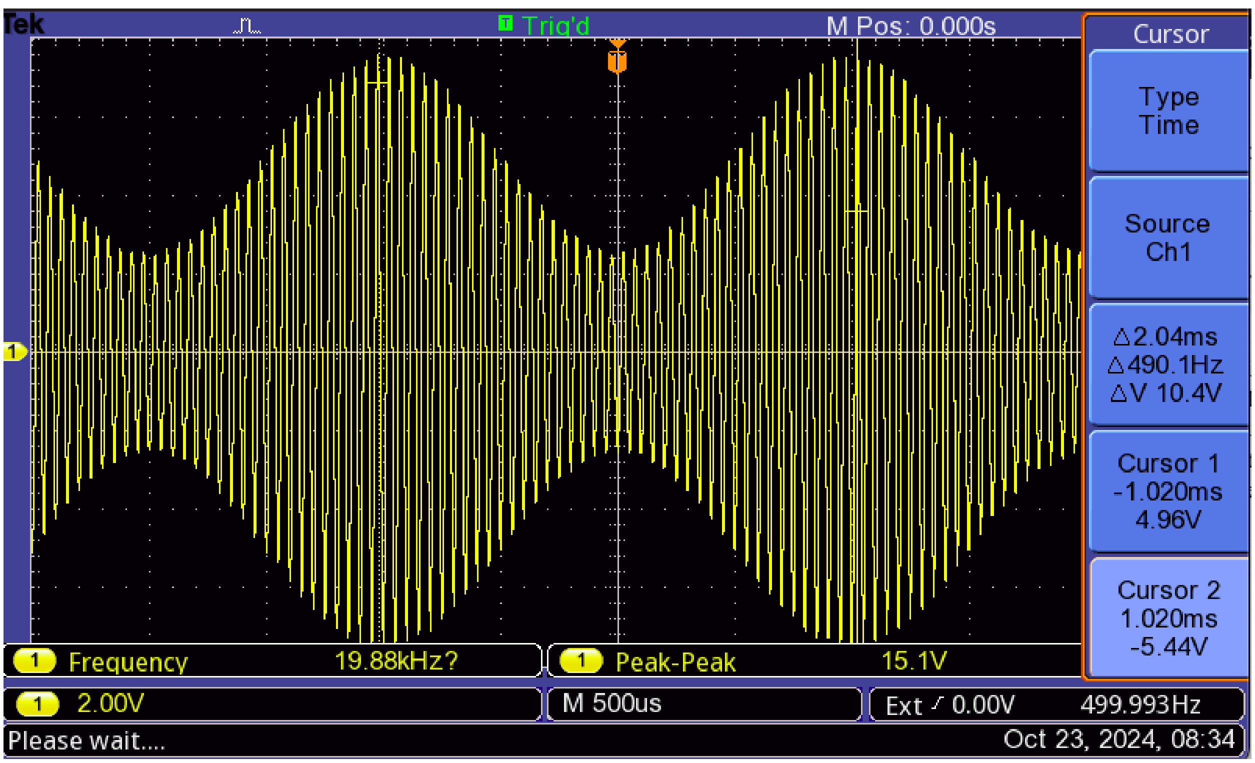
The following tools and instruments used include • BNB Cable • BNB T-connector • BNC-Banana connector • Lab wires • Generator • Oscilloscope • Breadboard • Auxiliary function generator • 2 DC source 10V • Resistors • 1N4148 diode • Capacitors 6n8F, 1u0F,2 of 100nF • Inductor 10mH • Resistors 180R, 22k0 .

## Results

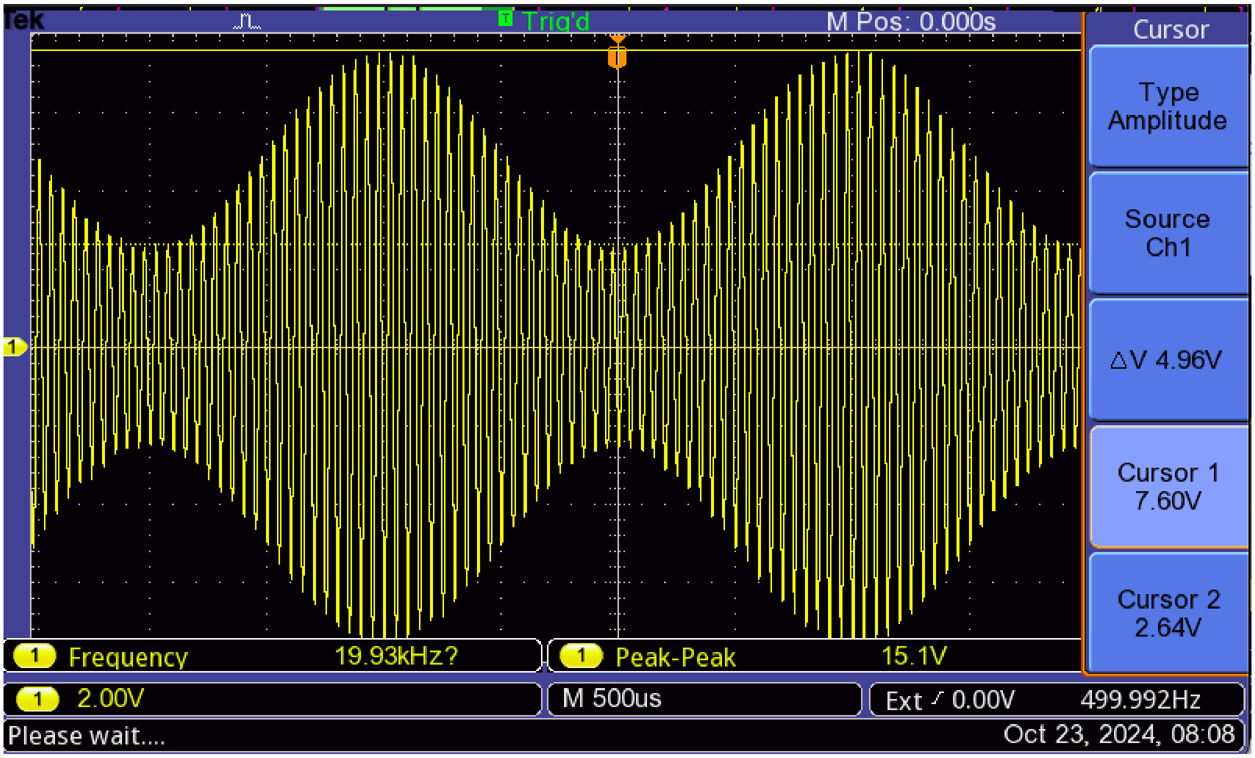
### Problem 1: AM modulated Signals in Time Domain

The AM signals were generated following the instructions in the handbook, which involved selecting a sine waveform and applying AM modulation. The configuration included setting the carrier frequency to 20 kHz, an amplitude of 10 Vpp, a modulation frequency of 500 Hz, and a modulation index of 50%. The function generator was then connected to the oscilloscope, where the signal was observed, and various parameters such as maximum amplitude, minimum amplitude, and frequency were measured.

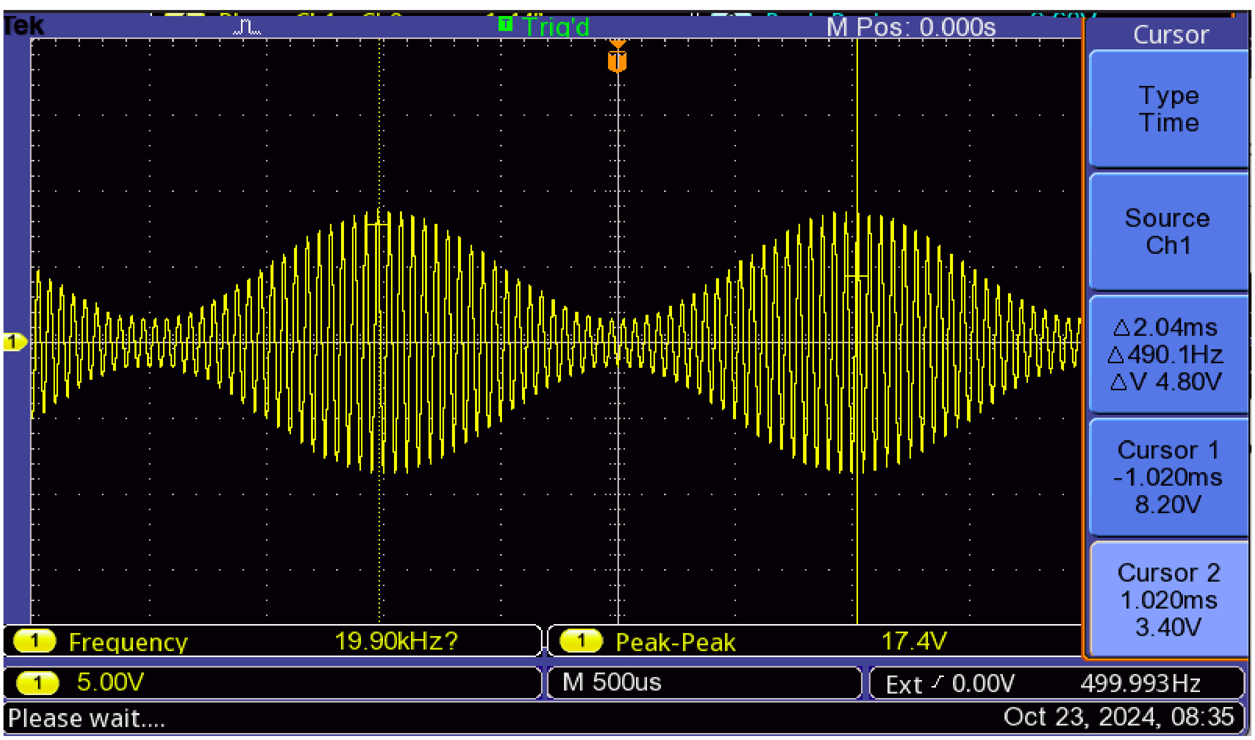
Below is a hardcopy showing frequency properties at 50% modulation index



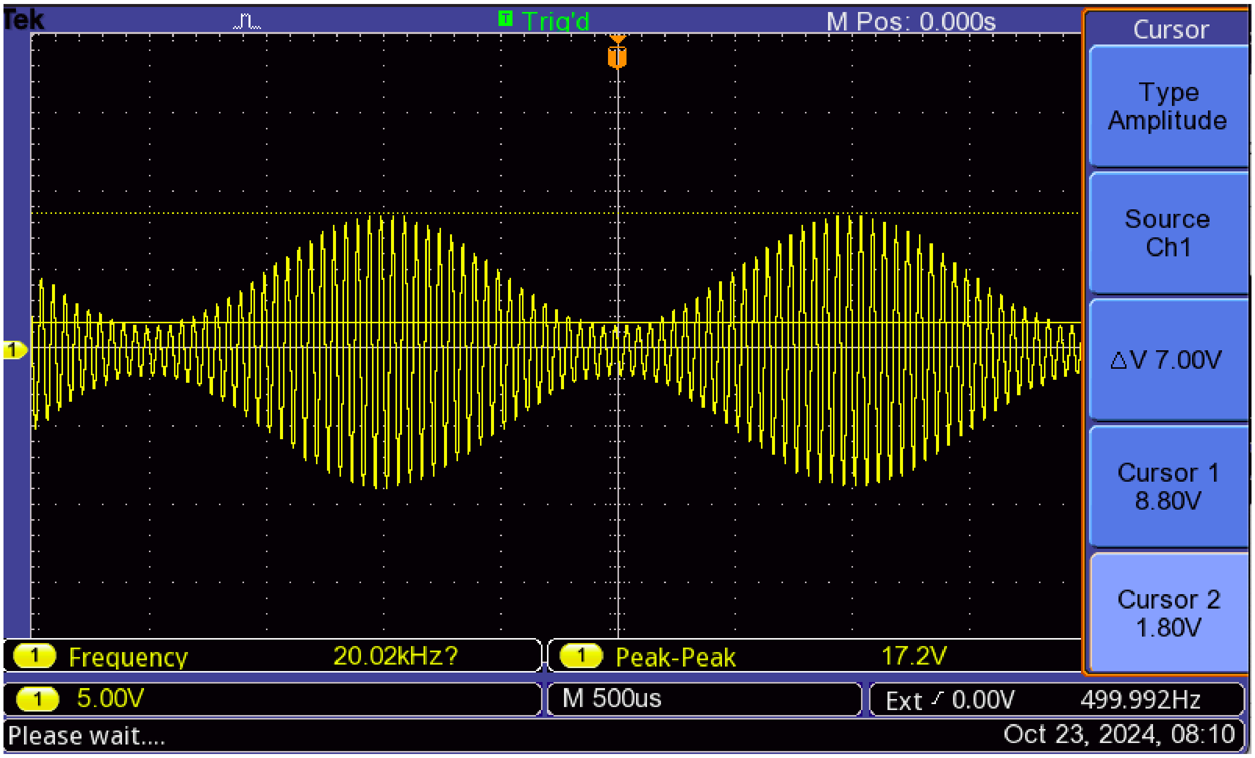
Below is a hardcopy showing amplitude properties at 50% modulation index




Below is a hardcopy showing frequency properties at 70% modulation index

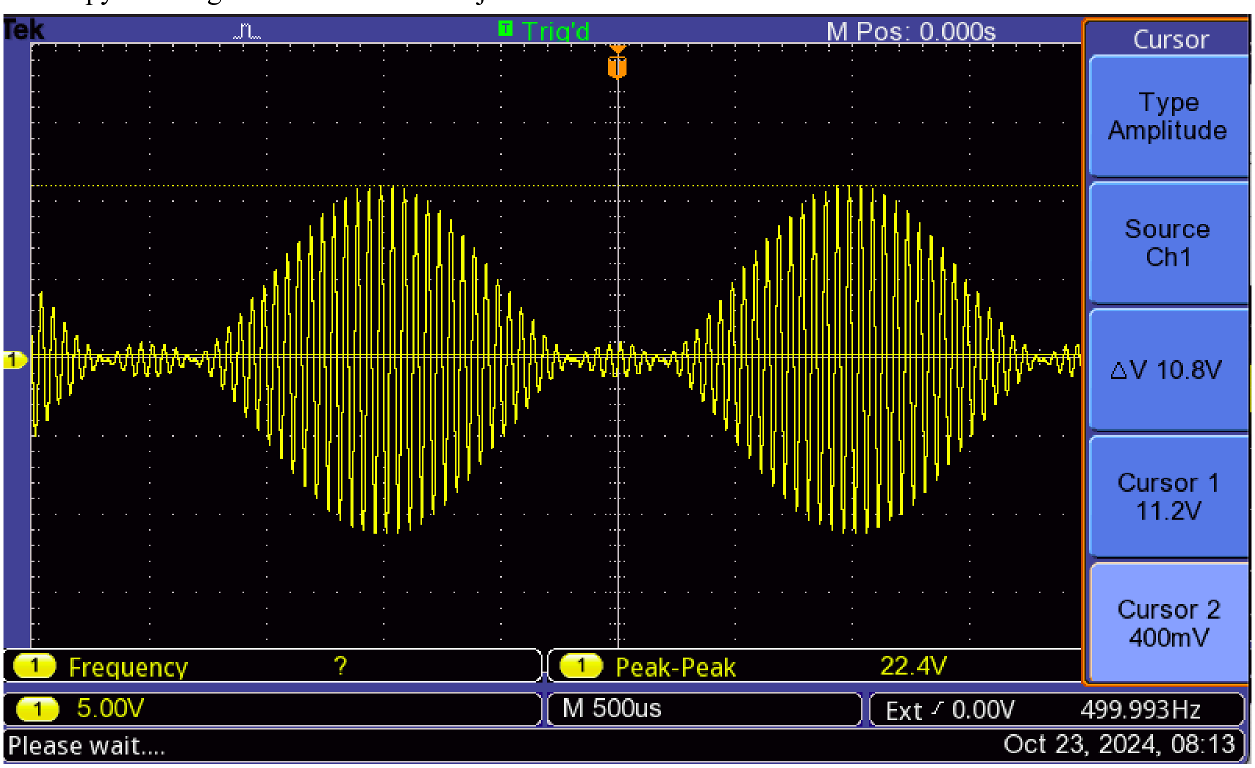


Below is a hardcopy showing Amplitude properties at 70% modulation index




calculated m values?

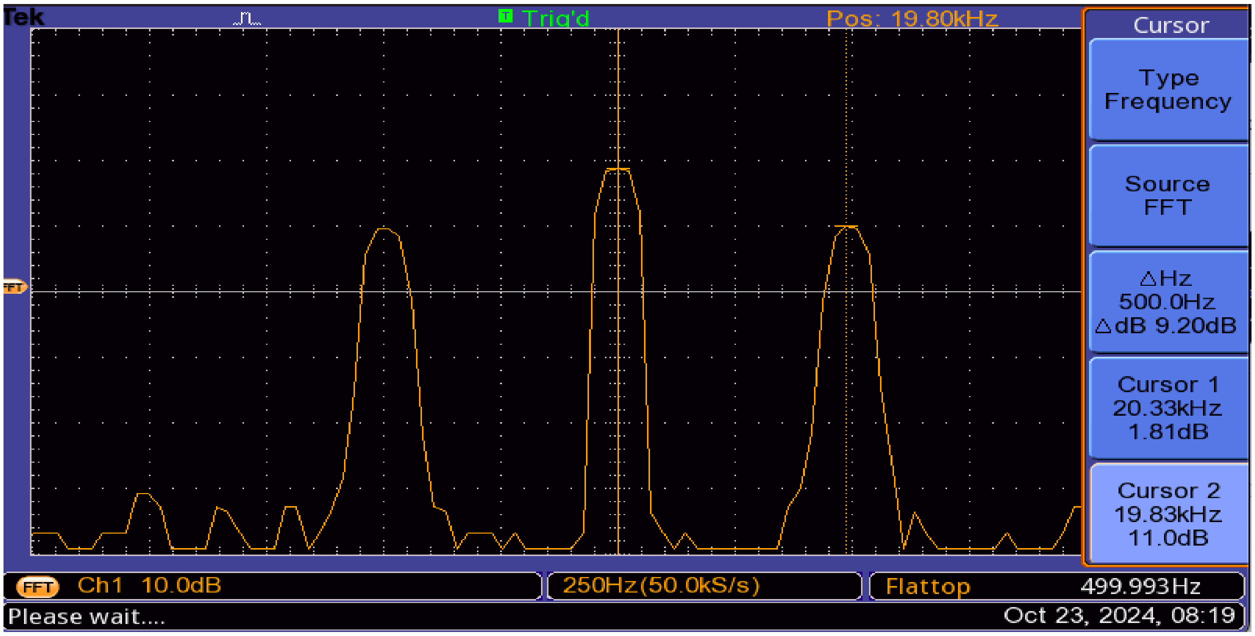
Hardcopy showing modulation index Adjusted to be 120%



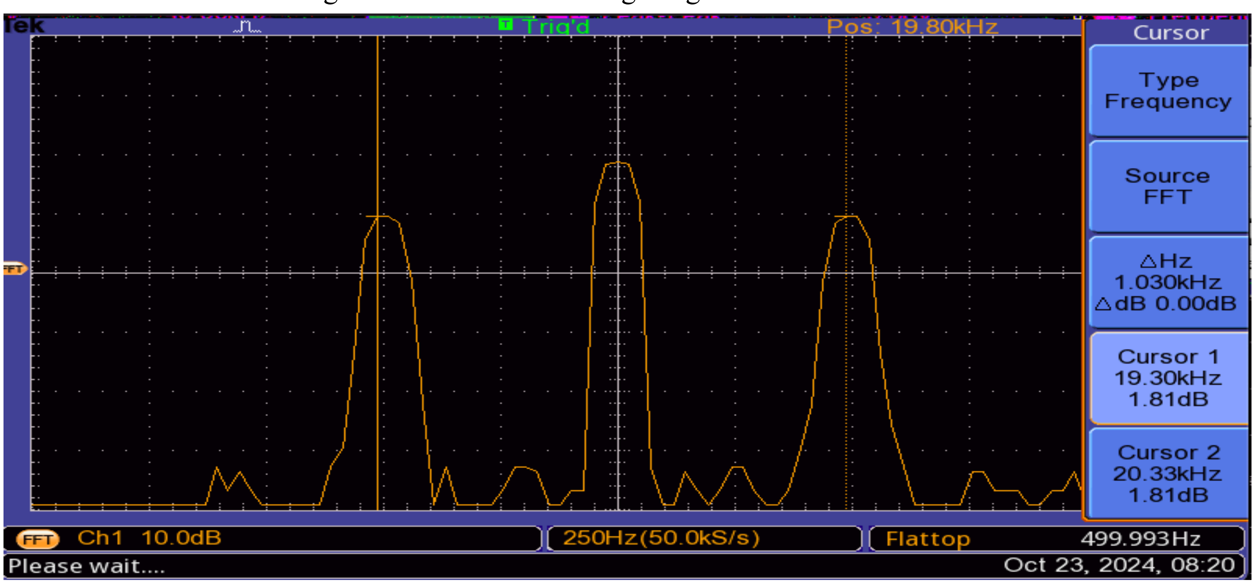
**Problem 2: AM Modulated Signals in Frequency Domain**

Using the same setup as before and setting the modulation index at the function generator to 70%. The oscilloscope displayed the amplitude modulated signal in the frequency domain. (FFT!). Using the cursors we measured  the magnitudes and the frequencies.

Hardcopy showing the magnitude and frequency of the middle signal(Cursor 2):




Hardcopy showing the magnitude and frequency of the two side signals:
Where cursor 1 is the left signal and cursor 2 is the right signal.



**Problem 3: Demodulation of a message signal**

We built the circuit below on the breadboard

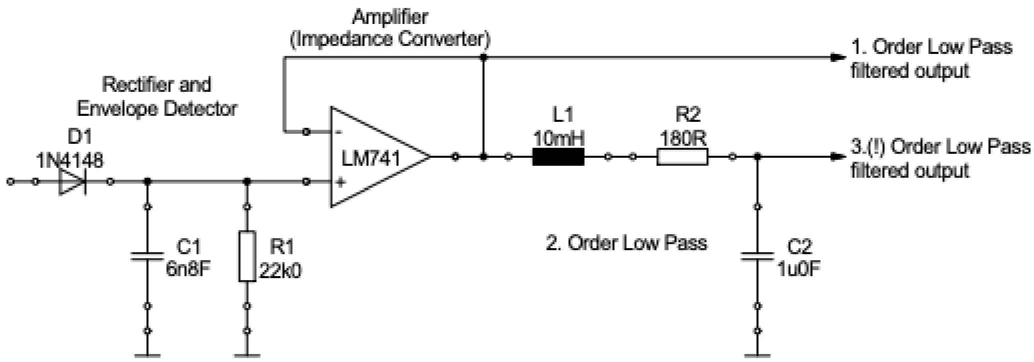



Fig. 1: Demodulation circuit

Where the detailed Op-Amp looked like this

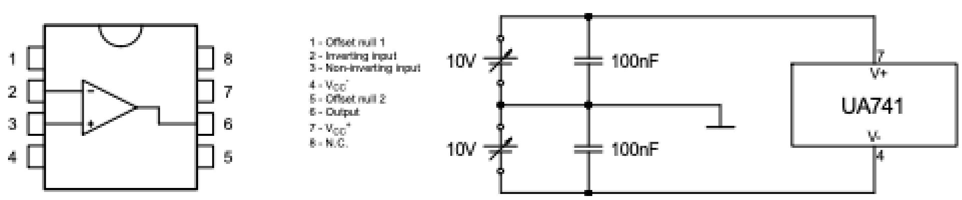
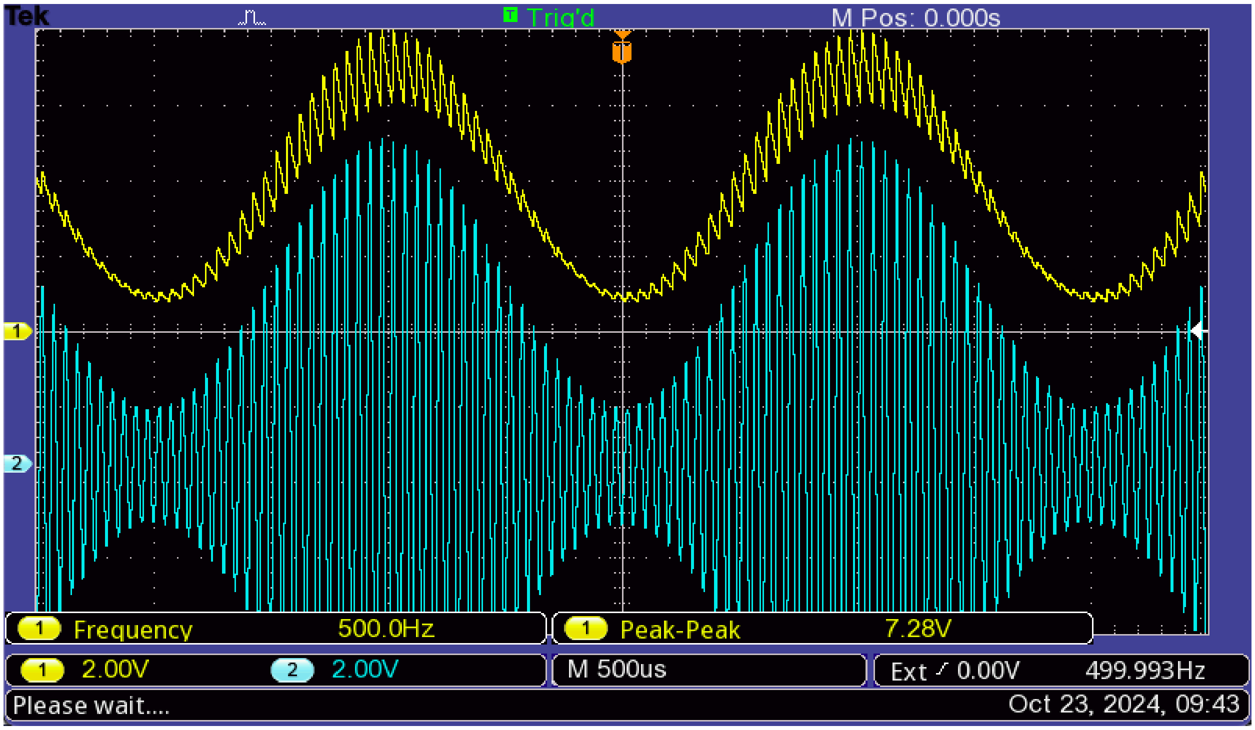


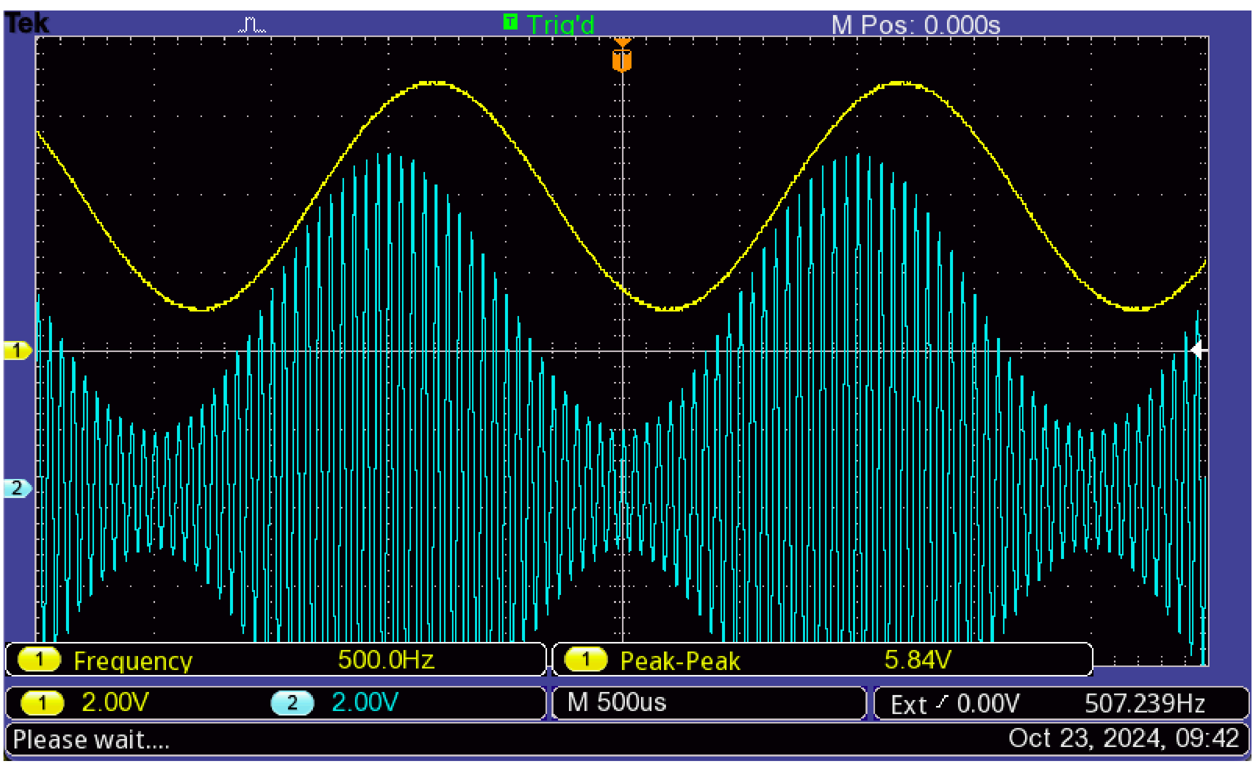

Fig. 2: LM741 pinout and supply circuit

The AM signals were generated following the instructions in the manual, which involved selecting a sine waveform and applying AM modulation. The configuration included setting the carrier frequency to 20 kHz, an amplitude of 10 Vpp, a modulation frequency of 500 Hz, and a modulation index of 50%. The function generator was connected to the input of the demodulating circuit, where the signals were observed.

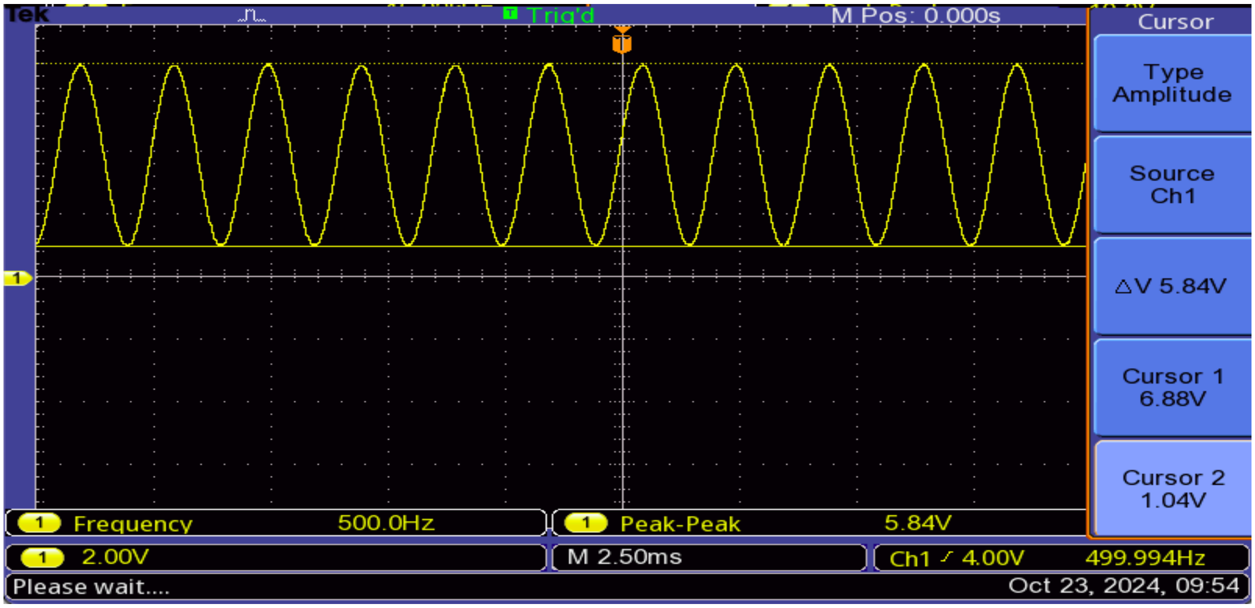
Below is a hardcopy displaying AM modulated signal together with the 1. order filter output:



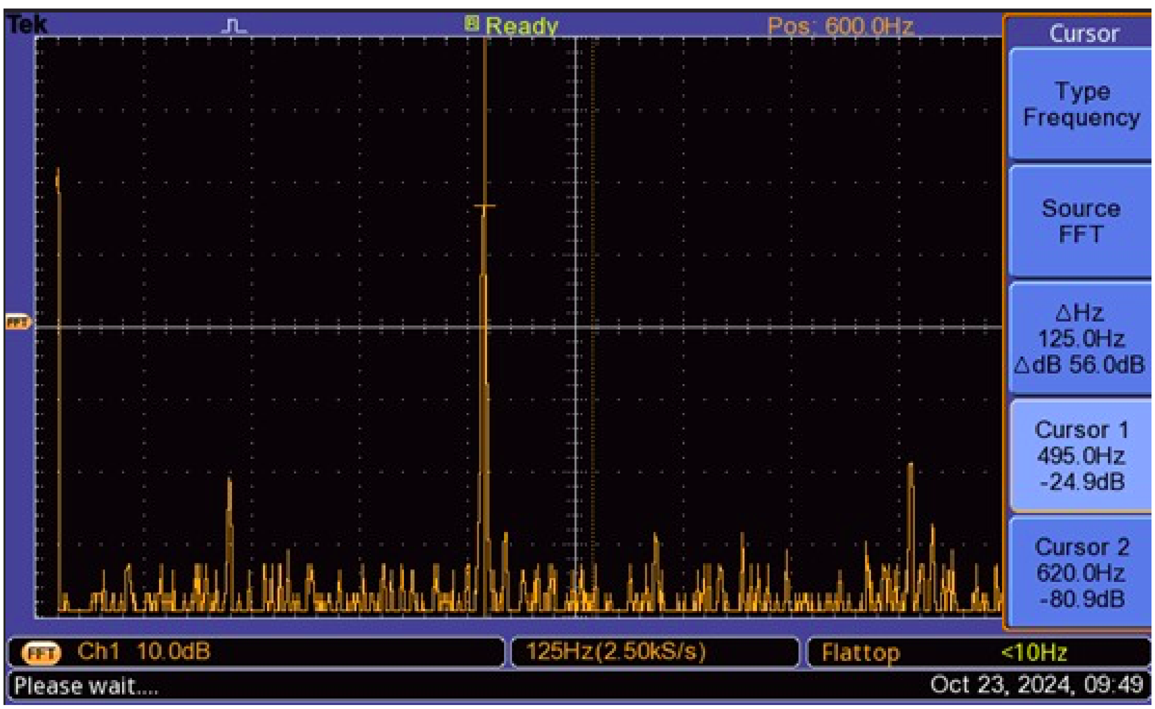

Below is a hardcopy displaying AM modulated signal together with the 3. order filter output:

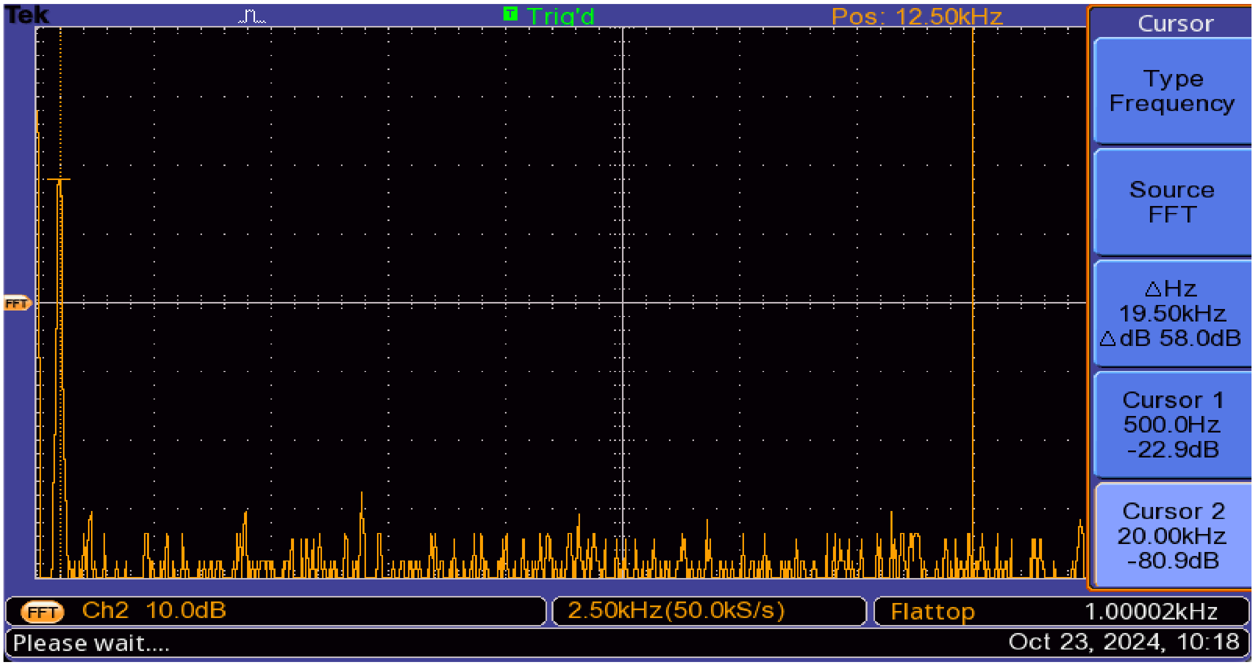


Measuring the amplitude of the demodulated signal at the 3. order output, we get:




Hardcopy showing FFT of the signal at the 3. order filter output.






From the hardcopy above, we can see that after passing the filter the carrier disappears, meaning the signal has been demodulated!

## Evaluation AM modulation

### Problem 1: AM modulated Signals in Time Domain

The modulation index $m$ affects the relative magnitudes of the frequency components in AM. As $m$ increases, the sideband amplitudes grow proportionally, while the carrier amplitude remains constant. At $m = 1$ sidebands are maximized without distortion. Beyond $m = 1$, over-modulation occurs, introducing distortion by adding extra frequency components.

Calculating the modulation index using the measurements in the hardcopies taken:

Modulation index at 50%:

$$m = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}$$

$$m = \frac{7.60 - 2.64}{7.60 + 2.64}$$

$$m = 0.484\%$$

Modulation index at 70%:

$$m = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}$$

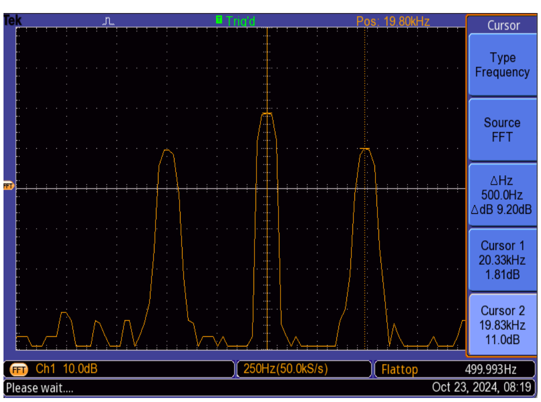
$$m = \frac{8.80 - 1.80}{8.80 + 1.80}$$

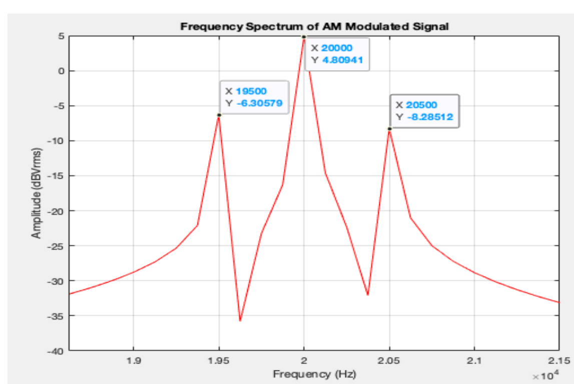
$$m = 0.6603\%$$

The calculated modulation index closely matches the set value on the generator, with minor differences likely due to instrument-related errors, such as the oscilloscope's limited resolution which is too low.

When the modulation index exceeds 100%, the amplitudes of the upper and lower envelope signals overlap, causing the envelope to distort—an effect known as overmodulation. As shown in the results, setting the modulation index to 120% led to this overlap in signal amplitudes.

### Problem 2: AM Modulated Signals in Frequency Domain



The figures shown have similar structures, each displaying an upper band, a carrier, and two sidebands. Theoretically, the magnitudes should be higher than those observed on the oscilloscope, likely due to the oscilloscope's accuracy limitations.

The generator outputs a Double-Sideband (DSB) signal because it retains the carrier, which appears as a carrier component in the Fourier domain of the modulated signal.

Adjusting the carrier frequency shifts the signal spectrum without affecting the envelope. However, if the carrier frequency is lower than the message frequency, frequency components overlap, distorting the modulation signal and complicating reconstruction.

Adjusting the message frequency shifts the spectrum, moving the upper and lower sidebands right and left, respectively, in proportion to the message frequency. When the message frequency surpasses the carrier frequency, spectrum overlap occurs, resulting in wider sidebands due to the higher modulating frequency.

carrier frequency should be always higher than the signal f

Determining the modulation index m using the measured values:

We first convert dB to linear amplitude

Since the side bands have the same amplitude, we only have to convert once

$$A = 10^{\frac{dB}{20}}$$

$$A = 10^{\frac{1.81}{20}}$$

$$A = 1.232$$

And for the carrier amplitude we get:

$$A = 10^{\frac{dB}{20}}$$

$$A = 10^{\frac{11.0}{20}}$$

$$A = 3.548$$

Now that we have done the conversions we can go ahead and calculate m

$$m = \frac{A_{USB}+A_{LSB}}{A_C}$$

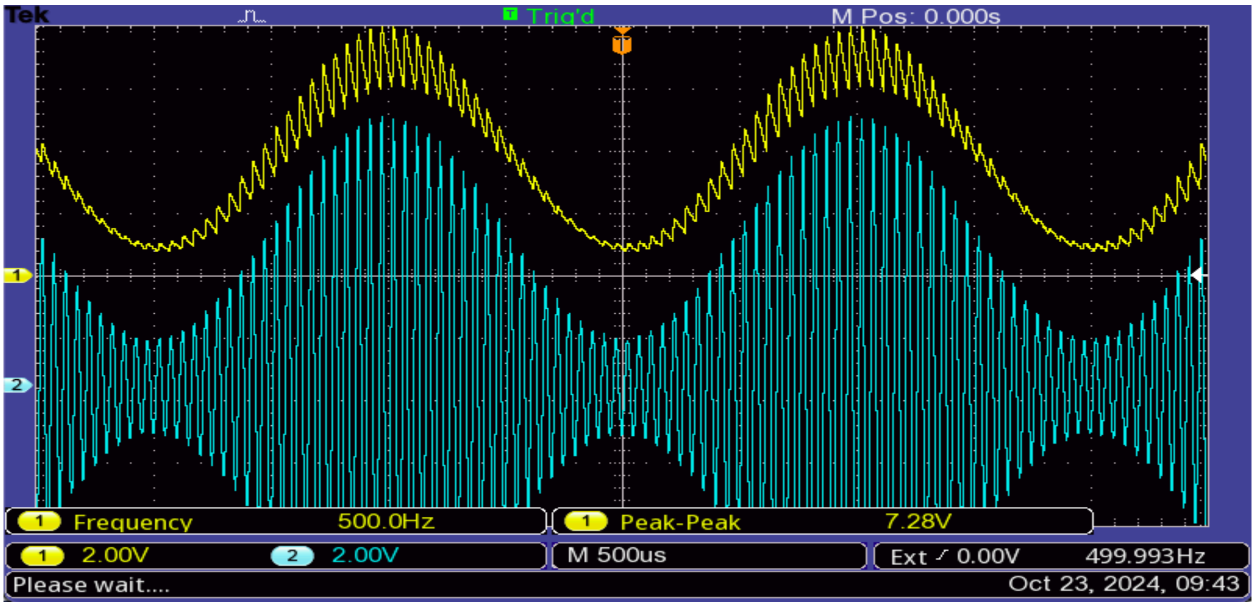
$$m = \frac{1.232+1.232}{3.548}$$

$$m = 0.694\%$$

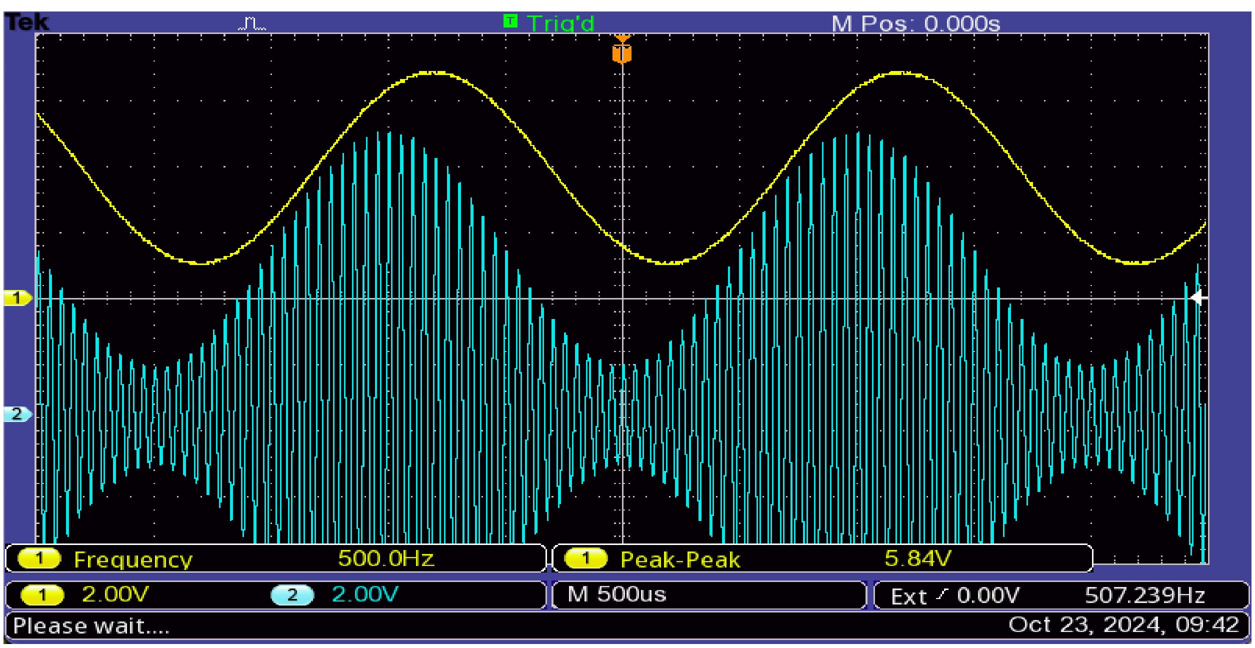
The calculated modulation index closely matches the set value on the generator, with minor differences likely due to instrument-related errors, such as the oscilloscope's limited resolution which is too low.

**Problem 3: Demodulation of a message signal**

1. order filter output:



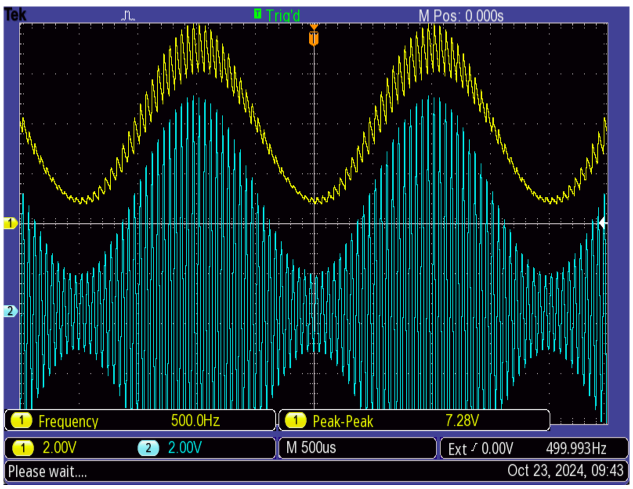

3. order filter output:

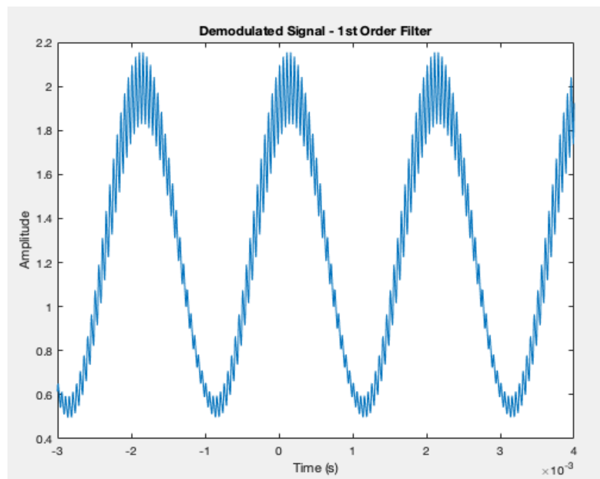


The images above indicate that the signals are synchronized or in phase. However, they reveal significantly more fluctuations or ripples in the first-order signal, which contains frequency components at 20,000 Hz, compared to the third-order filter output signal, which has only one frequency component. Therefore, to achieve greater accuracy in eliminating unwanted frequencies, a third-order filter is preferable to a first-order filter, as it results in fewer fluctuations.

**Comparing the measured signals with the MatLab results:**

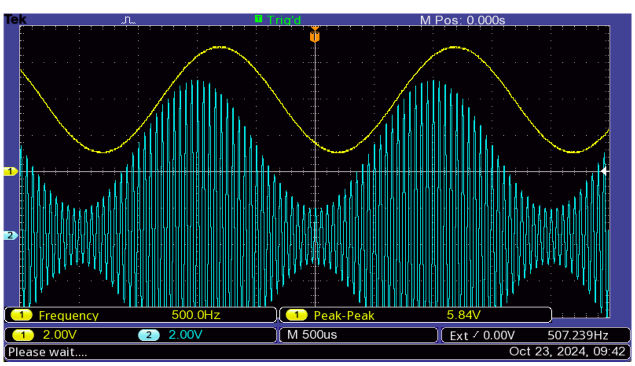
First order signals:

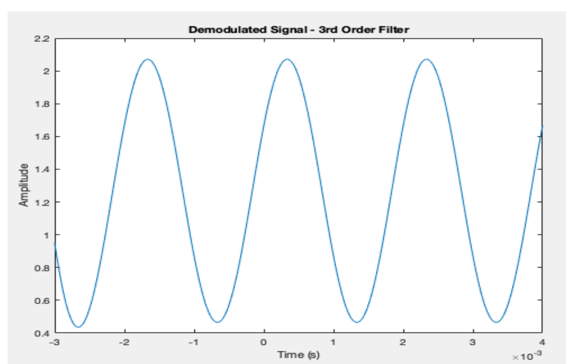



The first-order signals are similar, and both exhibit distortion due to noise, as anticipated. They also display ripples or fluctuations in frequency.

Third order signals:




There are some differences in the peak-to-peak amplitudes of the signals from the oscilloscope and MATLAB. This is due to the accuracy limitations of the oscilloscope. The phase shift seen in the oscilloscope signal is likely due to the real-world characteristics of the filter, whereas the MATLAB simulation, being ideal, does not display a phase shift.

what about remaining carrier components in the third order output?

## CONCLUSION

To enhance transmission efficiency, signals can be modified during transmission using amplitude or frequency modulation. AM comes in several types, including Double Sideband AM, Double Sideband with Suppressed Carrier, and Single Sideband Modulation, each with unique power consumption characteristics. In this experiment, we focused on Double Sideband AM without carrier suppression. Our findings indicate that the modulation index of the AM signal influences power usage. Additionally, excessive modulation, above 100%, can cause message signal distortion, making it essential to keep the modulation index below unity (1). To prevent overmodulation, the carrier frequency should be at least twice the highest frequency component of the message signal.

We conducted demodulation experiments using both MATLAB and laboratory setups. Higher-order low-pass filters, as demonstrated by both experimental and simulation results, are generally preferable as they more effectively filter out unwanted components without introducing ripples or fluctuations.

During the experiment, we observed some inaccuracies, particularly in the placement and amplitudes of frequency components. These discrepancies are primarily due to limitations in the oscilloscope's accuracy, which could be minimized by using a more precise oscilloscope.

## REFERENCE

A. V. Oppenheim, A. S. Willsky, S. H. Nawab, ”Signals and Systems,” Prentice Hall, Second Edition 1997. 2.

Theodore S. Rappaport, Wireless Communications: Principles and Practice (2nd Edition)

Uwe Pagel & Prof. Dr. Ing. Werner Henkel, Jacobs University Bremen CO-520-B Signals and Systems Lab Manual

# APPENDIX

**Prelab FM Modulation**

**Problem 1: Frequency Modulator**

$m(t) = 4cos(8000\pi t)$

$Amplitude\ modulation\ signal = A_m = 4$

$Frequency\ deviation\ constant = K_f = 10KHz$

$Frequency\ deviation = \Delta f = A_m K_f = 40KHz$

$f_m = \frac{\omega_m}{2\pi} = 4000Hz$

$Frequency\ modulation\ index = B_f = \frac{\Delta f}{f_m} = \frac{40KHz}{4000Hz} = 10$

**Problem 2: FM signal in the frequency domain**

Using Carson's rule to calculate the bandwidth:

$B_T \approx 2f_m(B_f + 1)$

$f_m = 5KHz$

When $B_f = 0.2$

$B_T \approx 2 \times 5000(0.2 + 1) = 12KHz$

When $B_f = 1$

$B_T \approx 2 \times 5000(1 + 1) = 20KHz$

When $B_f = 2$

$B_T \approx 2 \times 5000(2 + 1) = 30KHz$

Below is the table showing the results

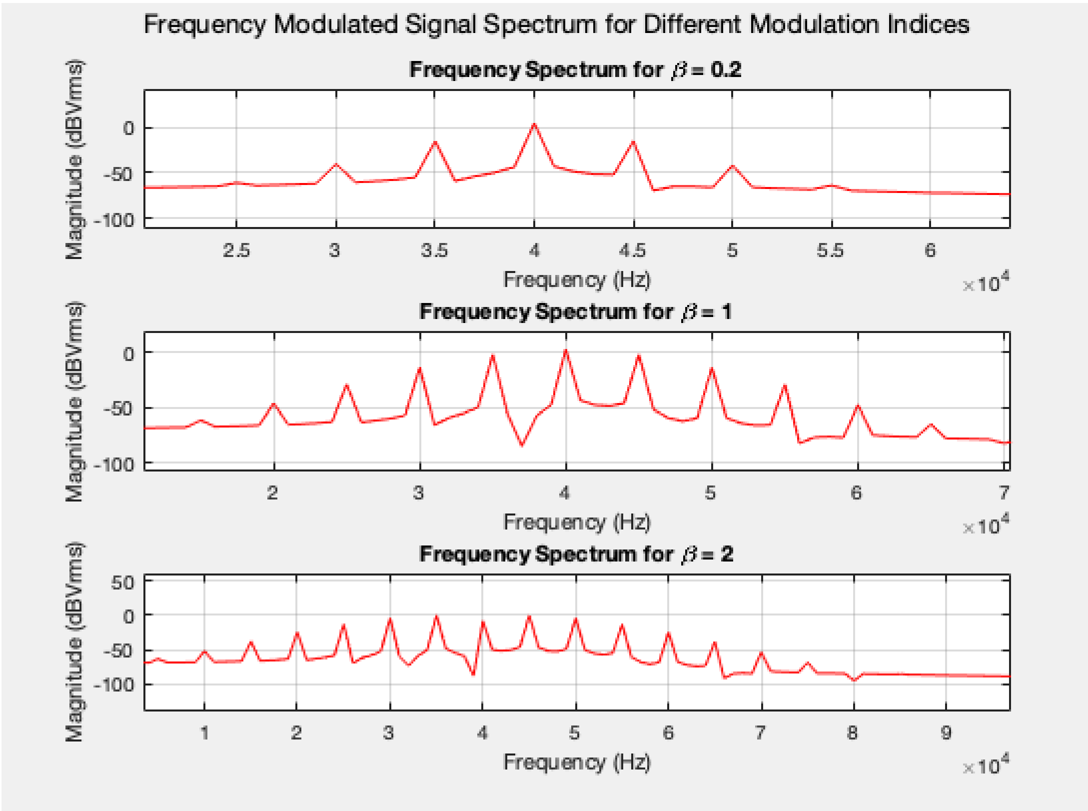
| $B_f$ | $B_T(KHz)$ |
|---|---|
| 0.2 | 12 |
| 1 | 20 |
| 2 | 30 |

Below the matlab code used to generate the diagrams below

```
>> fs = 10000000; % Sampling frequency
fn = fs / 2;    % Nyquist frequency
t = -0.0005:1/fs:0.0005; % Time vector

B_values = [0.2, 1, 2]; % Modulation indices (beta values)
fc = 40000;  % Carrier frequency in Hz
fm = 5000;   % Modulation frequency in Hz
Ac = 2.5;    % Carrier amplitude in V (peak)

% Initialize arrays to store results
bandwidths = zeros(1, length(B_values));
peak_magnitudes = cell(1, length(B_values));

figure;
for i = 1:length(B_values)
    B = B_values(i); % Current modulation index
    m = B * sin(2 * pi * fm * t); % Modulation signal
    y = Ac * cos((2 * pi * fc * t) + m); % FM signal

    n = length(y);
    Y = 2 * abs(fft(y)) / n;  % Compute FFT and normalize
    Y = Y(1:(n + 1) / 2);     % Take positive frequencies only
    d = 20 * log10(Y / sqrt(2)); % Convert to dBVrms
    f = linspace(0, fn, length(d)); % Frequency axis

    % Calculate bandwidth using Carson's rule
    BW = 2 * fm * (B + 1);
    bandwidths(i) = BW;

    % Find indices within the bandwidth range
    bw_indices = (f >= fc - BW / 2) & (f <= fc + BW / 2);
    peak_magnitudes{i} = max(d(bw_indices)); % Store the peak magnitud
```

```
    % Plot the frequency-domain representation for this B
    subplot(3, 1, i);
    plot(f, d, 'r', 'LineWidth', 1);
    xlim([fc - BW / 2, fc + BW / 2]); % Adjust to focus on bandwidth range
    ylim([-80 10]);      % Set y-axis for visibility
    title(['Frequency Spectrum for \beta = ' num2str(B)]);
    xlabel('Frequency (Hz)');
    ylabel('Magnitude (dBVrms)');
    grid on;
end

sgtitle('Frequency Modulated Signal Spectrum for Different Modulation Indices');

% Display results in a table
disp('Results:');
fprintf('%-20s %-20s %-20s\n', 'Modulation Index (β)', 'Bandwidth (Hz)', 'Peak Magnitude (dBrms)');
for i = 1:length(B_values)
    fprintf('%-20.1f %-20.0f %-20.2f\n', B_values(i), bandwidths(i), peak_magnitudes{i});
end
Results:
Modulation Index (β) Bandwidth (Hz)       Peak Magnitude (dBrms)
0.2                  12000                4.86
1.0                  20000                2.62
2.0                  30000                0.17
```

Below is the peak magnitude diagrams




Below is the table showing peak magnitude values of the plot at $B_f = 0.2$

| frequency(Hz) | Amplitude(dBVrms) |
|---|---|
| 25000 | -60.9429 |
| 30000 | -40.2841 |
| 35000 | -15.0258 |
| 40000 | 4.86024 |
| 45000 | -15.0277 |
| 50000 | -41.6164 |
| 55000 | -63.7573 |

Below is the table showing peak magnitude values of the plot at $B_f = 1$

| frequency(Hz) | Amplitude(dBVrms) |
|---|---|
| 15000 | -61.1418 |
| 20000 | -46.1792 |
| 25000 | -29.0448 |
| 30000 | -13.8035 |
| 35000 | -2.17328 |
| 40000 | 2.61643 |
| 45000 | -2.16904 |
| 50000 | -13.846 |
| 55000 | -29.1471 |
| 60000 | -47.5147 |
| 65000 | -64.6499 |

Below is the table showing peak magnitude values of the plot at $B_f = 2$

| frequency(Hz) | Amplitude(dBVrms) |
|---|---|
| 5000 | -63.3786 |
| 10000 | -51.9508 |
| 15000 | -37.7744 |
| 20000 | -24.3351 |
| 25000 | -12.8144 |
| 30000 | -4.09128 |
| 35000 | 0.164767 |
| 40000 | -8.08495 |
| 45000 | 0.16959 |
| 50000 | -4.09699 |
| 55000 | -12.828 |
| 60000 | -24.4257 |
| 65000 | -38.0179 |
| 70000 | -53.755 |
| 75000 | -68.4799 |

**Problem 3: Frequency demodulation**

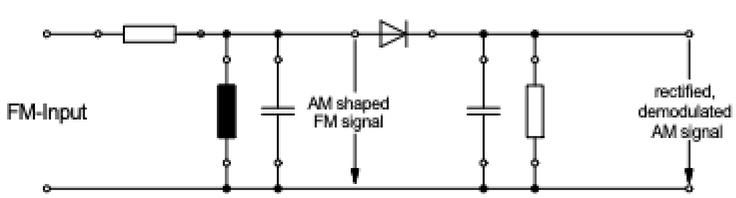



Figure 10.4: Schematic circuit of a slope detector.

The transfer function of the circuit above is:

$$H(s) = \frac{s/Q}{s^2 + \frac{s}{Q} + \omega_0^2}$$

Where:

$s = j\omega$

$Q = \frac{\omega_0 L}{R}$

$\omega_0 = 2\pi f_0$

$f_0 = \frac{1}{2\pi\sqrt{LC}}$

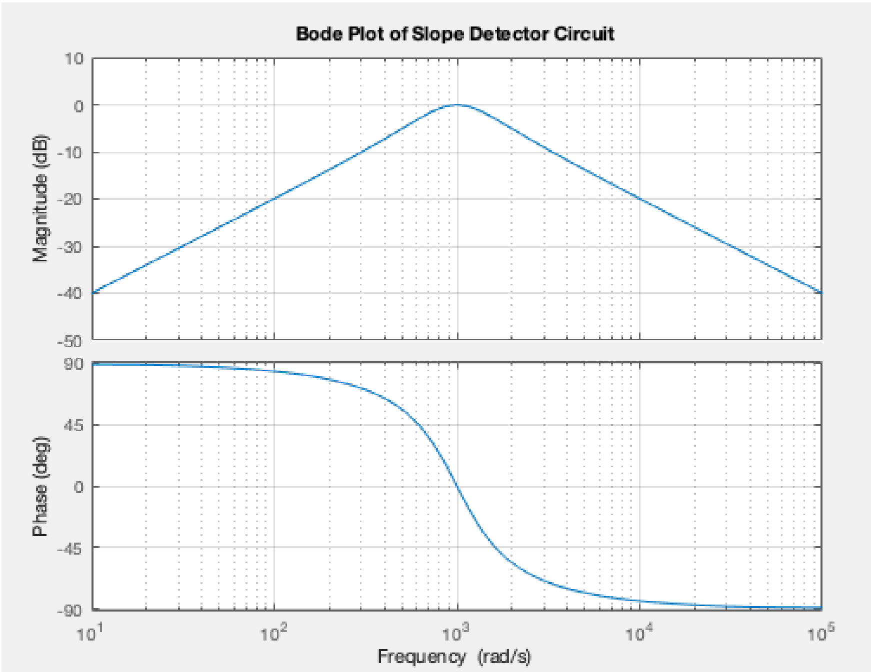
Giving us:

$H(s) = \frac{Ls}{RLCs^2 + Ls + R}$

Matlab code for the plots below:

```
>> % Define component values
C = 1e-6;       % Capacitance in farads (Te-06 should be 1e-6)
R = 100;        % Resistance in ohms
L = 0.1;        % Inductance in henries (1e-01 is correct here)

% Define the transfer function variable s
s = tf('s');

% Define the transfer function H(s)
H = (L * s) / (R * C * s^2 + L * s + R);

% Plot the Bode diagram
figure;
bode(H);
title('Bode Plot of Slope Detector Circuit');
grid on;
```

Below is the Bode plots of the circuit:




The basic principle of the slope detector circuit is to convert frequency fluctuations in the FM signal into amplitude variations by using phase shifts generated near the circuit's resonance. This amplitude modulation reflects the original communication signal. By tuning the circuit's resonance frequency slightly above the carrier frequency, the circuit can effectively capture these phase shifts, allowing for successful demodulation of the FM signal.