Design, Implementation, and Analysis of an AM Modulator and Demodulator

Electronic Workshop 2 - Project 2

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***Abstract*—The project focuses on the design, implementation and analysis of a Double Sideband Suppressed Carrier (DSB-SC) Amplitude Modulation(AM) modulator and demodulator circuit prototype.**

The main goal was to modulate the carrier signal, generated by the Local Oscillator (LO), using the modulating signal from the DSO, and to successfully recover the original signal.

Through practical implementation, the project demonstrates the fundamental concepts of Communication Theory. The circuit helps in understanding how information can be encoded and transmitted over a carrier signal using AM techniques.

I. Introduction

A. Background of Amplitude Modulation

Amplitude Modulation (AM) is a modulation technique where the amplitude (signal strength) of a carrier wave is varied in proportion to the message signal (e.g., audio). It's commonly used in electronic communication for transmitting messages via radio waves. In AM, carrier signal's amplitude, A(t), changes according to the message signal. This message signal defines the envelope of the transmitted waveform. In the frequency domain, AM produces a signal with power concentrated at the carrier frequency and two adjacent sidebands.

Types of Amplitude Modulation:

- *Double-Sideband Amplitude Modulation (DSB-AM):* The standard AM method generates sidebands on both sides of the carrier frequency. These sidebands contain the frequency components of the modulating signal and are symmetrically placed around the carrier frequency.
- *Single-Sideband Modulation (SSB):* Uses bandpass filters to eliminate one sideband and possibly the carrier, improving power efficiency and bandwidth utilization.
- *Quadrature Amplitude Modulation (QAM):* A more complex form of Amplitude modulation is often used with digital data to enable more efficient use of bandwidth.
- *Double-Sideband Suppressed Carrier (DSB-SC):* This technique removes the carrier signal, transmitting only the upper and lower sidebands. It improves power efficiency but requires coherent detection at the receiver.
- *Vestigial Sideband Modulation (VSB):* VSB retains part of one sideband while suppressing the rest of the unwanted sideband. This method helps reduce the bandwidth required for transmission, offering a more efficient use of spectrum compared to standard AM, while still allowing for accurate signal recovery at the receiver.
- *Pulse Amplitude Modulation (PAM):* PAM varies the amplitude of pulses rather than a continuous wave. It is used in digital communication systems as an intermediate step before converting signals to binary formats.

B. Objectives

Following are the primary objectives of this project:

1) To design and implement a Double Sideband Suppressed Carrier (DSB-SC) Amplitude Modulation (AM) modulator circuit capable of encoding a modulating signal onto a carrier signal generated by a Local Oscillator (LO).
2) To design and implement a corresponding demodulator circuit that can successfully recover the original modulating signal from the received DSB-SC AM signal.

II. Theoretical Background of AM

Modulation: In conventional AM, we add a large carrier component to a DSB-SC signal, so that the passband transmitted signal is of the form:

$$u_{AM}(t) = Am(t)\cos(2\pi f_c t) + A_c \cos(2\pi f_c t)$$

Taking the Fourier transform, we have

$$U_{AM}(f) = \frac{A}{2}\left(M(f-f_c) + M(f+f_c)\right) + \frac{A_c}{2}\left(\delta(f-f_c) + \delta(f+f_c)\right)$$

which means that, in addition to the USB and LSB due to the message modulation, we also have impulses at $\pm f_c$ due to the unmodulated carrier.

The key concept behind conventional AM is that, by making A_c large enough, the message can be demodulated using a simple envelope detector. Large A_c corresponds to expending

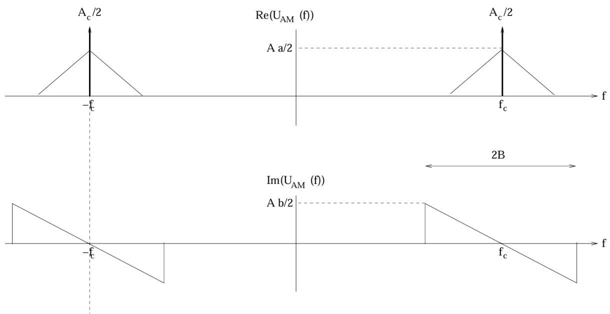

Fig. 1. Spectrum of Conventional AM

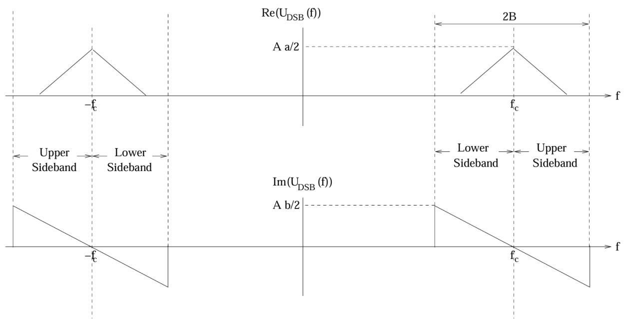

Fig. 2. Spectrum of DSB-SC AM

transmitter power on sending an unmodulated carrier which carries no message information, in order to simplify the receiver. This tradeoff makes sense in a broadcast context, where one powerful transmitter may be sending information to a large number of low-cost receivers, and is the design approach that has been adopted for broadcast AM radio.

The key issue with conventional AM is its inefficiency in terms of power utilization. This led to the development of DSB-SC modulation which addresses these issues by suppressing the carrier and improving the power efficiency.

$$u_{DSB}(t) = Am(t)\cos(2\pi f_c t)$$

Taking Fourier transforms, we have

$$U_{DSB}(f) = \frac{A}{2}\left(M(f - f_c) + M(f + f_c)\right)$$

Demodulation:

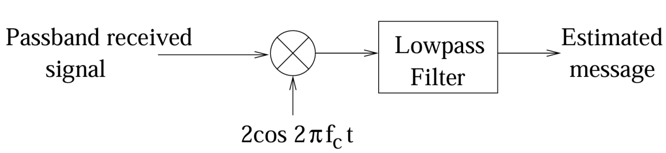

Fig. 3. Coherent Demodulation of AM

Multiply the received signal with the cosine of the carrier, and pass it through a low-pass filter. Ignoring noise, the received signal is given by

$$y_p(t) = Am(t)\cos(2\pi f_c t + \theta_r)$$

where θ_r is the phase of the received carrier relative to the local copy of the carrier produced by the receiver's local oscillator (LO), and A is the received amplitude, taking into account the propagation channel from the transmitter to the receiver. In order for this demodulator to work well, we must have θ_r as close to zero as possible; that is, the carrier produced by the LO must be coherent with the received carrier.

The effect of phase mismatch:

$$2y_p(t)\cos(2\pi f_c t) = Am(t)\cos(2\pi f_c t + \theta_r)\cos(2\pi f_c t)$$

$$= Am(t)\cos\theta_r + Am(t)\cos(4\pi f_c t + \theta_r)$$

We recognize the second term on the right-hand side as being a passband signal at $2f_c$ (since it is a baseband message multiplied by a carrier whose frequency exceeds the message bandwidth). It is therefore rejected by the low-pass filter. The first term is a baseband signal proportional to the message, which appears unchanged at the output of the LPF (except possibly for scaling), as long as the LPF response has been designed to be flat over the message bandwidth. The output of the demodulator is therefore given by

$$\hat{m}(t) = Am(t)\cos\theta_r$$

The demodulator output is proportional to the message, which is what we want, but the proportionality constant varies with the phase of the received carrier relative to the LO. In particular, the signal gets significantly attenuated as the phase mismatch increases, and gets completely wiped out for $\theta_r = \frac{\pi}{2}$.

Note that, if the carrier frequency of the LO is not synchronized with that of the received carrier (say with frequency offset Δf), then $\theta_r(t) = 2\pi\Delta f t + \phi$ is a time-varying phase that takes all values in $[0, 2\pi)$, which leads to time-varying signal degradation in amplitude, as well as unwanted sign changes. Thus, for coherent demodulation to be successful, we must drive Δf to zero, and make ϕ as small as possible; that is, we must synchronize to the received carrier.

III. Circuit Design & Implementation

A. Local Oscillator

To generate the carrier signal (both at the transmitter end and the Local Oscillator (LO) signal for coherent demodulation at the receiver end), we have used a Wein Bridge Oscillator. The Wein Bridge Oscillator is a type of electronic oscillator that generates sine waves. It is based on the principle of positive and negative feedback, using a bridge circuit with resistors and capacitors to determine the frequency of oscillation. The circuit is known for its simplicity and ability to produce low-distortion sine waves.

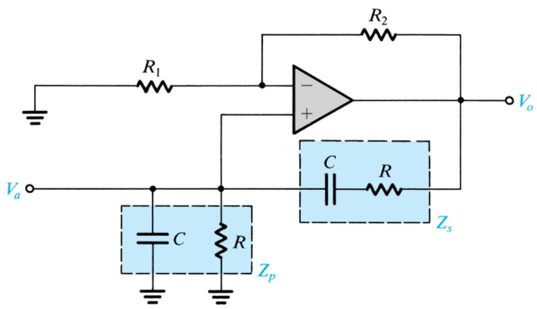

Fig. 4. Circuit diagram of the Wein Bridge Oscillator

$$
\begin{aligned}
L(s) &= \left(1 + \frac{R_2}{R_1}\right) \cdot \frac{Z_p}{Z_p + Z_s} \\
&= \frac{1 + \frac{R_2}{R_1}}{1 + Z_s Y_p} \\
&= \frac{1 + \frac{R_2}{R_1}}{3 + sCR + \frac{1}{sCR}} \qquad (1)\\
L(j\omega) &= \frac{1 + \frac{R_2}{R_1}}{3 + j\left(\omega CR - \frac{1}{\omega CR}\right)} \qquad (2)
\end{aligned}
$$

For zero phase (i.e., real loop gain):

$$
\begin{aligned}
\omega_0 CR &= \frac{1}{\omega_0 CR} \\
\Rightarrow \quad \omega_0 &= \frac{1}{CR} \qquad (3)
\end{aligned}
$$

Barkhausen Criterion requires:

$$|L(j\omega_0)| = 1 \quad \text{and} \quad \angle L(j\omega_0) = 2n\pi \qquad (4)$$

To satisfy this, the gain condition is:

$$\frac{R_2}{R_1} > 2$$

So, to get oscillation of 100kHz, we used values:

$$R = 12k$$

$$C = 35p$$

We get

$$\frac{1}{2\pi RC} \approx 100kHz$$

. In simulations we get the oscillating frequency as 103 kHz.

The amplitude of oscillation can be determined and stabilized by using a nonlinear con trol network. The below circuit is the Wein Bridge Oscillator with amplitude stabilization. The circuit uses the popular limiter circuit using diodes for amplitude control. We used diodes to limit the amplitude of the oscillations. It is formed by diodes D_1 and D_2 together with resistors R_3, R_4, R_5, and R_6. The limiter operates in the following manner: At the positive peak of the output voltage v_O, the voltage at node b will exceed the voltage v_1 (which

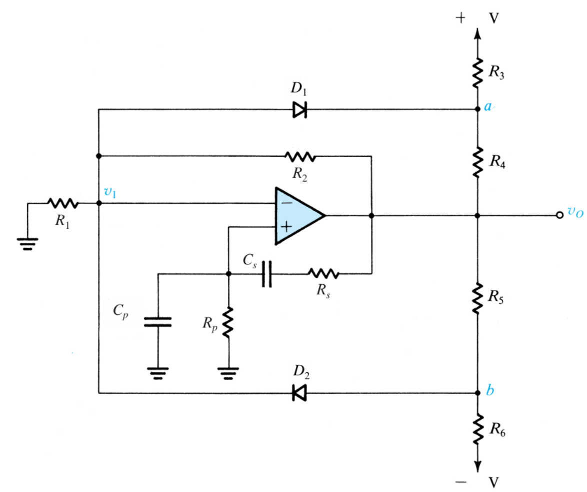

Fig. 5. Circuit diagram of the Wein Bridge Oscillator with amplitude stabilization

is about 1 $3v_O$), and diode D_2 conducts. This will clamp the positive peak to a value determined by R_5, R_6, and the negative power supply. To be specific, the value of the positive output peak can be calculated by setting $v_b = v_1 + V_{D2}$ and writing a node equation at node b while neglecting the current through D_2. Similarly, the negative peak of the output sine wave will be clamped to the value that causes diode D_1 to conduct. The value of the negative peak can be determined by setting. Taking

$$R_4 = R_7 = 2k$$

$$R_5 = R_6 = 1k$$

We get a max amplitude of 1.5V.

B. Modulator

To perform modulation in our circuit, we have chosen to implement a switching mode modulator using a MOSFET as a switch. These kind of mixer circuits have high efficiency because a square wave is generated by turning the circuit on and off. When the circuit is off, no power is consumed. This circuit provides more linearity compared to non-linear mixers with a simpler design.

To make this switching modulator in LTSpice, we have used a npn MOSFET. The circuit is designed to modulate the message signal with the carrier signal. The message signal is fed into the gate of the MOSFET, while the carrier signal is fed into the drain. The output is taken from the source of the MOSFET. The circuit is designed to work in a switching mode, where the MOSFET operates as a switch (Voltage controlled resistor), turning on and off based on the input signals.

Considering m(t) as the message signal and f(t) as the carrier signal. For the given design to work as a mixer, it is really important to set the DC bias of the npn MOSFET used to its threshold. When the carrier wave (sinusoid produced by the oscillator) is added to this bias, then the MOSFET turns on

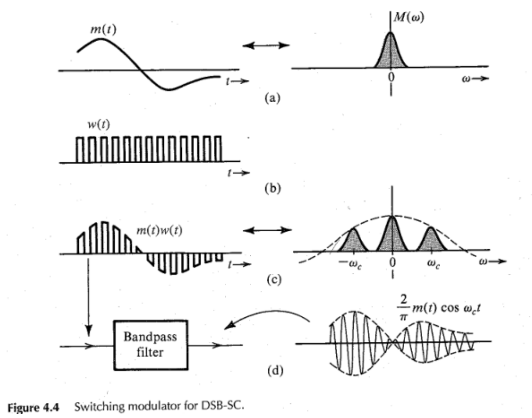

Fig. 6. Working of the Switching Modulator Used in the Project

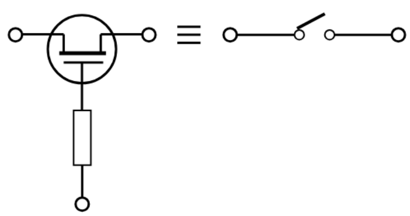

Fig. 7. FET as a Mixer

in the positive cycle and turns off during the negative cycle. This happens because the operating point is set at V_{TH}. This effectively multiplies our message signal with a square wave of the frequency of our carrier. Let this square wave be denoted by w(t). A a square wave is a periodic wave with a valid Fourier series expression. The Fourier series representation of w(t) and the result of the multiplication is given by:

$$w(t) = \frac{1}{2} + \frac{2}{\pi}(cos\ w_c t - \frac{1}{3}cos\ 3w_c t + \frac{1}{5}cos\ 5\omega_c t\ -\ ....)$$

$$m(t)w(t) = \frac{1}{2}m(t) + \frac{2}{\pi}(m(t)cos\ w_c t - \frac{1}{3}m(t)cos\ 3w_c t +$$

$$\frac{1}{5}m(t)cos\ 5\omega_c t\ -\ ....)$$

The required harmonic can be retrieved by using a bandpass filter whenever required.

Applying Kirchoff's current law, we can say that $i_2 + i_1 = i_3 + i_4$. But as there is a capacitor blocking the path of i_2 and there will be no flow of current into the gate of the MOSFET. Hence, $i_3 = 0$. This gives us, $i_1 = i_4$. Now, we want the voltage divider formed by $R_8 and R_9$ to give 720 mV (V_{th} of our MOSFET) as the voltage output across R_9. Choosing $R_8 = 770k$ (using a high value to reduce current draw), we can calculate R_9 as:

$$R_9 = \frac{V_{out} \cdot R_8}{V_{in} - V_{out}}$$

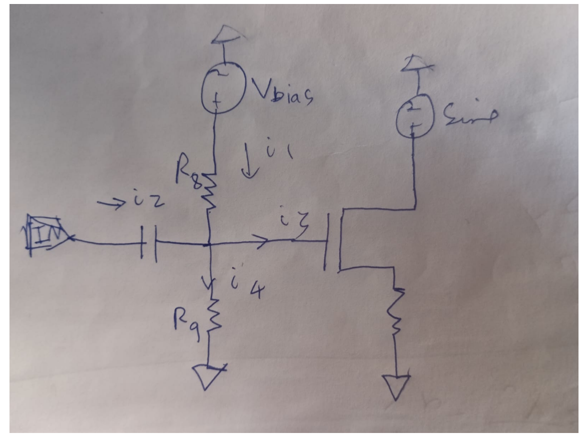

Fig. 8. Mixer DC Bias Calculation

$$R_9 = \frac{0.720 \times 770000}{10 - 0.720}$$

$$R_9 \approx 60k$$

C. Demodulator

The demodulator circuit is designed to recover the original message signal from the modulated DSB-SC signal. The demodulation process involves multiplying the received DSB-SC signal with a local oscillator (LO) signal that is coherent with the carrier frequency of the received signal. This multiplication process effectively shifts the frequency components of the modulated signal down to baseband, allowing for easy extraction of the original message signal.

This multiplication is done using a switching mode demodulator, similar to the modulator circuit. The output of the demodulator is then passed through a low-pass filter (LPF) to remove the high-frequency components and recover the original message signal.

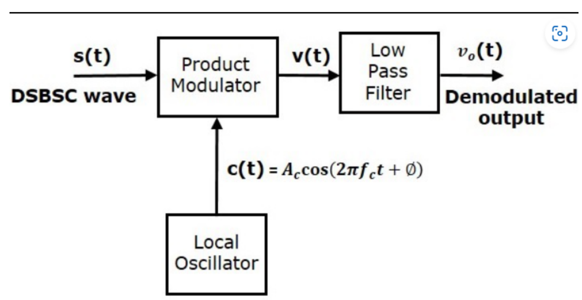

Fig. 9. DSBSC Demodulator Block Diagram

The output of the multiplier is then passed through a low-pass filter (LPF) to remove the high-frequency components and recover the original message signal. The LPF is designed to have a cutoff frequency that allows the baseband message signal to pass through while attenuating the higher frequency components.

The LPF is designed to have a cutoff frequency that allows the baseband message signal to pass through while attenuating

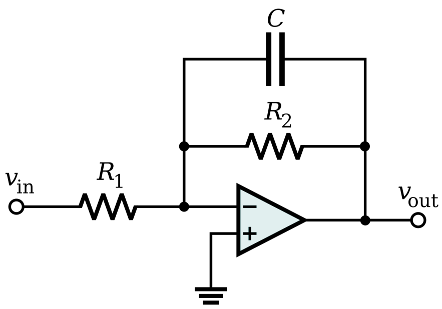

Fig. 10. Active Low Pass Filter

the higher frequency components. The transfer function of the LPF is given by:

The impedance of the feedback network is:

$$Z_f = R_2 \parallel \frac{1}{sC} = \frac{R_2}{1 + sR_2C}$$

The input impedance is simply:

$$Z_1 = R_1$$

Using the standard gain formula for a non-inverting amplifier:

$$H(s) = \frac{V_{\text{out}}(s)}{V_{\text{in}}(s)} = 1 + \frac{Z_f}{Z_1}$$

Substituting Z_f and Z_1:

$$H(s) = 1 + \frac{1}{R_1} \cdot \frac{R_2}{1 + sR_2C}$$

$$\boxed{H(s) = 1 + \frac{R_2}{R_1(1 + sR_2C)}}$$

To get the 3Db cut off frequency we can equate the modulus of the transfer function to $\frac{H(j\infty)}{\sqrt{2}}$.
Doing that we get:

$$1 + j\omega_c R_2 C = 3$$

Putting $\omega = 2\pi f$ we get,

$$f_c = \frac{1}{2\pi R_2 C}$$

We chose our $C = 1n$ (for easy availability). We wanted a cutoff of around 1kHz, so we got

$$R_2 = \frac{1}{2\pi f_c C}$$

$$R_2 \approx 150k\Omega$$

The gain of the filter is given by $-\frac{R_1}{R_2}$. We wanted a gain of around 2.5 so, $R_1 = 60k$.

The output of the LPF is the recovered message signal, which should closely resemble the original message signal before modulation. The performance of the demodulator circuit is evaluated by comparing the recovered message signal with the original message signal.

IV. Experimental Setup & Methodology

A. LTSpice Simulation

The full circuit designed in LTSpice with **initial calculations**.

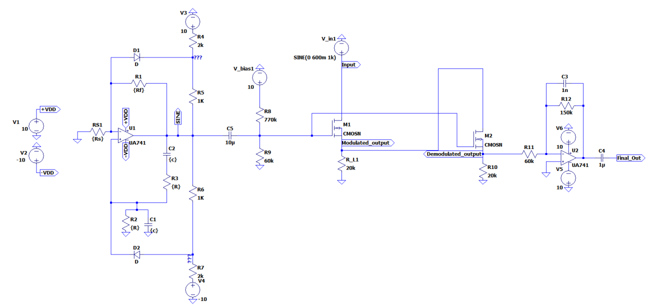

Fig. 11. LTSpice Simulation of the Full Circuit

Final Output of the circuit in LTSpice:
Individual Stages in LTSpice & their outputs:

- *Local Oscillator:* The local oscillator circuit is designed to generate a carrier signal at a frequency of 100 kHz. The output of the oscillator is a sine wave with a frequency of 100 kHz. The circuit uses a Wein Bridge Oscillator configuration with amplitude stabilization.

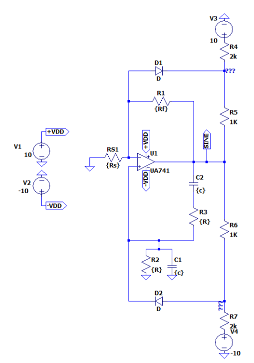

Fig. 12. Circuit of the Local Oscillator (Simulation)

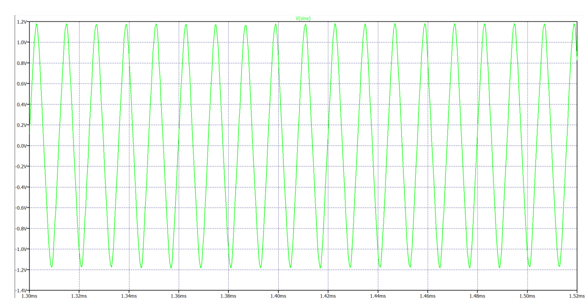

Fig. 13. Simulation output of the Oscillator

The fft of the output of the oscillator is shown below:

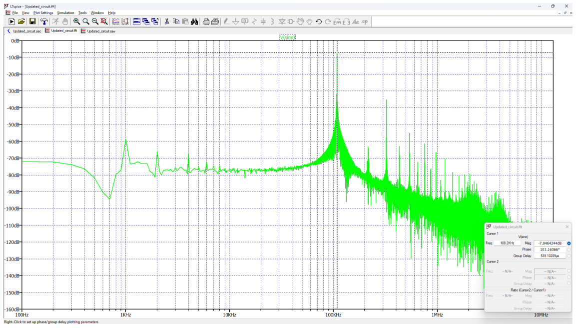

Fig. 14. FFT of the output of the oscillator (Simulation)

- Modulator : The modulator circuit is designed to modulate the message signal with the carrier signal generated by the local oscillator. The circuit uses a MOSFET as a switch to perform the modulation. The output of the modulator is a DSB-SC signal.

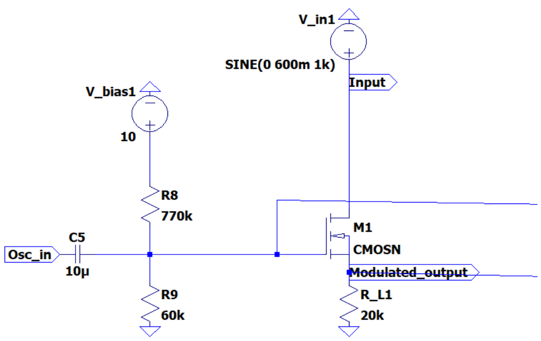

Fig. 15. Circuit diagram of the modulator

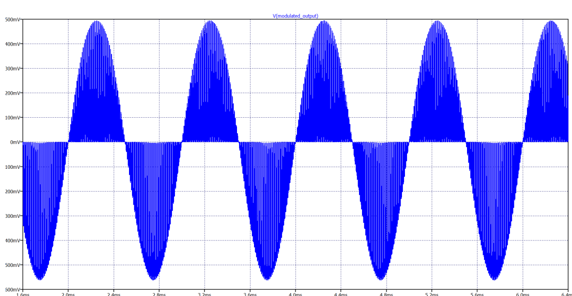

Fig. 16. Output of the modulator circuit (Simulation)

- Demodulator: The demodulator circuit is designed to recover the original message signal from the modulated DSB-SC signal. The circuit uses a switching mode demodulator, similar to the modulator circuit. The output of the demodulator is then passed through a low-pass filter (LPF) to remove the high-frequency components and recover the original message signal.

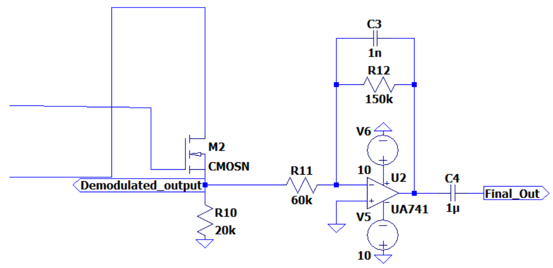

Fig. 17. Circuit diagram of the demodulator

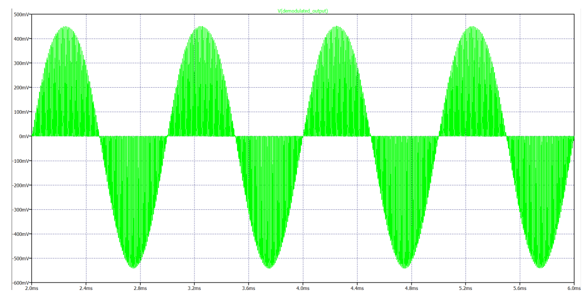

Fig. 18. Output of the demodulator circuit (Simulation)

The demodulated output is then passed through a low-pass filter (LPF) to remove the high-frequency components and recover the original message signal. The LPF is designed to have a cutoff frequency that allows the baseband message signal to pass through while attenuating the higher frequency components.

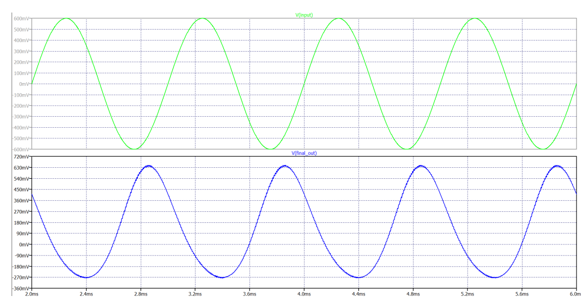

Fig. 19. Message and retrieved message signal (Simulation)

B. Hardware Implementation

The hardware implementation of the circuit was done using a breadboard and various electronic components. The components used in the circuit include resistors, capacitors, MOSFETs, and operational amplifiers. The circuit was powered using a DC power supply, and the output signals were observed using an oscilloscope. The circuit was built in a modular fashion, with each stage (oscillator, modulator, and demodulator) being tested individually before integrating them into the complete system. The performance of the circuit was evaluated by observing the output waveforms at various points in the circuit using an oscilloscope.

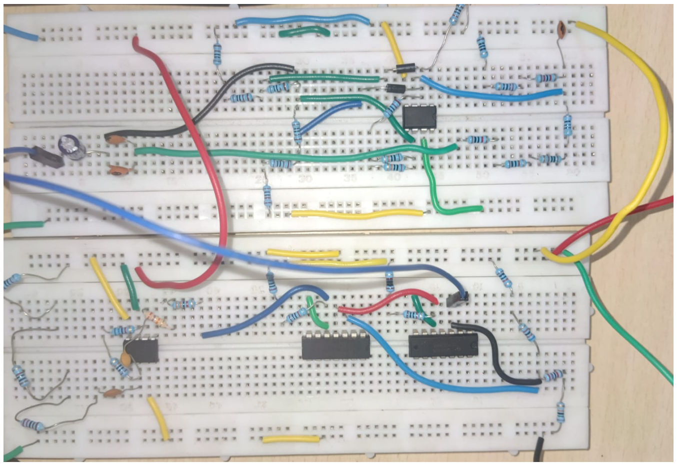

Fig. 20. Image of the hardware implementation of the circuit

Final output from the hardware implementation:

- Local Oscillator:

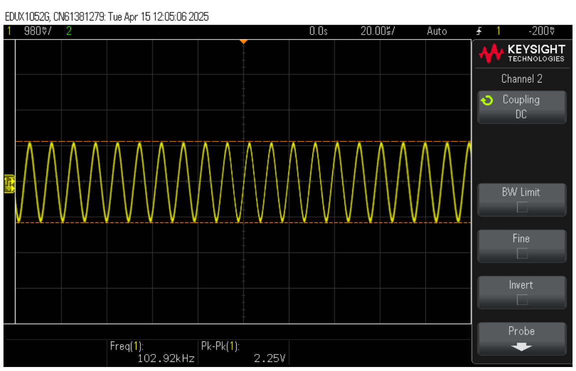

Fig. 21. Output of the oscillator (Hardware)

- Modulator

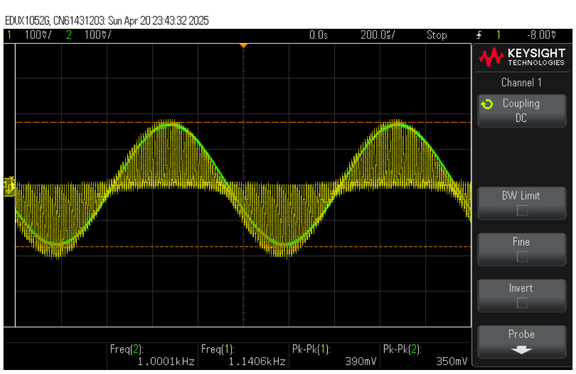

Fig. 22. Modulated output of the modulator circuit (Hardware)

The FFt of the output of the modulator circuit is shown below:

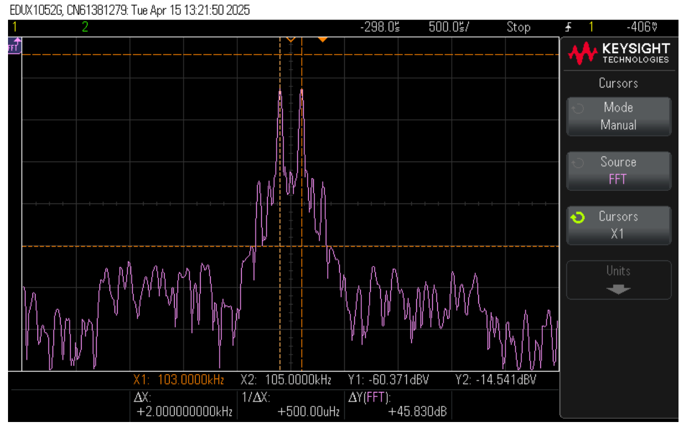

Fig. 23. FFT of the modulated output (Hardware)

- Demodulator Channel 1 (Yellow) is the modulated output from the modulator circuit. Channel 2 (Green) is the output of the demodulator circuit.

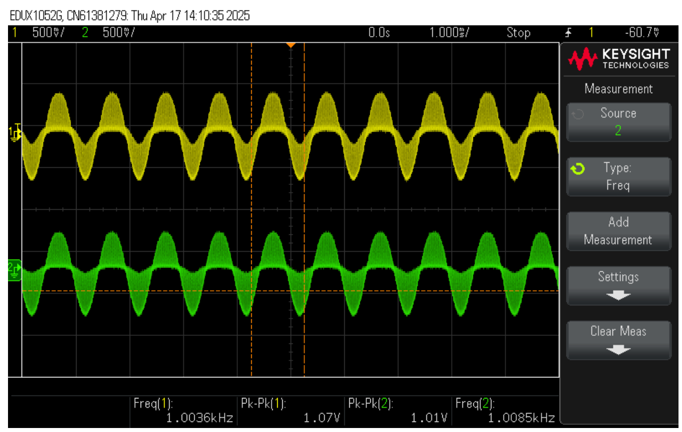

Fig. 24. Modulated and demodulated Output (Hardware)

The FFT of the output of the demodulator circuit is shown below:

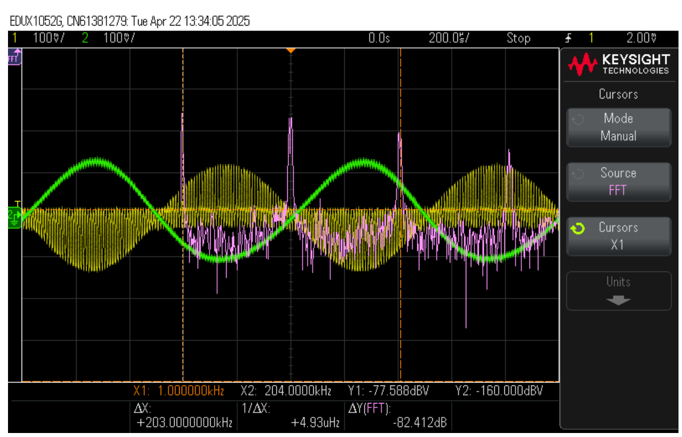

Fig. 25. FFT of the demodulated output (Hardware)

The filter response of the low-pass filter used in the demodulator circuit is shown below:

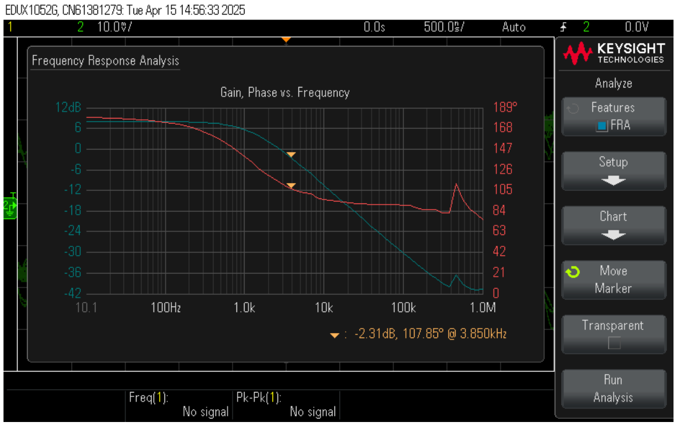

Fig. 26. Bode plot of the filter circuit (Hardware)

After passing the output of the demodulator circuit through the low-pass filter, we get the final output. The FFT of the final output is shown below:

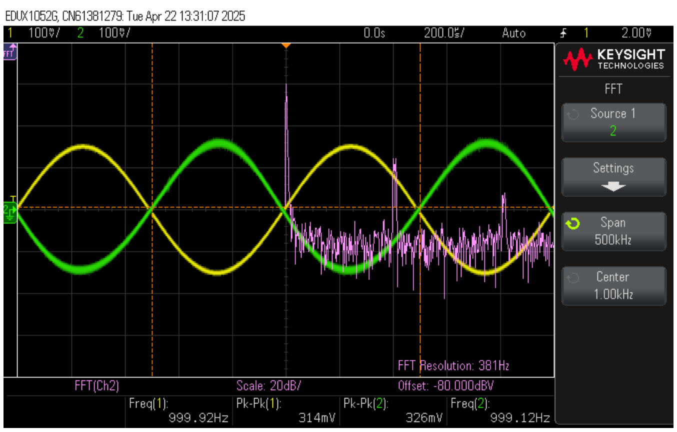

Fig. 27. FFT of the recovered message signal (Hardware)

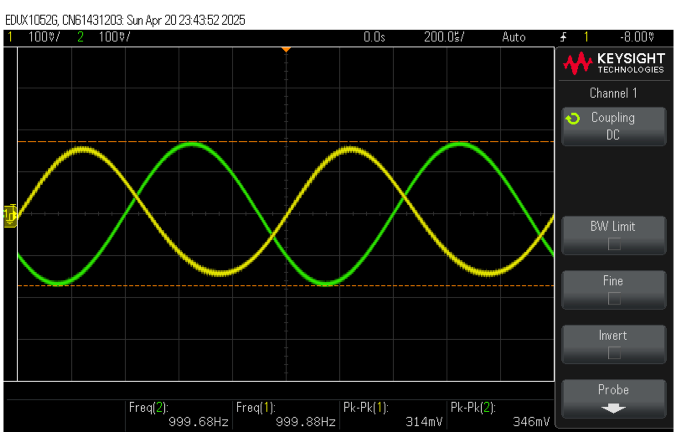

Fig. 28. Input message signal and retrieved message signal (Hardware)

At the end of the demodulator circuit, we get the retrieved message signal. to compensate the extra phase shift that was introduced to the non linearities in the mixer we've added a phase shiftor at the end . The retrieved message signal is compared with the original message signal (with a 180 degree phase difference, due to the inverting filter used) to evaluate the performance of the demodulator circuit. The output of the demodulator circuit closely resembles the original message signal, indicating that the demodulation process was successful.

TABLE I
COMPONENT VALUES: SIMULATION VS. BREADBOARD IMPLEMENTATION

No.	Component	Simulated Value	Breadboard Value
1	RS1	10 kΩ	10 kΩ
2	R_1	50 kΩ	50.4 kΩ
3	R_2, R_3	12 kΩ	12.2 kΩ
4	R_4	2 kΩ	2 kΩ
5	R_7	2 kΩ	1.9 kΩ
6	R_5	1 kΩ	1.1 kΩ
7	R_6	1 kΩ	1 kΩ
8	R_8	770 kΩ	775 kΩ
9	R_9	60 kΩ	66 kΩ
10	R_{L1}	20 kΩ	19.8 kΩ
11	R_{10}	20 kΩ	20 kΩ
12	R_{11}	60 kΩ	60.7 kΩ
13	R_{12}	150 kΩ	155.4 kΩ
14	C_1, C_2	35 pF	31 pF

V. CONCLUSION

The project successfully demonstrated the design, implementation, and analysis of a Double Sideband Suppressed Carrier (DSB-SC) Amplitude Modulation (AM) modulator and demodulator circuit prototype. The experimental results confirmed the effectiveness of the DSB-SC technique in modulating and demodulating signals, showcasing its potential for efficient communication systems.The project also highlighted the importance of coherent detection and the challenges associated with phase synchronization in demodulation.

Github: Repository Link

Youtube: Video Link

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