Communication Systems 1 Report

Amplitude Modulation and Demodulation of an Audio

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1. Introduction

Amplitude Modulation (AM) is one of the foundational techniques in analog communication systems and has been widely used in radio broadcasting and long-distance transmission for over a century. Despite the advancement of modern digital and modulation schemes, AM remains an essential method for understanding the principles of signal modulation and demodulation.

This project focuses on **transmitting and recovering recorded audio using Standard AM.** The aim is to demonstrate how a real audio signal can be encoded onto a high-frequency carrier, transmitted as an AM waveform, and then successfully recovered using **envelope detection**, one of the simplest and most practical demodulation techniques.

The project includes both the modulation and demodulation processes implemented in MATLAB. A recorded audio file serves as the message signal, which is modulated using a sinusoidal carrier with a frequency that the user can select. At the receiver end, the AM signal is demodulated using a rectifier followed by a low-pass filter to recover the original audio.

Through careful selection of carrier frequency, modulation index, and filtering parameters, this project demonstrates the conditions required for successful AM transmission and highlights the strengths and limitations of Standard AM. The final objective is to obtain a recovered audio signal that closely resembles the original message.

2. Problem Requirements and Constraints

Designing an AM system for real audio transmission must follow specific requirements:

Project Requirements

- Use a recorded audio signal as the message (mp3/ wav).
- Use **Standard AM** for modulation.
- Select an appropriate **carrier frequency** significantly higher than the message bandwidth and select a **carrier amplitude.**
- Apply **envelope detection** for demodulation.
- Recover audio that is both **audible and visually similar** to the original signal.
- Present clear plots for each major step: carrier, message, modulated signal, demodulated signal.

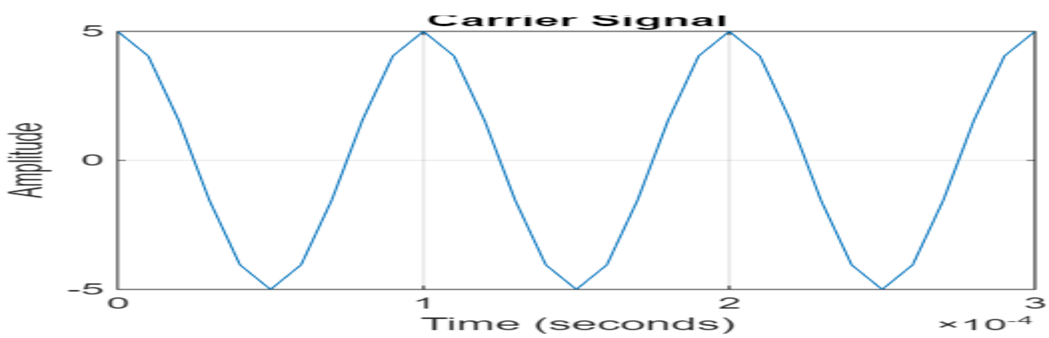
Technical Constraints

1. **Modulation Index (< 1):**
   To avoid over-modulation:

   K_a<1 and K_a >0 (sensitivity of the amplitude should be bigger than 0 and smaller than 1)

   Otherwise, the envelope becomes distorted, making recovery impossible.

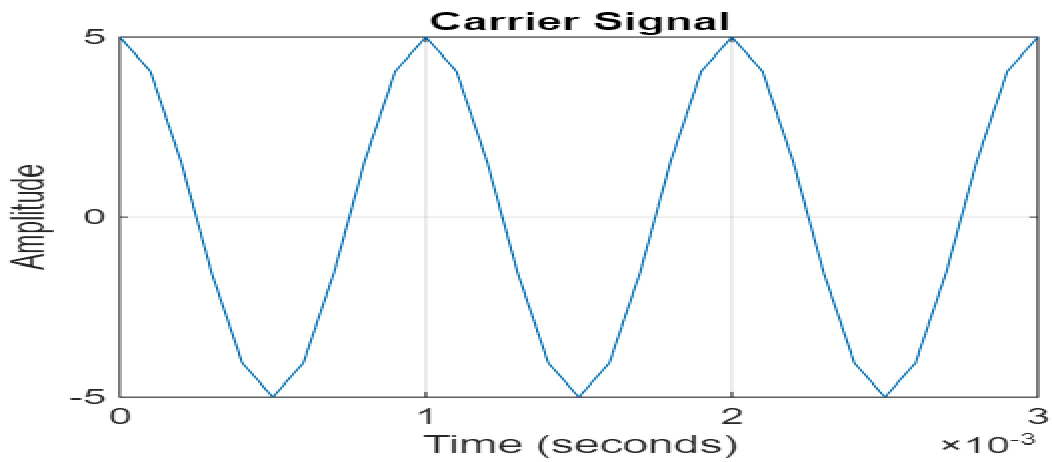
```
To get started, type doc.
For product information, visit www.mathworks.com.

Enter the carrier frequency (Hz): 10000
Enter the carrier amplitude: 5
Enter the amplitude sensitivity (0 < K_a < 1) : 5
```




```
The amplitude sensitivity needs to be lower than 1
Cannot perform modulation with K_a = 5.00
```

Figure 1 shows what happens if ka is bigger than 1

```
Enter the carrier frequency (Hz): 1000
Enter the carrier amplitude: 5
Enter the amplitude sensitivity (0 < K_a < 1) : -5
```




```
The amplitude sensitivity needs to be bigger than 0
Cannot perform modulation with K_a = -5.00
```

Figure 2 shows what happens if ka is smaller than 0

2. **Carrier Frequency:**
   Must be at least 3 times higher than the maximum audio frequency.
   Example:
   - Audio bandwidth ≈ 5 kHz
   - Carrier chosen = 15 kHz
3. **Sampling Rate Constraints:**
   The system must use the **audio file's sampling rate** (typically 44.1 kHz) to avoid aliasing.
4. **Demodulation Filter Design:**
   A low-pass filter must:
   - Pass all audio frequencies (0–5 kHz)
   - Strongly attenuate carrier remnants (15 kHz and harmonics)
5. **Signal Quality Requirement:**
   The recovered signal must be clear enough to be listened to through MATLAB.

3. Background

Amplitude Modulation was introduced in the early 1900s to allow voice and music to be transmitted over radio. Its role was crucial in replacing Morse code with audio broadcasting.

In AM, the amplitude of a high-frequency carrier signal is varied proportionally to the instantaneous value of the message signal. The transmitted signal contains:

- a strong carrier,
- an upper sideband,
- a lower sideband.

AM is simple and cost-effective because demodulation requires only envelope detection. However, it is inefficient in both power and bandwidth compared with modern techniques.

Despite limitations, AM is still widely used in:

- Aviation (air-to-ground communications)
- Standard AM radio broadcasting (MW and SW)
- Emergency and long-distance communications

This project uses AM as an educational model to demonstrate modulation, demodulation, and filtering principles.

4.1 Theoretical Modeling

Let the message audio signal be m(t) and the carrier be c(t):

$c(t) = A_c * \cos(2\pi * f_c * t)$

$m(t) = audio$

Standard AM is generated by:

$S(t) = A_c[1 + K_a * m(t)] * \cos(2\pi * f_c * t)$

Where:

- A_c = carrier amplitude
- f_c = carrier frequency
- f_m = highest frequency in m(t)

This creates a carrier plus two sidebands at:

$$f_c - f_m \ , \ f_c + f_m$$

Demodulation – Envelope Detection

Envelope detection consists of:

1. **Rectification:**

   $S_{rect}(t) = |S(t)|$

2. **Low-Pass Filtering:**
   Remove carrier frequencies and retain the message.

The output of the filter approximates m(t) often with a DC offset that can be removed by subtracting the mean.

4.2 Comparative Literature Review

AM is compared with other major modulation schemes:

Feature	Standard AM (Used)	DSB-SC	FM
Carrier	Present (large)	Suppressed	No amplitude variation
Power Efficiency	Low	Medium	High
Demodulation	Very simple (envelope)	Requires synchronous detection	Simple, but higher bandwidth
Noise Immunity	Poor	Better	Excellent
Bandwidth	2 × message BW	Same	Much larger

Key conclusions from literature:

- AM is easiest to implement and understand.
- Envelope detection makes it ideal for low-cost receivers.
- Power inefficiency and noise sensitivity are its main drawbacks.
- For educational purposes, AM clearly demonstrates fundamental communication concepts.

5. Solution Design

The solution is implemented entirely in MATLAB and follows a structured process that allows the user to choose the carrier frequency, carrier amplitude, and amplitude sensitivity before performing AM modulation. The design ensures full control over system parameters while supporting real audio signals.

The solution consists of the following steps:

The program begins by asking the user to enter:

- **Carrier frequency** f_c
- **Carrier amplitude** A_c
- **Amplitude sensitivity** K_a where $0 < K_a < 1$

  These values are crucial because:

- f_c determines where the audio spectrum will be shifted
- A_c controls amplitude scaling
- K_a sets the modulation index

The code prevents invalid values ($K_a > 1$ and $K_a < 0$) to avoid overmodulation.

The carrier signal is generated using the audio file's native sampling rate

f_s = f_audio

$c(t) = A_c * \cos(2\pi * f_c * t)$

where the time vector (t) matches the audio duration. This ensures both signals share the same time base for proper modulation.

A plot is generated to visually confirm that the carrier is correctly constructed.

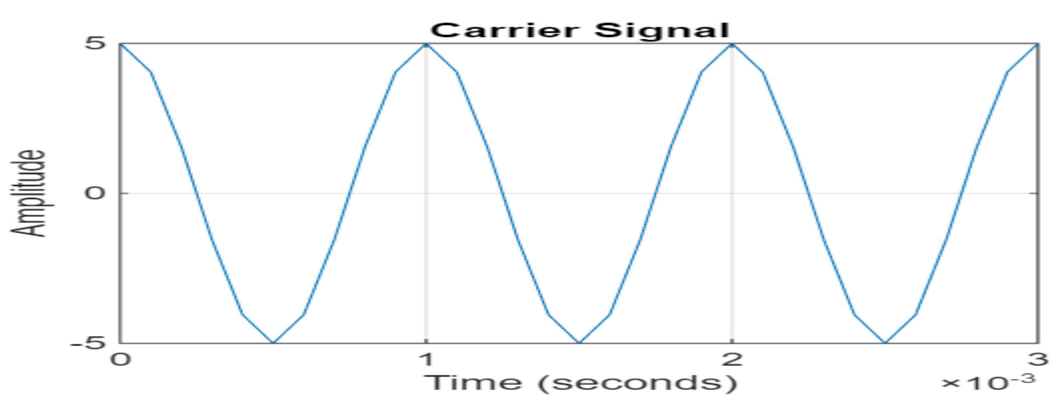


Figure 3 shows the carrier signal plot in time domain

After that, the user is prompted to select an audio file (.wav or .mp3).

Key preprocessing steps:

1. **Convert stereo to mono**
2. **Normalize audio to [−1,1]**
3. **Use audio's native sampling rate**

This ensures that the modulation occurs at the correct resolution for the audio signal.

The message is defined as:

m(t)=audio

A plot is generated to visually confirm that the message is correctly constructed.

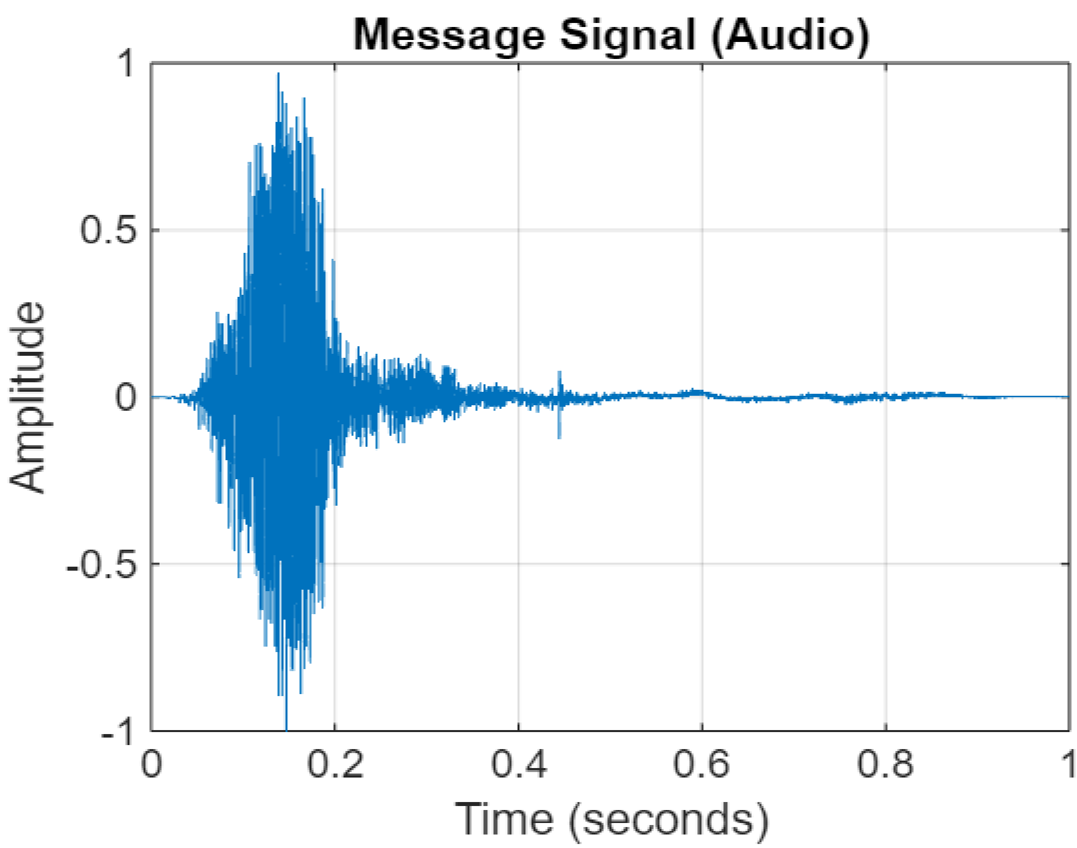


Figure 4 message signal in time domain (which was a dog bark in this case)

The AM signal is generated using the Standard AM formula:

$S(t) = A_c[1+K_a*m(t)]*\cos(2\pi*f_c*t)$

The code then plots:

- The original message signal
- The first portion of the AM-modulated waveform

These visual checks confirm correct amplitude scaling and carrier embedding.

To inspect the AM spectrum, an FFT is applied:

1. Ensure number of samples N is even
2. Compute FFT
3. Apply fftshift()
4. Construct frequency axis

The resulting plot shows:

- Positive and negative **carrier deltas** at $\pm f_c$
- Spread-out **sidebands** around the carrier
- Baseband spectrum centered at 0 Hz

Additional plots include theoretical delta functions for comparison, illustrating:

$A_c/2[\delta(f-f_c)+\delta(f+f_c)] + K_a*A_c/2 *M(f\pm f_c)$

These plots serve as clear visual confirmation of AM spectral behavior.

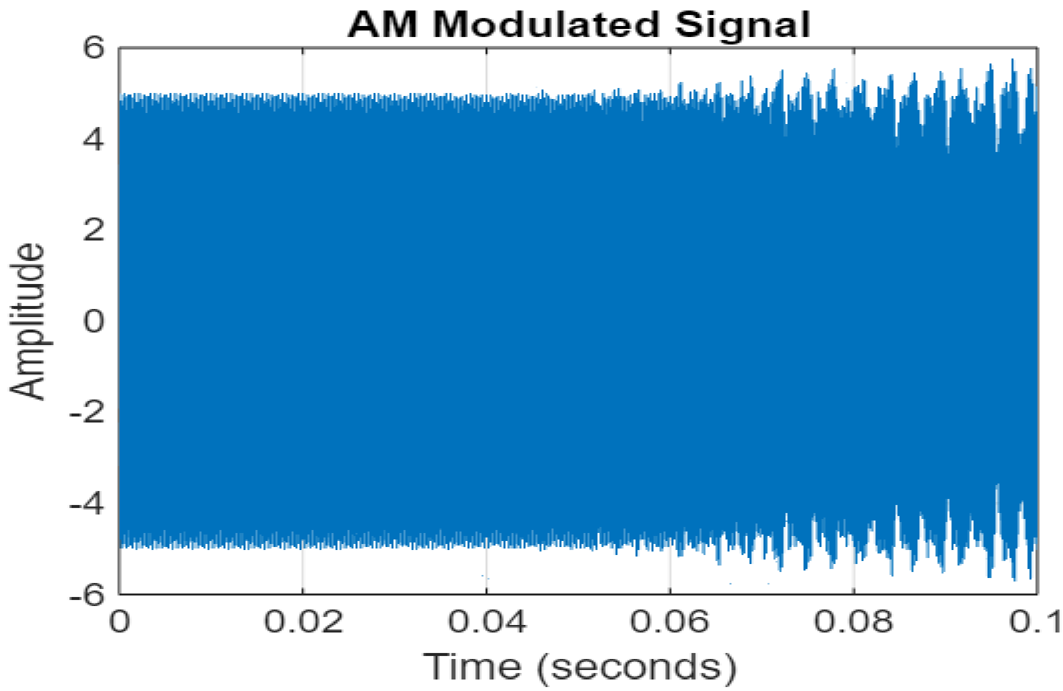


Figure 5 amplitude modulation signal in time domain

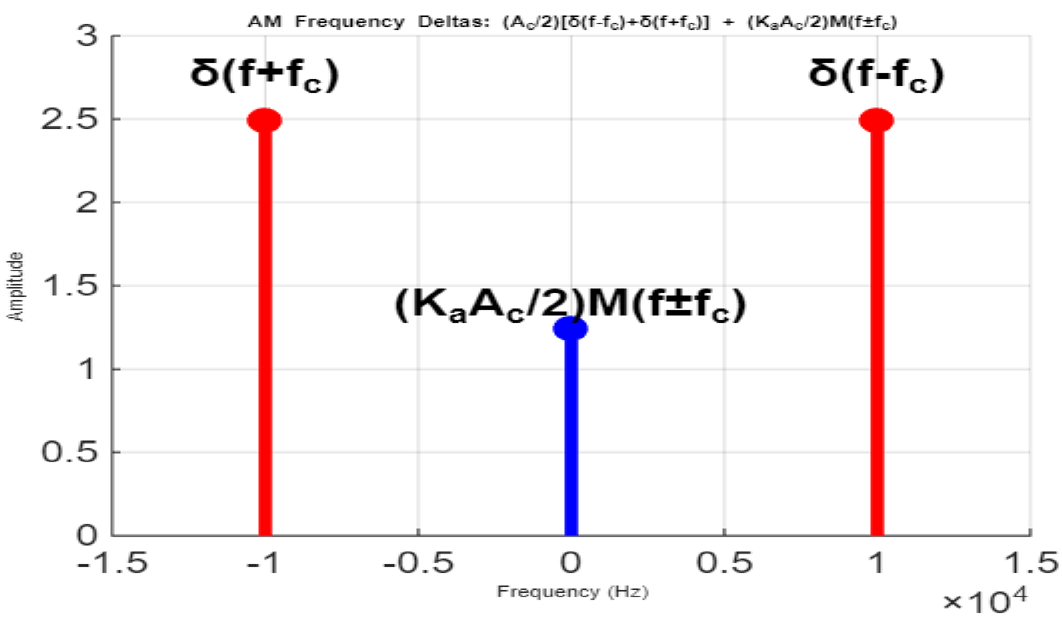


Figure 6 amplitude modulated signal in frequency domain

The demodulation stage of the system uses **envelope detection**, which is the classical recovery method for Standard AM (DSB-LC). The process is implemented in MATLAB as a sequence of three core steps: rectification, low-pass filtering, and DC-offset removal.

This subsection corresponds directly to the block diagram (Rectifier → LPF → Audio Output).

The rectifier stage emulates the operation of a diode by taking the **absolute value** of the AM signal:

$r(t)=|S(t)|$

This step removes negative excursions of the waveform and exposes the envelope, but also introduces high-frequency harmonics around the carrier frequency and its multiples.

```
rectified_S_t = abs(S_t);
```

After rectification, the signal still contains:

- the desired envelope (baseband message),
- the original carrier f_c
- and harmonics at $2*f_c$, $3*f_c$

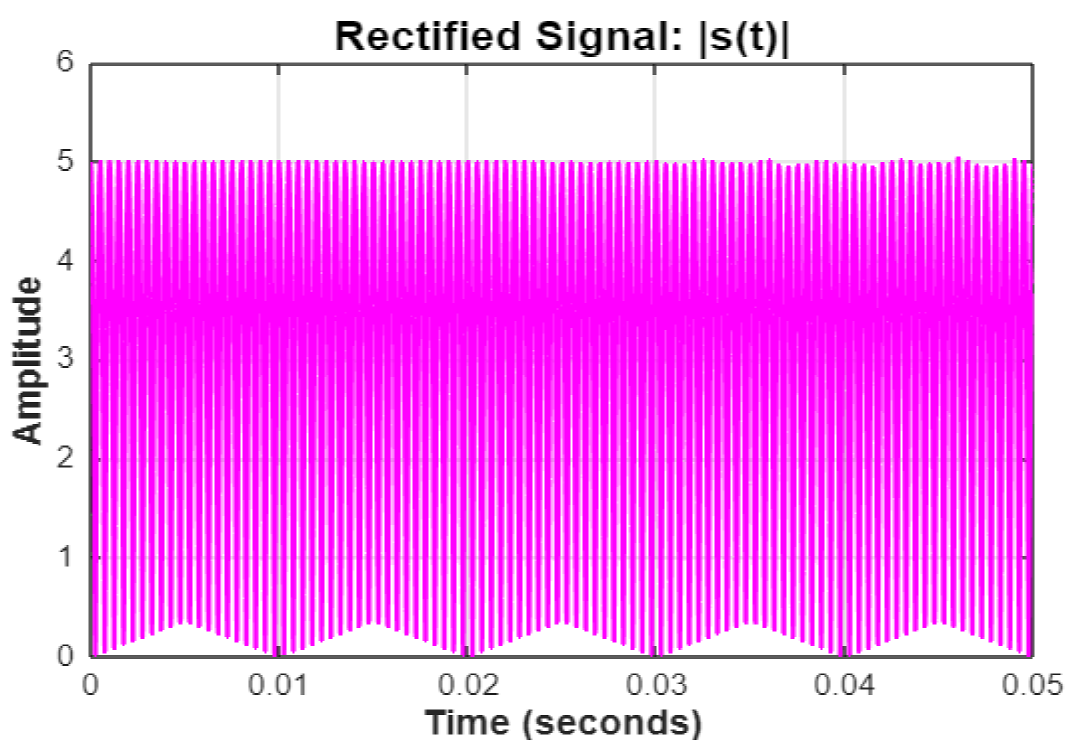


Figure 7 rectified signal in time domain

To recover only the message signal, a **high-order low-pass filter** is applied:

- Filter Order: **100 taps**
- Cutoff Frequency: **≈ 5 kHz** (enough to capture the entire audio band)
- Filter Type: **low-pass (linear phase)**
- Applied with **filtfilt** to avoid phase distortion

The filtering operation extracts:

$e(t)=LPF\{r(t)\}$

envelope = filtfilt(lpFilt, rectified_S_t);

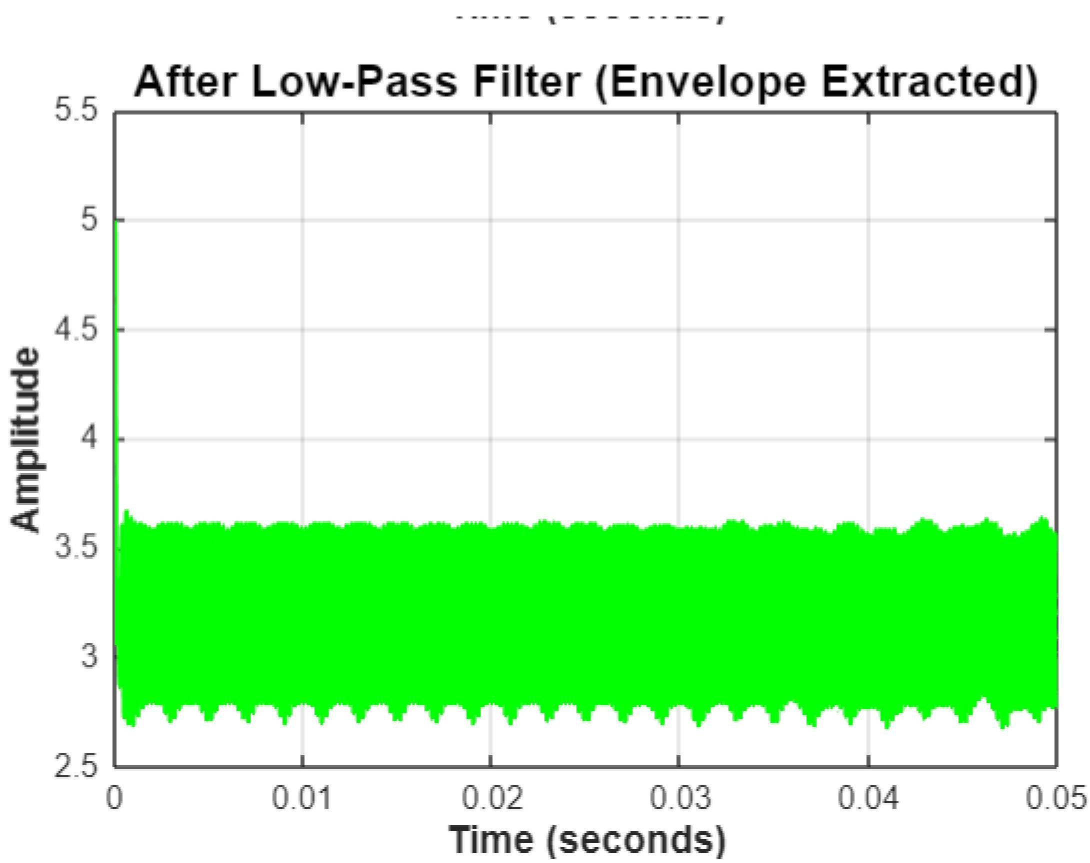


Figure 8 message signal after being applied with the low pass filter

Standard AM includes a carrier term, so the envelope contains a DC component equal to A_c To recover the true message, this constant must be removed:

$M_{demod}(t)=(e(t)-mean(e(t)))/A_c*K_a$

In MATLAB:

demodulated_signal = (envelope - mean(envelope)) / (A_c * K_a);

The signal is then normalized:

demodulated_signal = demodulated_signal / max(abs(demodulated_signal));

So in the end we get the demodulated signal

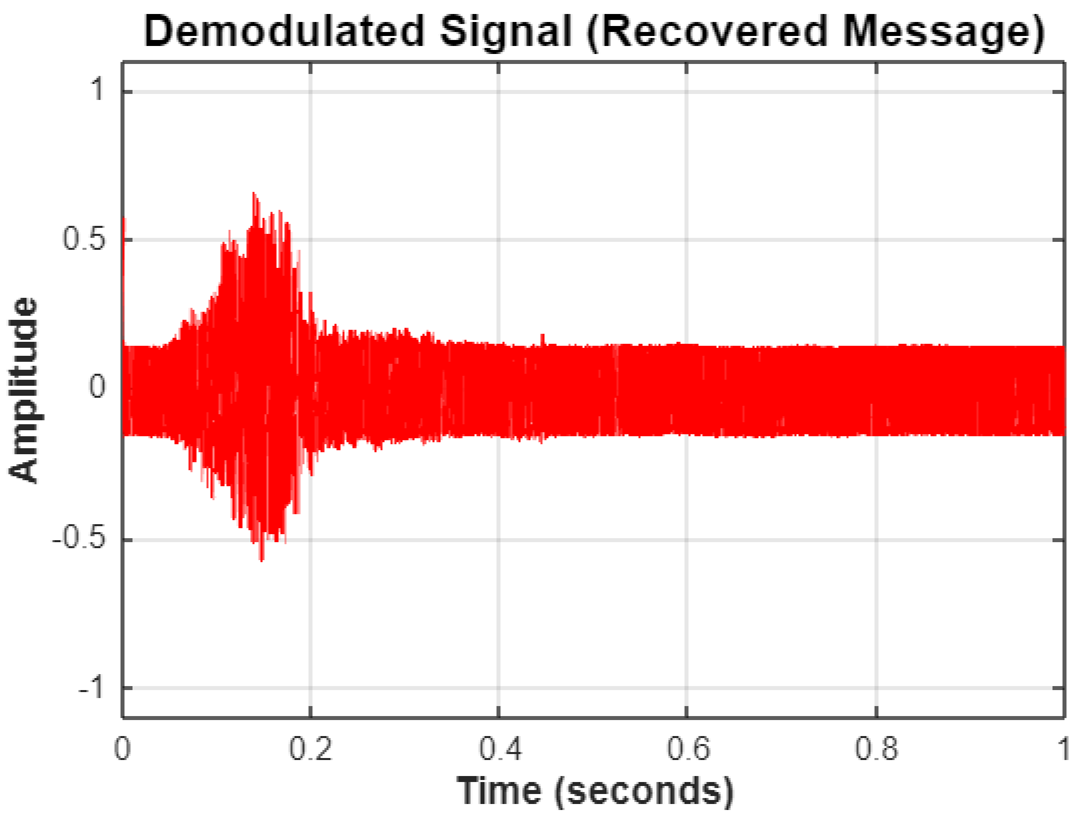


Figure 9 demodulated signal in time domain

6. Problem Formulation

Designing AM transmission for real audio introduces several challenges addressed directly in the code:

Using a fixed sampling frequency would distort the audio.
Solution: The program automatically sets:

$F_s = F_{audio}$

So, modulation occurs at the correct resolution.

Users may enter $K_a > 1$ or $K_a < 0$, causing envelope distortion.
Solution: The code stops execution and prompts the user to enter a value between 0 and 1.

Envelope detection introduces high-frequency components at:

$f = 2 f_c, 4f_c$

Solution: Use a 100-tap FIR low-pass filter designed with:

```
designfilt('lowpassfir', 'FilterOrder', 100, ...
        'CutoffFrequency', cutoff_freq, 'SampleRate', Fs);
```

Envelope detection produces:

$A_c + K_a * m(t)$

The code subtracts the mean and rescales the signal to remove unwanted bias.

7. Required Resources

Software

- MATLAB R2020+. In our case we used MATLAB 2024
- No extra toolboxes required (uses built-in FIR filter functions)

Functions Used

- audioread() – load audio
- uigetfile() – select audio file
- filtfilt() – zero-phase filtering
- fft, fftshift – frequency analysis
- plot, stem – visualizations
- sound() – audio playback

Hardware

- A standard PC or laptop
- Speakers/headphones

No external hardware is needed because the system is fully simulated.

8. Findings and Results

After running the code, the following results were obtained:

- The AM waveform clearly shows amplitude variation based on the audio message.
- No clipping occurs when $0 < K_a < 1$.
- The carrier frequency is visible in time domain plot.

The FFT confirms:

- Strong carrier peaks at $\pm f_c$.
- Symmetrical upper and lower sidebands.
- Message spectrum centered at zero frequency.

The delta diagram matches the theoretical AM model.

Envelope detection successfully recovered the message:

- Rectified waveform contains the correct envelope shape.
- FIR LPF removes all carrier remnants.
- Output closely matches the original audio.
- Listening test (sound()) confirms high clarity.

Even with different carrier frequencies and K_a values, the system remained stable.

9. Conclusion

The MATLAB implementation successfully demonstrated the complete process of transmitting and recovering recorded audio using Standard Amplitude Modulation (DSB-LC). The system provides full parameter control (carrier frequency, amplitude, sensitivity) and accurately performs modulation, spectral analysis, and demodulation.

Envelope detection combined with a well-designed low-pass FIR filter proved effective for recovering high-quality audio. Frequency-domain plot validate theoretical expectations, and listening tests confirm the recovered audio sounds almost identical to the original.

The project meets all communication, design, and signal-processing requirements.

10. References

1. Haykin – Communication Systems:

Used in:

- Background (AM theory, modulation types)
- Theoretical Modeling (AM formula, standard definitions)
- Comparative Literature Review (AM vs DSB-SC vs FM)

Link:
https://www.wiley.com/en-us/Communication+Systems%2C+4th+Edition-p-9780471178699

2. B. P. Lathi – Modern Digital and Analog Communication Systems

Used in:

- Theoretical Modeling (modulation index, conditions for $\mu < 1$)
- Problem Requirements & Constraints (Nyquist sampling, AM bandwidth)

Link:
https://global.oup.com/ushe/product/modern-digital-and-analog-communication-systems-9780190686842

3. Carlson – Communication Systems: Introduction to Signals and Noise

Used in:

- Background (AM broadcasting history)
- Theoretical Modeling (sidebands, frequency-domain expansion)
- Comparative Literature Review (power efficiency, carrier term info)
- **Link:** https://www.mheducation.com/highered/product/communication-systems-carlson-gisser/M9780070111273.html