AMPLIIUDE MODULATION IN MATLAB

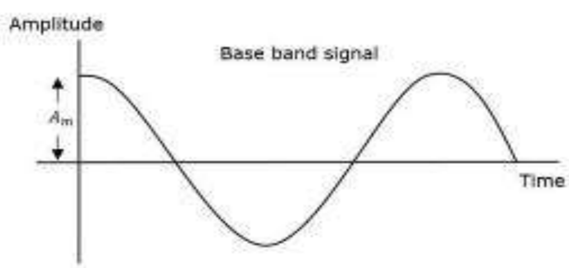
Experiment No: **Date:**

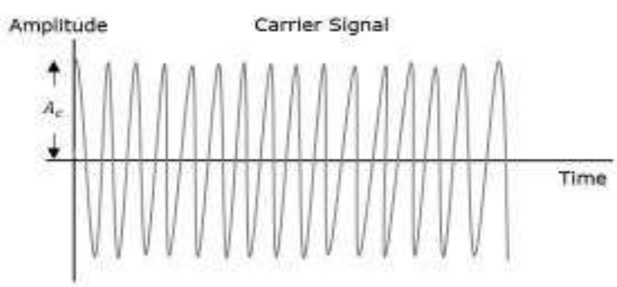
Aim: To implement amplitude modulation and demodulation using MATLAB.

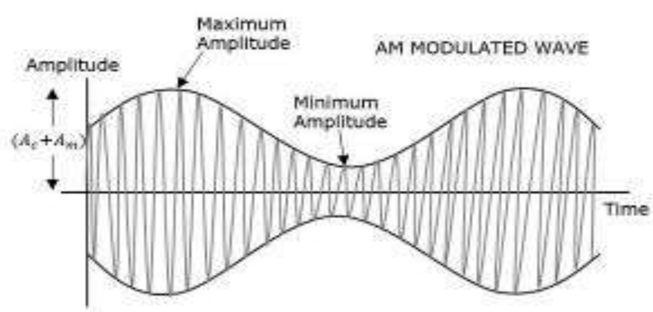
Brief Theory/Equations:

A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated.

According to the standard definition, “The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.” Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following figures.







The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as Envelope. It is the same as that of the message signal.

Time-domain Representation of the Waves

Let the modulating signal be,

$$m(t) = A_m cos(2\pi f_m t)$$

and the carrier signal be,

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

Where,

Am and *Ac* are the amplitude of the modulating signal and the carrier signal respectively. *fm* and *fc* are the frequency of the modulating signal and the carrier signal respectively. Then, the equation of Amplitude Modulated wave will be

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

Modulation index:

A carrier wave, after being modulated, if the modulated level is calculated, then such an attempt is called as Modulation Index or Modulation Depth. It states the level of modulation that a carrier wave undergoes.

$$s(t) = A_c [1 + \left(\frac{A_m}{A_c}\right) cos(2\pi f_m t)]cos(2\pi f_c t)$$

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

Where, μ is Modulation index and it is equal to the ratio of Am and Ac. Hence, we can calculate the value of modulation index by using the above formula, when the amplitudes of the message and carrier signals are known.

Now, let us derive one more formula for Modulation index by considering Equation. We can use this formula for calculating modulation index value, when the maximum and minimum amplitudes of the modulated wave are known.

Let Amax and Amin be the maximum and minimum amplitudes of the modulated wave. We will get the maximum amplitude of the modulated wave, when *cos(2πfmt)* is 1. *Amax=Ac+Am.*

We will get the minimum amplitude of the modulated wave, when *cos(2πfmt)*is -1. *Amin=Ac−Am*

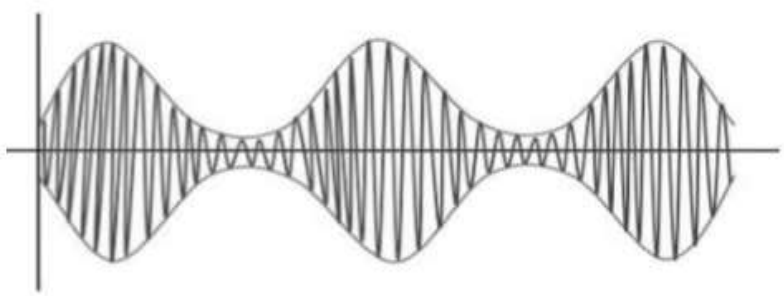
$$Amax+Amin = Ac+Am+Ac-Am=2\ Ac \ \Rightarrow\ Ac = \frac{Amax+Amin}{2}$$

$$Amax-Amin = Ac+Am-(Ac-Am)=2\ Am \quad \Rightarrow Am=\frac{Amax-Amin}{2}$$

$$\frac{\mathrm{Am}}{\mathrm{Ac}} = \frac{(Amax-Amin)/2}{(Amax+Amin)/2} \Rightarrow \mu = \frac{Amax-Amin}{Amax+Amin}$$

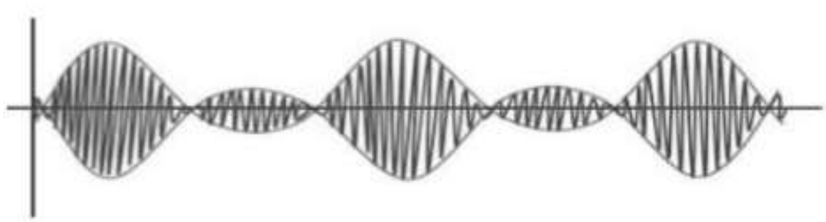
Therefore, there are the two formulas for Modulation index. The modulation index or modulation depth is often denoted in percentage called as Percentage of Modulation. We will get the **percentage of modulation**, just by multiplying the modulation index value with 100. For a perfect modulation, the value of modulation index should be 1, which implies the percentage of modulation should be 100%. For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as **Under-modulation**. Such a wave is called as an **under-modulated wave**.

Under-Modulated wave



If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an **over-modulated wave**. It would look like the following figure. As the value of the modulation index increases, the carrier experiences a 180° phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such an over-modulated wave causes interference, which cannot be eliminated.

Over-Modulated wave



Reference: Modern Digital and Analog Communication by B.P.Lathi

Bandwidth of AM:

Bandwidth (BW) is the difference between the highest and lowest frequencies of the signal. Mathematically, we can write it as $BW = f_{max} - f_{min}$ Consider the following equation of amplitude modulated wave.

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

$$s(t) = A_c\, cos(2\pi f_c t) + A_c\, \mu\, cos(2\pi f_c t)cos(2\pi f_m t)$$

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

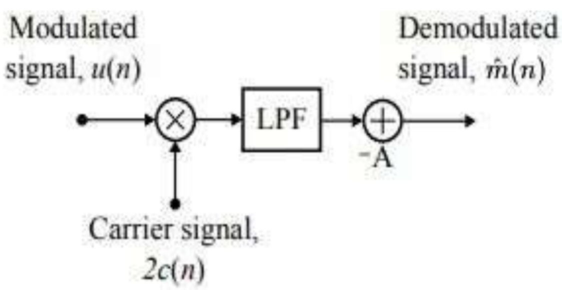
Hence, the amplitude modulated wave has three frequencies. Those are carrier frequency f_c, upper sideband frequency f_c+f_m and lower sideband frequency f_c-f_m Here, $f_{max}=f_c+f_m$ and $f_{min}=f_c-f_m$. Substitute, f_{max} and f_{min} values in bandwidth formula.

$$BW=(f_c+f_m) - (f_c-f_m) \Rightarrow BW=2f_m$$

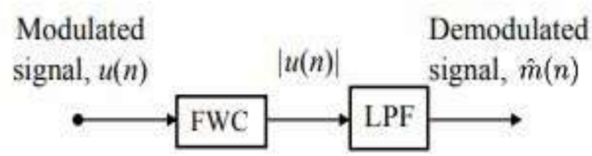
Thus, it can be said that the bandwidth required for amplitude modulated wave is twice the frequency of the modulating signal.

Amplitude Demodulation:

There are two types of demodulation: synchronous demodulation and asynchronous demodulation. They differ in the use of a carrier signal in demodulation processes. The receiver must generate a carrier signal synchronized in phase and frequency for synchronous demodulation, but the carrier signal is not needed in asynchronous demodulation.



(a) Synchronous demodulator



(b) Asynchronous demodulator

Synchronous Demodulation. This kind of demodulation can be referred to as a coherent or product detector. Figure (a) shows a flow chart of a synchronous demodulator. At the receiver, we multiply the incoming modulated signal by a local carrier of frequency and phase in synchronism with the carrier used at the transmitter.

$$e(t) = u(t)c(t) = [A + m(t)]\, cos^2(w_c t)$$
$$= \frac{1}{2}\, [A + m(t)] + \frac{1}{2}\, [A + m(t)]\, cos(2w_c t)$$

Let us denote $m_a(t) = A + m(t)$

$$e(t) = \frac{1}{2}\, m_a(t) + \frac{1}{2}\, m_a(t)\, cos(2w_c t)$$

The fourier transform of the signal e(t) is

$$E(w) = \frac{1}{2}\, M_a(w) + \frac{1}{2}[\, M_a(w + 2w_c\,) + M_a(w - 2w_c\,)$$

The spectrum $E(w)$ consists of three components as shown in figure. The first component is the message spectrum. The two other components, which are the modulated signal of m(t) with carrier frequency $2w_c$., are centered at $\pm 2w_c$. The signal *e(t)* is then filtered by a lowpass filter (LPF) with a cut-off frequency of fc to yield $\frac{m_a(t)}{2}$. We can fully get $m_a(t)$ by multiplying the output by two. We can also get rid of the inconvenient fraction 1 2 from the output by using the carrier 2 $cos(w_c)$. instead of $cos(w_c)$. Finally, the message signal m(t) can be recovered by ˆm(t) = $m_a(t)$ − A.

Asynchronous Demodulation. An asynchronous demodulator can be referred to as an envelope detector. For an envelope detector, the modulated signal *u(t)* must satisfy the requirement that *A + m(t) ≥ 0, ∀t.* A block diagram of the asynchronous demodulator is shown in Figure (b). The incoming modulated signal, u(t), is passed through a full wave rectifier (FWR) which acts as an absolute function. The FWR output which is the absolute value of *u(t)*, |*u(t)*|, is then filtered by a low-pass filter resulting in the demodulated signal, *ˆm(t).*

Algorithm:

- Define the sampling frequency $f_s \geq 2(f_m)$ say, $f_s = 100Hz$;
- Define the time range using the sampling frequency t= -10 : 1/f_s: 10
- Consider, message signal, $m = A_m \cos \omega_m t$, where $f_m = 1Hz$ and carrier signal $c = A_c \cos \omega_c t$where $f_c = 10Hz$. Keep the amplitude of message and carrier signal same.
- Assume $w_m = 2\pi f_m$ and $w_c = 2\pi f_c$
- For AM, $\boldsymbol{s(t) = [A_c + A_m\, cos(2\pi f_m t)] cos(2\pi f_c t)}$
- Figure1: Plot input signal, carrier signal, AM signal using subplot;
- Figure 2: plot the frequency spectrum of AM signal using the commands fft and fftshift.
    - Define n=length(t) %gives no. of columns in t
    - Define the step size for frequency axis fp which should be of same size as that of t. (matrix dimensions must match for plotting).
        - df=fs/n; where fs is the sampling frequency
    - Define frequency axis fp = -fs/2:df:fs/2-df ($\pm$df for getting same size matrices)
    - Take the fourier transform of AM using fft command

- Y = fft(x) returns the discrete Fourier transform (DFT) of vector x, computed with a fast Fourier transform (FFT) algorithm.
- Y = fftshift(X) rearranges the outputs of fft by moving the zero-frequency component to the center of the array. It is useful for visualizing a Fourier transform with the zero-frequency component in the middle of the spectrum.
- Can write in a single syntax as y=fftshift(fft(x));
- Plot the frequency spectrum of AM using the command plot(fp,y).

- **Demodulate the AM signal**
  - Multiply the above signal with carrier signal.
    - Do .* element wise multiplication
- Do the low pass filtering of the above signal using butter and filter command.
    - [b,a] = butter(n,Wn,'ftype'). **This creates a filter**
    - [b,a] = butter(n,Wn) designs an order n lowpass digital Butterworth filter with normalized cutoff frequency Wn. It returns the filter coefficients in length n+1 row vectors b and a, with coefficients in descending powers of z.

$$H(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(n+1)z^{-n}}{1 + a(2)z^{-1} + \ldots + a(n+1)z^{-n}}$$

    - where the string 'ftype' is one of the following:
      - 'high' for a highpass digital filter with normalized cutoff frequency Wn
      - 'low' for a lowpass digital filter with normalized cutoff frequency Wn
      - 'stop' for an order 2*n bandstop digital filter if Wn is a two-element vector, Wn = [w1 w2]. The stopband is w1 < ω < w2.
      - 'bandpass' for an order 2*n bandpass filter if Wn is a two-element vector, Wn = [w1 w2]. The passband is w1 < ω < w2. Specifying a two-element vector, Wn, without an explicit 'ftype' defaults to a bandpass filter.
      - Cutoff frequency, Wn is that frequency where the magnitude response of the filter is . For butter, the normalized cutoff frequency Wn must be a number between 0 and 1, where 1 corresponds to the Nyquist frequency, π radians per sample.
      - Choose order, n=3 ; Wn=fm/fs; ftype='low'
- Use the filter command to apply the filter
  - y = filter(b,a,X), where b and a are the coefficients obtained in the previous step and X will the multiplied signal output. y is the filtered output. Adjust order so that you get approximate input signal.
- Figure 3: Plots for AM demodulation
  - Include three figures using subplot(211) to subplot(212)
    - 1st - multiplied signal output (modulated AM signal*Carrier Signal)
    - 2nd – filtered output y. Include message signal as well in this
      - Example: plot(t,m,t,y)

Expected Output waveforms:

Figure 1:

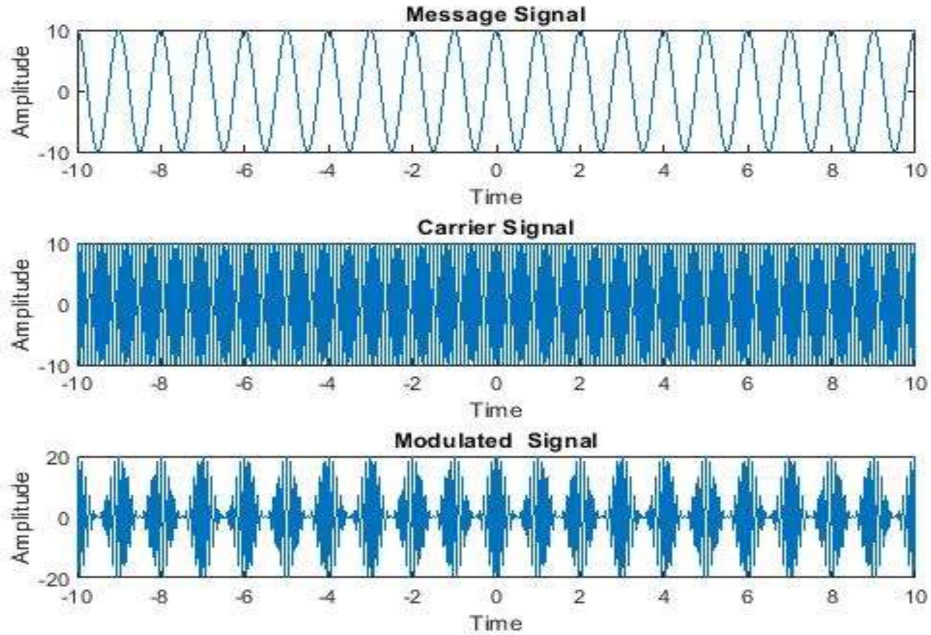


Figure 2:

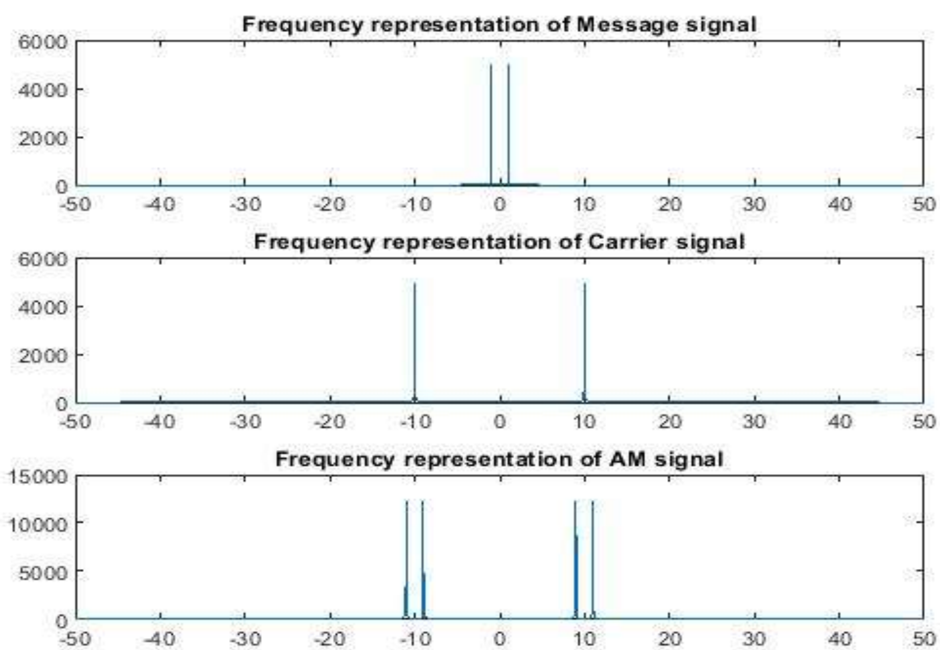
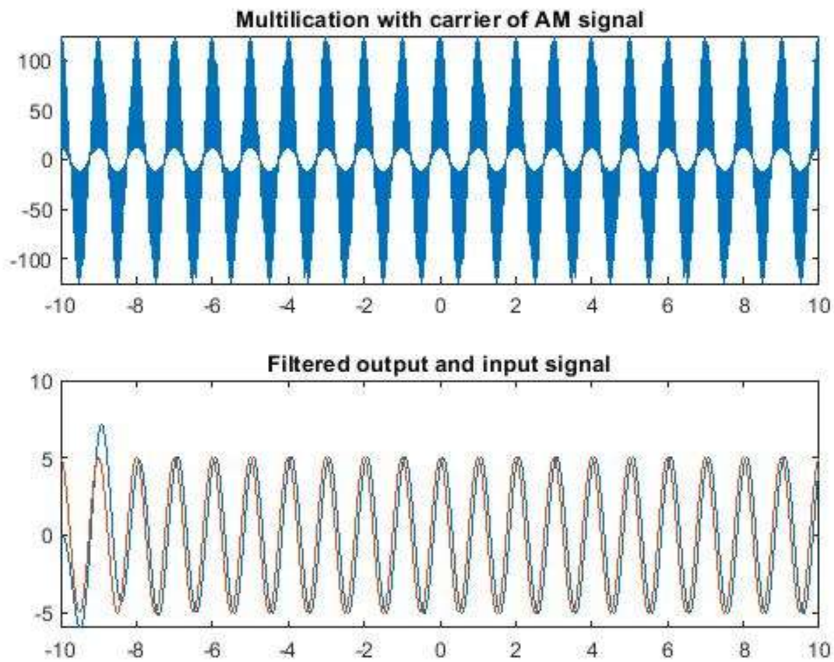


Figure 3:



Code:

Output Waveforms:

Conclusion:

Remarks: **Signature:**

References:

- NPTEL communication systems lectures
  https://www.youtube.com/watch?v=S8Jod9AtpN4&list=PL7748E9BEC4ED83CA&index=8
  https://www.youtube.com/watch?v=NTcDup0_B4w&list=PL7748E9BEC4ED83CA&index=7
- Modern Analog and Digital Communication by B.P. Lathi (3^{rd} or 4^{th} edition)
- Communication Systems by Simon Haykin (4^{th} edition)